

Communications Toolbox™

User's Guide



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Communications Toolbox™ User's Guide

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Input, Output, and Display

Learn how to input, output and display data and signals with Communications Toolbox.

- “Signal Terminology” on page 1-2
- “Export Data to MATLAB” on page 1-3
- “Sources and Sinks” on page 1-8
- “Spreading Sequences” on page 1-26
- “Read Signals From Hardware Devices” on page 1-33

Signal Terminology

This section defines important Communications Toolbox terms related to matrices, vectors, and scalars, as well as frame-based and sample-based processing.

Matrices, Vectors, and Scalars

This document uses the unqualified words *scalar* and *vector* in ways that emphasize a signal's number of elements, not its strict dimension properties:

- A *scalar* signal contains a single element. The signal could be a one-dimensional array with one element, or a matrix of size 1-by-1.
- A *vector* signal contains one or more elements, arranged in a series. The signal could be a one-dimensional array, a matrix that has exactly one column, or a matrix that has exactly one row. The number of elements in a vector is called its *length* or, sometimes, its *width*.

In cases when it is important for a description or schematic to distinguish among different types of scalar signals or different types of vector signals, this document mentions the distinctions explicitly. For example, the terms *one-dimensional array*, *column vector*, and *row vector* distinguish among three types of vector signals.

The *size* of a matrix is the pair of numbers that indicate how many rows and columns the matrix has. The *orientation* of a two-dimensional vector is its status as either a row vector or column vector. A one-dimensional array has no orientation – this is sometimes called an unoriented vector.

A matrix signal that has more than one row and more than one column is called a *full matrix* signal.

Export Data to MATLAB

In this section...

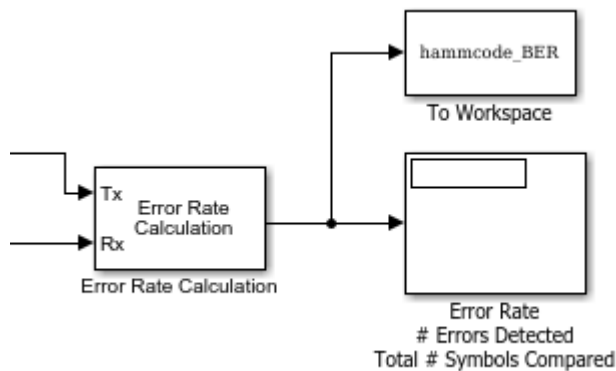
- “Use a To Workspace Block” on page 1-3
- “Configure the To Workspace Block” on page 1-4
- “View Error Rate Data in Workspace” on page 1-4
- “Send Signal and Error Data to Workspace” on page 1-5
- “View Signal and Error Data in Workspace” on page 1-6
- “Analyze Signal and Error Data” on page 1-7

Use a To Workspace Block

This section explains how to send data from a Simulink® model to the MATLAB® workspace so you can analyze the results of simulations in greater detail.

You can use a To Workspace block, from the DSP System Toolbox™/Sinks library to send data to the MATLAB workspace as a vector. For example, you can send the error rate data from the Hamming code model, described in the section Reducing the Error Rate Using a Hamming Code on page 8-82. To insert a To Workspace block into the model, follow these steps:

- 1** To open the model, at the MATLAB prompt, enter `doc_hamming`.
- 2** To add a To Workspace block, begin typing the name 'to workspace' in the model window and select the To Workspace block from the DSP System Toolbox/Sinks library. Connect it as shown.



Tip More than one To Workspace block is available. Be sure to select To Workspace from the DSP System Toolbox / Sinks sublibrary.

Configure the To Workspace Block

To configure the To Workspace block, follow these steps:

- 1 Double-click the block to display its dialog box.
- 2 Type `hamcode_BER` in the **Variable name** field.
- 3 Type `1` in the **Limit data points to last** field. This limits the output vector to the values at the final time step of the simulation.
- 4 Click **OK**.

When you run a simulation, the model sends the output of the Error Rate Calculation block to the workspace as a vector of size 3, called `hamming_BER`. The entries of this vector are the same as those shown by the Error Rate Display block.

View Error Rate Data in Workspace

After running a simulation, you can view the output of the To Workspace block by typing the following commands at the MATLAB prompt:

```
format short e  
hamcode_BER
```


The vector output is the following:

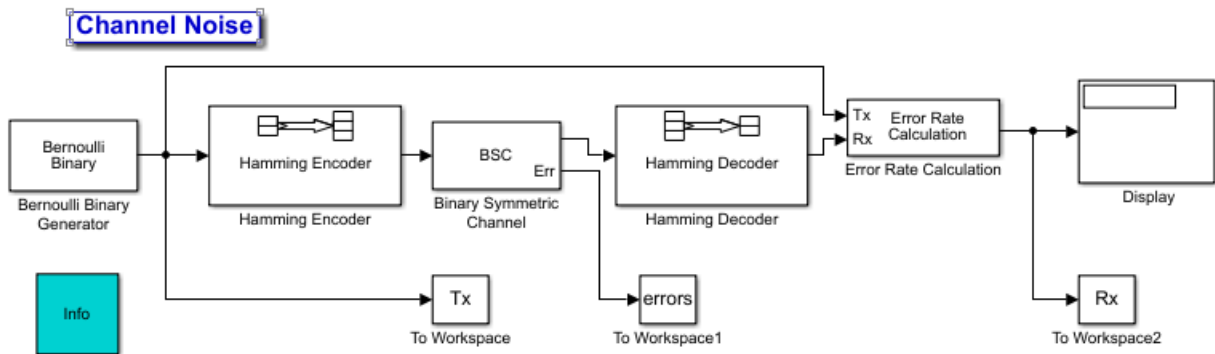
```
hammcode_BER =
5.4066e-003  1.0000e+002  1.8496e+004
```

The command `format short e` displays the entries of the vector in exponential form. The entries are as follows:

- The first entry is the error rate.
- The second entry is the total number of errors.
- The third entry is the total number of comparisons made.

Send Signal and Error Data to Workspace

To analyze the error-correction performance of the Hamming code, send the transmitted signal, the received signal, and the error vectors, created by the Binary Symmetric Channel block, to the workspace. An example of this is shown in the following figure.



- 1 Type `doc_channel` at the MATLAB command line to open the starter model,
- 2 Double-click the Binary Symmetric Channel block to open its dialog box, and select **Output error vector**. This creates an output port for the error data.
- 3 Move blocks to make room so that you can insert Hamming Encoder and Hamming Decoder blocks. To find them, start typing `Hamming` in the model window. Select them from the options presented. These Hamming Encoder and Hamming Decoder blocks are in the Communications Toolbox/Error Detection and Correction /Block sublibrary.

- 4 Add three To Workspace blocks into the model window and connect them as shown in the preceding figure.

Tip More than one To Workspace block is available. Be sure to select To Workspace from the DSP System Toolbox / Sinks sublibrary.

- 5 Double-click the left To Workspace block.
 - Type Tx in the **Variable name** field in the block's dialog box. The block sends the transmitted signal to the workspace as an array called Tx.
 - In the **Save 2-D signals as** field, select 3-D array (concatenate along third dimension). This preserves each frame as a separate column of the array Tx.
 - Click **OK**.
- 6 Double-click the middle To Workspace block:
 - Type errors in the **Variable name** field.
 - In the **Save 2-D signals as** field, select 3-D array (concatenate along third dimension). This preserves each frame as a separate column of the array Tx.
 - Click **OK**.
- 7 Double-click the right To Workspace block:
 - Type Rx in the **Variable name** field.
 - In the **Save 2-D signals as** field, select 3-D array (concatenate along third dimension). This preserves each frame as a separate column of the array Tx.
 - Click **OK**.

View Signal and Error Data in Workspace

After running a simulation, you can display individual frames of data. For example, to display the tenth frame of Tx, at the MATLAB prompt type

```
Tx(:, :, 10)
```

This returns a column vector of length 4, corresponding to the length of a message word. Usually, you should not type Tx by itself, because this displays the entire transmitted signal, which is very large.

To display the corresponding frame of errors, type

```
errors(:, :, 10)
```

This returns a column vector of length 7, corresponding to the length of a codeword.

To display frames 1 through 5 of the transmitted signal, type

```
Tx(:, :, 1:5)
```

Analyze Signal and Error Data

You can use MATLAB to analyze the data from a simulation. For example, to identify the differences between the transmitted and received signals, type

```
diffs = Tx~=Rx;
```

The vector `diffs` is the XOR of the vectors `Tx` and `Rx`. A 1 in `diffs` indicates that `Tx` and `Rx` differ at that position.

You can determine the indices of frames corresponding to message words that are incorrectly decoded with the following MATLAB command:

```
error_indices = find(diffs);
```

A 1 in the vector `not_equal` indicates that there is at least one difference between the corresponding frame of `Tx` and `Rx`. The vector `error_indices` records the indices where `Tx` and `Rx` differ. To view the first incorrectly decoded word, type

```
Tx(:, :, error_indices(1))
```

To view the corresponding frame of errors, type

```
errors(:, :, error_indices(1))
```

Analyze this data to determine the error patterns that lead to incorrect decoding.

Sources and Sinks

In this section...

“Data Sources” on page 1-8

“Noise Sources” on page 1-11

“Sequence Generators” on page 1-12

“Scopes” on page 1-16

“View a Sinusoid” on page 1-17

“View a Modulated Signal” on page 1-19

Communications Toolbox provides sources, sinks, and display devices that facilitate analysis of communication system performance.

Data Sources

Use the functions and blocks listed in “Sources and Sinks” to generate random data to simulate a signal source.

Random Symbols

The `randsrc` function generates random matrices whose entries are chosen independently from an alphabet that you specify, with a distribution that you specify. A special case generates bipolar matrices.

For example, the command below generates a 5-by-4 matrix whose entries are random, independently chosen, and uniformly distributed in the set {1,3,5}.

```
a = randsrc(5,4,[1,3,5])
```

```
a = 5×4
```

```
5    1    1    1
5    1    5    3
1    3    5    5
5    5    3    5
3    5    5    5
```

To skew the distribution so that 1 is twice as likely to occur as either 3 or 5, use the command below. The third input argument has two rows, one of which indicates the possible values of b and the other indicates the probability of each value.

```
b = randsrc(5,4,[1,3,5; .5, .25, .25])
```

```
b = 5×4
```

```

3     5     3     5
1     3     1     3
5     1     1     1
5     3     1     5
3     1     1     1

```

Random Integers

In MATLAB®, the `randi` function generates random integer matrices whose entries are in a range that you specify. A special case generates random binary matrices.

For example, the command below generates a 5-by-4 matrix containing random integers between 2 and 10.

```
c = randi([2,10],5,4)
```

```
c = 5×4
```

```

5     6     4     6
5     6     8    10
8     7     7     5
9     8     3     7
3     8     3     4

```

If your desired range is $[0,10]$ instead of $[2,10]$, you can use either of the commands below. They produce different numerical results, but use the same distribution.

```
d = randi([0,10],5,4);
```

```
e = randi([0 10],5,4);
```

In Simulink®, the Random Integer Generator and Poisson Integer Generator blocks both generate vectors containing random nonnegative integers. The Random Integer Generator block uses a uniform distribution on a bounded range that you specify in the block mask. The Poisson Integer Generator block uses a Poisson distribution to determine its output. In particular, the output can include any nonnegative integer.

Random Bit Error Patterns

In MATLAB, the `randerr` function generates matrices whose entries are either 0 or 1. However, its options are different from those of `randi`, because `randerr` is meant for testing error-control coding. For example, the command below generates a 5-by-4 binary matrix, where each row contains exactly one 1.

```
f = randerr(5,4)
```

```
f = 5×4
```

```
0     0     0     1
1     0     0     0
1     0     0     0
0     0     1     0
0     0     0     1
```

You might use such a command to perturb a binary code that consists of five four-bit codewords. Adding the random matrix `f` to your code matrix (modulo 2) introduces exactly one error into each codeword.

On the other hand, to perturb each codeword by introducing one error with probability 0.4 and two errors with probability 0.6, use the command below instead. Each row has one '1' with probability 0.4, otherwise two '1's

```
g = randerr(5,4,[1,2; 0.4,0.6])
```

```
g = 5×4
```

```
1     0     1     0
0     1     0     1
0     0     0     1
1     0     0     1
0     0     1     0
```

Adding the random matrix `g` to your code matrix (modulo 2) introduces one or two errors into each codeword with the specified probability of occurrence for each. The probability matrix that is the third argument of `randerr` affects only the number of 1s in each row, not their placement.

As another application, you can generate an equiprobable binary 100-element column vector using any of the commands below. The three commands produce different

numerical outputs, but use the same distribution. The third input arguments vary according to each function's particular way of specifying its behavior.

```
binarymatrix1 = randsrc(100,1,[0 1]); % Possible values are 0,1
binarymatrix2 = randi([0 1],100,1); % Two possible values
binarymatrix3 = randerr(100,1,[0 1; 0.5 0.5]); % No 1s, or one 1
```

In Simulink, the Bernoulli Binary Generator block generates random bits and is suitable for representing sources. The block considers each element of the signal to be an independent Bernoulli random variable. Also, different elements need not be identically distributed.

Noise Sources

Construct noise generator blocks in Simulink to simulate communication links.

Random Noise Generators

You can construct random noise generators to simulate channel noise by using the MATLAB Function block with random number generating functions. Construct different types of channel noise by using the following combinations.

Distribution	Block	Function
Gaussian	MATLAB Function	wgn
Rayleigh	MATLAB Function	randn
Rician	MATLAB Function	randn
Uniform on a bounded interval	MATLAB Function	rand

See “Random Noise Generators” on page 4-39 for an example of how Rayleigh and Rician distributed noise is created.

Gaussian Noise Generator

In MATLAB®, the wgn function generates random matrices using a white Gaussian noise distribution. You specify the power of the noise in either dBW (decibels relative to a watt), dBm, or linear units. You can generate either real or complex noise.

For example, the command below generates a column vector of length 50 containing real white Gaussian noise whose power is 2 dBW. By default, the power type in dBW and load impedance is 1 ohm.

```
y1 = wgn(50,1,2);
```

To generate complex white Gaussian noise whose power is 2 watts, across a load of 60 ohms, use either of the commands below.

```
y2 = wgn(50,1,2,60,'complex','linear');  
y3 = wgn(50,1,2,60,'linear','complex');
```

To send a signal through an additive white Gaussian noise channel, use the `awgn` function. See “AWGN Channel” on page 14-2 for more information.

Sequence Generators

Use the functions, System objects, and blocks listed in “Sources and Sinks” to generate sequences for spreading or synchronization in a communication system. You can generate pseudorandom sequences, synchronization codes, and orthogonal codes. For examples comparing correlation properties of these sequence generators, see “Spreading Sequences” on page 1-26.

- “Pseudorandom Sequences” on page 1-12
- “Model PN Sequence Generation With Linear Feedback Shift Register” on page 1-14
- “Synchronization Codes” on page 1-15
- “Orthogonal Codes” on page 1-15

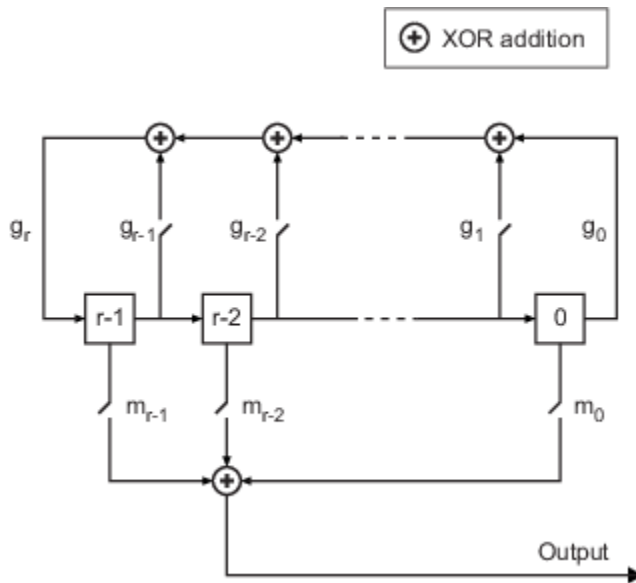
Pseudorandom Sequences

You can generate pseudorandom or pseudonoise (PN) sequences using these System objects in MATLAB and these blocks in Simulink. The applications of these sequences range from multiple-access spread spectrum communication systems to ranging, synchronization, and data scrambling.

Sequence	System object™	Block
Gold sequences	<code>comm.GoldSequence</code>	Gold Sequence Generator
Kasami sequences	<code>comm.KasamiSequence</code>	Kasami Sequence Generator

Sequence	System object™	Block
PN sequences	comm.PNSequence	PN Sequence Generator

To generate pseudorandom sequences, the underlying code implements shift registers, as illustrated in this diagram.



All r registers in the generator update their values at each time step according to the value of the incoming arrow to the shift register. The adders perform addition modulo 2. The shift register can be described by a binary polynomial in z , $g_r z^r + g_{r-1} z^{r-1} + \dots + g_0$. The coefficient g_i is 1 if there is a connection, or 0 if there is no connection, from the i th shift register to the adder.

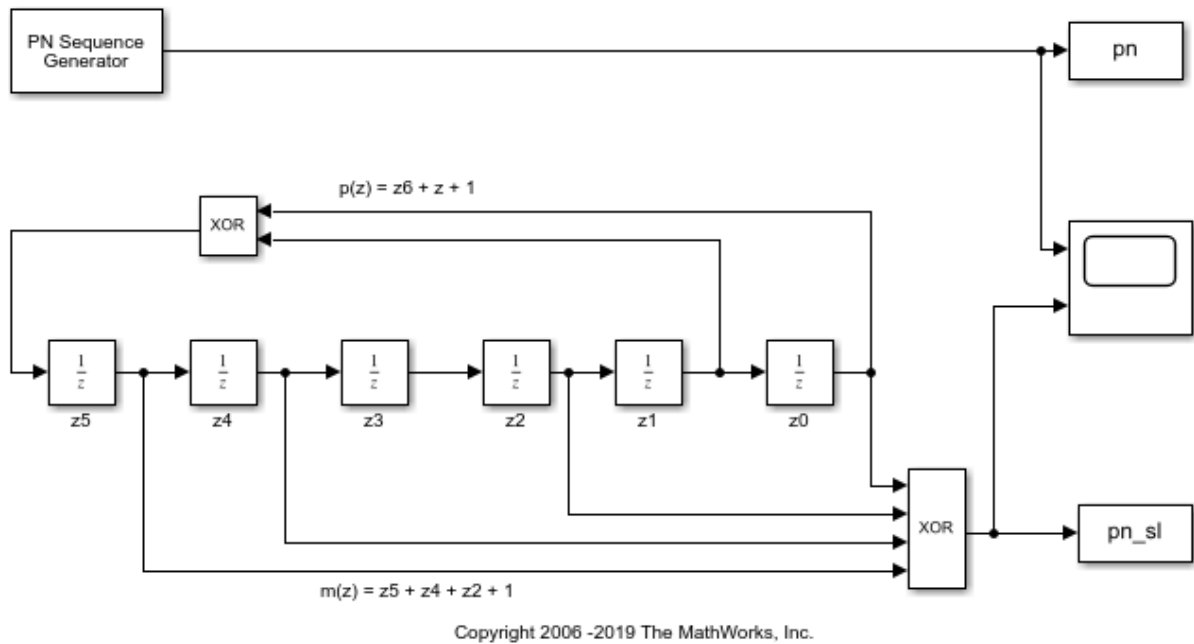
The coefficient m_i is 1 if there is a delay, or a 0 if there is no delay, from the i th shift register to the adder preceding the output. If the shift is zero, the m_0 switch is closed while all other m_k switches are open.

The Kasami and PN sequence generators use this polynomial description for their generator polynomial. The Gold sequence generator uses this polynomial description for the preferred first and second generator polynomial PN sequences.

Model PN Sequence Generation With Linear Feedback Shift Register

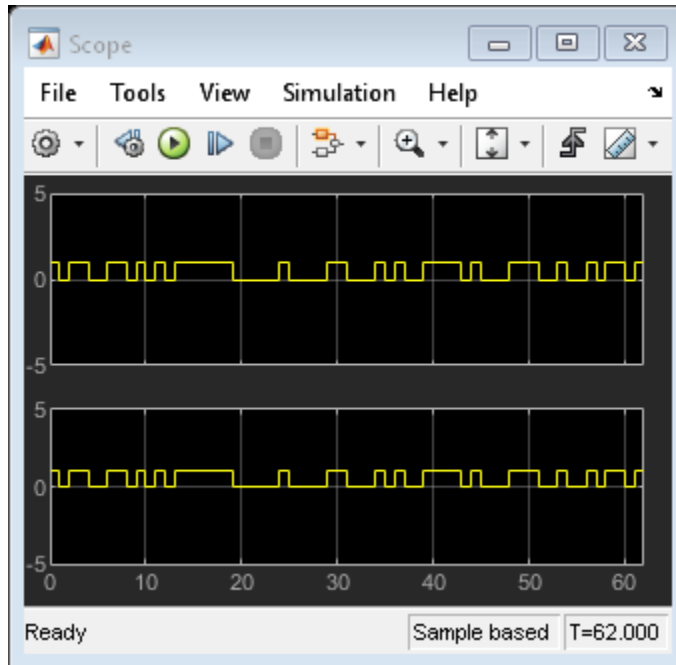
This example shows that sequences output from the PN Sequence Generator can be modeled using a linear feedback shift register (LFSR) built with primitive Simulink® blocks.

PN Sequence Generation



For the chosen generator polynomial, $p(z)=z^6+z+1$, the model generates a PN sequence of period 63, by using the PN Sequence Generator block and by modeling a LFSR using primitive Simulink blocks. The two parameters, Initial states and Output mask vector (or scalar shift value), are interpreted in the LFSR model schematic. The PreLoadFcn callback function is used to initialize runtime parameters. To view the callback functions, go to MODELING> SETUP> Model Settings> Model Properties, and select the Callbacks tab.

The scope output shows the two implementations produce matching PN sequences.



Using the PN Sequence Generator block allows you to easily generate PN sequences of large periods. To experiment further, open the model. Modify settings to see how the performance varies for different path delays or adjust the PN sequence generator parameters. You can experiment with different initial states, by changing the value of Initial states prior to running the simulation. For all values, the two generated sequences are the same.

Synchronization Codes

Use the `comm.BarkerCode` System object and Barker Code Generator block to generate Barker codes to perform synchronization. Barker codes are subsets of PN sequences. They are short codes, with a length at most 13, which are low-correlation sidelobes. A correlation sidelobe is the correlation of a codeword with a time-shifted version of itself.

Orthogonal Codes

Orthogonal codes are used for spreading to benefit from their perfect correlation properties. When used in multi-user spread spectrum systems, where the receiver is perfectly synchronized with the transmitter, the despreading operation is ideal.

Code	System object	Block
Hadamard codes	<code>comm.HadamardCode</code>	Hadamard Code Generator
OVSF codes	<code>comm.OVSFCode</code>	OVSF Code Generator
Walsh codes	<code>comm.WalshCode</code>	Walsh Code Generator

Scopes

The Comm Sinks block library contains scopes for viewing three types of signal plots:

- “Eye Diagrams” on page 1-16
- “Scatter Plots” on page 1-17
- “Signal Trajectories” on page 1-17

The following table lists the blocks and the plots they generate.

Block Name	Plots
Eye Diagram	Eye diagram of a signal
Constellation Diagram	Constellation diagram and signal trajectory of a signal

Eye Diagrams

An eye diagram is a simple and convenient tool for studying the effects of intersymbol interference and other channel impairments in digital transmission. When this software product constructs an eye diagram, it plots the received signal against time on a fixed-interval axis. At the end of the fixed interval, it wraps around to the beginning of the time axis. As a result, the diagram consists of many overlapping curves. One way to use an eye diagram is to look for the place where the eye is most widely opened, and use that point as the decision point when demapping a demodulated signal to recover a digital message.

The Eye Diagram block produces eye diagrams. This block processes discrete-time signals and periodically draws a line to indicate a decision, according to a mask parameter.

Examples appear in “View a Sinusoid” on page 1-17 and “View a Modulated Signal” on page 1-19.

Scatter Plots

A constellation diagram of a signal plots the signal's value at its decision points. In the best case, the decision points should be at times when the eye of the signal's eye diagram is the most widely open.

The Constellation Diagram block produces a constellation diagram from discrete-time signals. An example appears in “View a Sinusoid” on page 1-17.

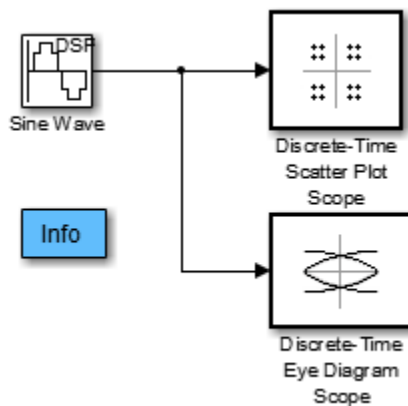
Signal Trajectories

A signal trajectory is a continuous plot of a signal over time. A signal trajectory differs from a scatter plot in that the latter displays points on the signal trajectory at discrete intervals of time.

The Constellation Diagram block produces signal trajectories. The Constellation Diagram block produces signal trajectories when the `ShowTrajectory` property is set to true. A signal trajectory connects all points of the input signal, irrespective of the specified decimation factor (`Samples per symbol`)

View a Sinusoid

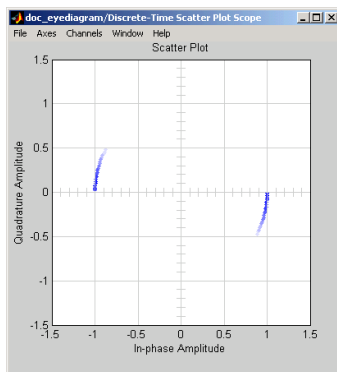
The following model produces a constellation diagram and an eye diagram from a complex sinusoidal signal. Because the decision time interval is almost, but not exactly, an integer multiple of the period of the sinusoid, the eye diagram exhibits drift over time. More specifically, successive traces in the eye diagram and successive points in the scatter diagram are near each other but do not overlap.



To open the model, enter `doc_eyediagram` at the MATLAB command line. To build the model, gather and configure these blocks:

- Sine Wave, in the Sources library of the DSP System Toolbox (*not* the Sine Wave block in the Simulink Sources library)
 - Set **Frequency** to `.502`.
 - Set **Output complexity** to `Complex`.
 - Set **Sample time** to `1/16`.
- Constellation Diagram, in the Comm Sinks library
 - On the **Constellation Properties** panel, set **Samples per symbol** to `16`.
- Eye Diagram, in the Comm Sinks library
 - On the **Plotting Properties** panel, set **Samples per symbol** to `16`.
 - On the **Figure Properties** panel, set **Scope position** to `figposition([42.5 55 35 35]);`.

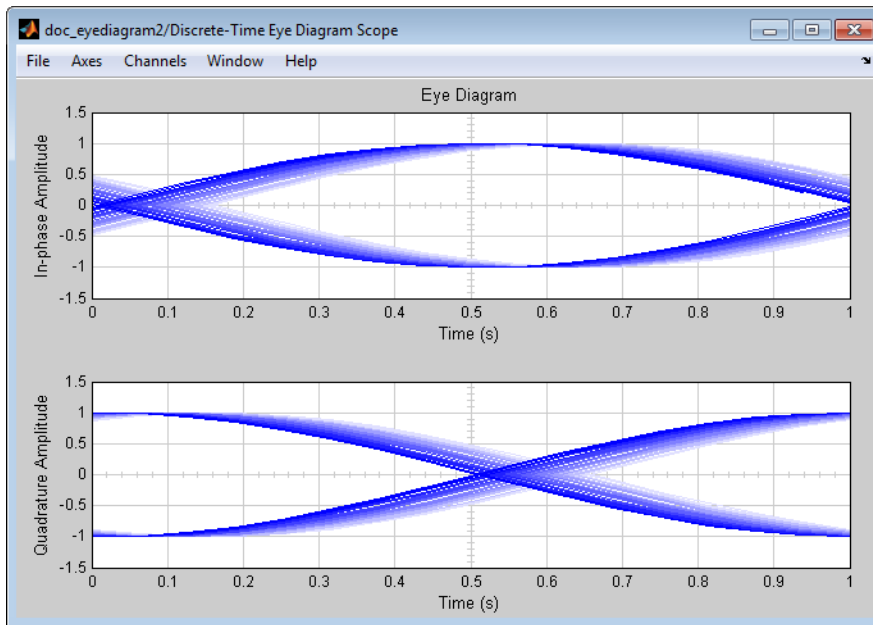
Connect the blocks as shown in the preceding figure. From the model window's **Simulation** menu, choose **Model Configuration parameters**. In the **Configuration Parameters** dialog box, set **Stop time** to `250`. Running the model produces the following scatter diagram plot.



The points of the scatter plot lie on a circle of radius 1. Note that the points fade as time passes. This is because the box next to **Color fading** is checked under **Rendering Properties**, which causes the scope to render points more dimly the more time that passes after they are plotted. If you clear this box, you see a full circle of points.

The Constellation Diagram block displays a circular trajectory.

In the eye diagram, the upper set of traces represents the real part of the signal and the lower set of traces represents the imaginary part of the signal.



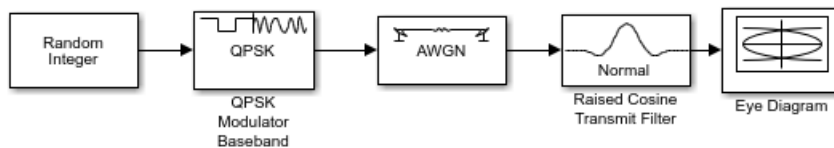
View a Modulated Signal

This multipart example creates an eye diagram, scatter plot, and signal trajectory plot for a modulated signal. It examines the plots one by one in these sections:

- “Eye Diagram of a Modulated Signal” on page 1-19
- “Constellation Diagram of a Modulated Signal” on page 1-22
- “Signal Trajectory of a Modulated Signal” on page 1-23

Eye Diagram of a Modulated Signal

The following model modulates a random signal using QPSK, filters the signal with a raised cosine filter, and creates an eye diagram from the filtered signal.



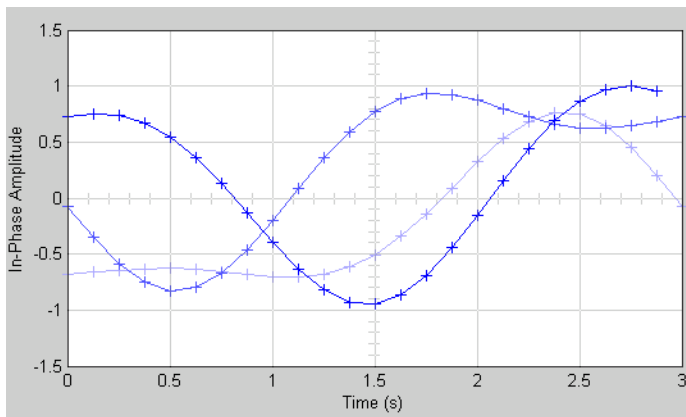
To open the model, enter `doc_signaldisplays` at the MATLAB command line. To build the model, gather and configure the following blocks:

- Random Integer Generator, in the Random Data Sources sublibrary of the Comm Sources library
 - Set **M-ary number** to 4.
 - Set **Sample time** to 0.01.
- QPSK Modulator Baseband, in PM in the Digital Baseband sublibrary of the Modulation library of Communications Toolbox, with default parameters
- AWGN Channel, in the Channels library of Communications Toolbox, with the following changes to the default parameter settings:
 - Set **Mode** to Signal-to-noise ratio (SNR).
 - Set **SNR (dB)** to 15.
- Raised Cosine Transmit Filter, in the Comm Filters library
 - Set **Filter shape** to Normal.
 - Set **Rolloff factor** to 0.5.
 - Set **Filter span in symbols** to 6.
 - Set **Output samples per symbol** to 8.
 - Set **Input processing** to Elements as channels (sample based).
- Eye Diagram, in the Comm Sinks library
 - Set **Samples per symbol** to 8.
 - Set **Symbols per trace** to 3. This specifies the number of symbols that are displayed in each trace of the eye diagram. A *trace* is any one of the individual lines in the eye diagram.
 - Set **Traces displayed** to 3.
 - Set **New traces per display** to 1. This specifies the number of new traces that appear each time the diagram is refreshed. The number of traces that remain in

the diagram from one refresh to the next is **Traces displayed** minus **New traces per display**.

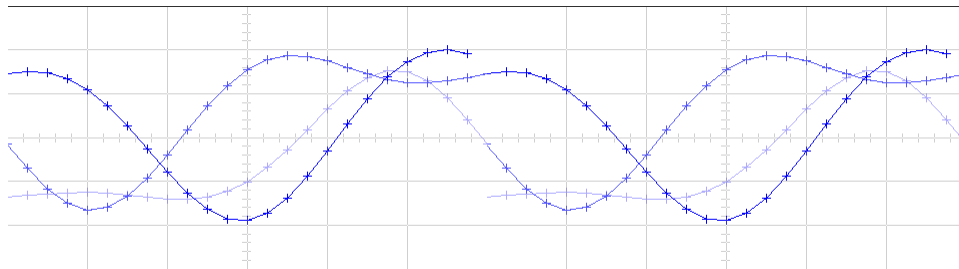
- On the **Rendering Properties** panel, set **Markers** to + to indicate the points plotted at each sample. The default value of **Markers** is empty, which indicates no marker.
- On the **Figure Properties** panel, set **Eye diagram to display** to In-phase only.

When you run the model, the Eye Diagram displays the following diagram. Your exact image varies depending on when you pause or stop the simulation.

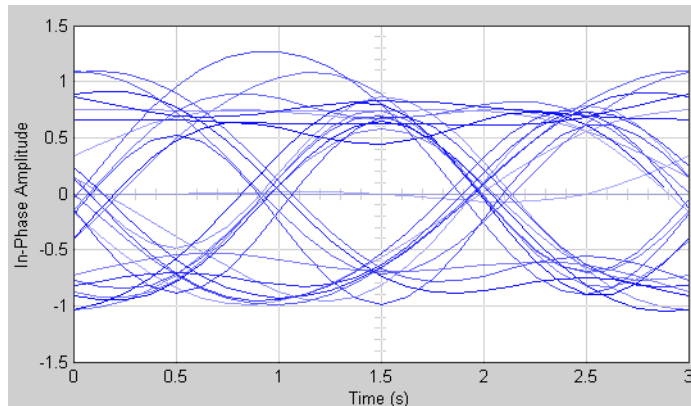


Three traces are displayed. Traces 2 and 3 are faded because **Color fading** under **Rendering Properties** is selected. This causes traces to be displayed less brightly the older they are. In this picture, Trace 1 is the most recent and Trace 3 is the oldest. Because **New traces per display** is set to 1, only Trace 1 is appearing for the first time. Traces 2 and 3 also appear in the previous display.

Because **Symbols per trace** is set to 3, each trace contains three symbols, and because **Samples per trace** is set to 8, each symbol contains eight samples. Note that trace 1 contains 24 points, which is the product of **Symbols per trace** and **Samples per symbol**. However, traces 2 and 3 contain 25 points each. The last point in trace 2, at the right border of the scope, represents the same sample as the first point in trace 1, at the left border of the scope. Similarly, the last point in trace 3 represents the same sample as the first point in trace 2. These duplicate points indicate where the traces would meet if they were displayed side by side, as illustrated in the following picture.



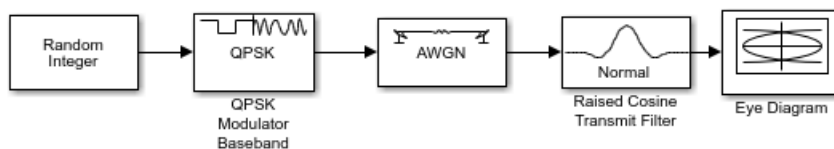
You can view a more realistic eye diagram by changing the value of **Traces displayed** to 40 and clearing the **Markers** field.



When the **Offset** parameter is set to 0, the plotting starts at the center of the first symbol, so that the open part of the eye diagram is in the middle of the plot for most points.

Constellation Diagram of a Modulated Signal

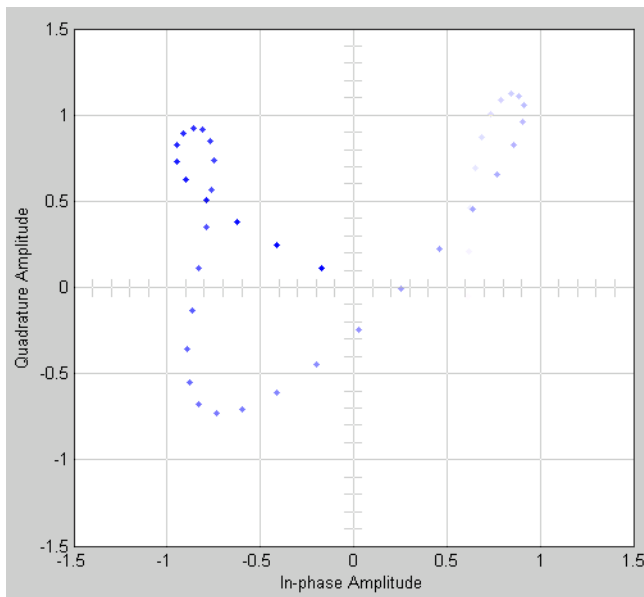
The following model creates a scatter plot of the same signal considered in “Eye Diagram of a Modulated Signal” on page 1-19.



To build the model, follow the instructions in “Eye Diagram of a Modulated Signal” on page 1-19 but replace the Eye Diagram block with the following block:

- Constellation Diagram, in the Communications Toolbox/Comm Sinks library
 - Set **Samples per symbol** to 2
 - Set **Offset** to 0. This specifies the number of samples to skip before plotting the first point.
 - Set **Symbols to display** to 40.

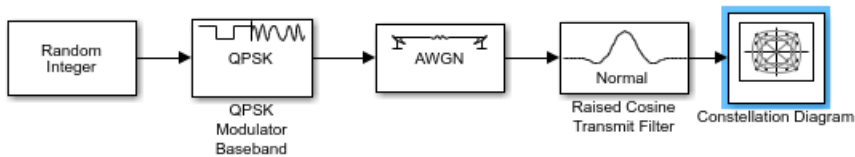
When you run the simulation, the Constellation Diagram block displays the following plot.



The plot displays 30 points. Because **Color fading** under **Rendering Properties** is selected, points are displayed less brightly the older they are.

Signal Trajectory of a Modulated Signal

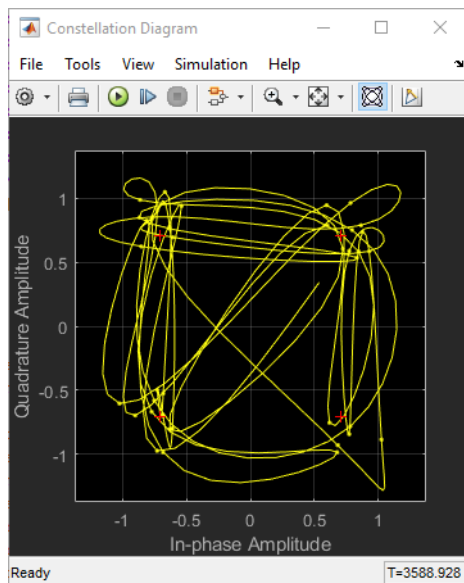
The following model creates a signal trajectory plot of the same signal considered in “Eye Diagram of a Modulated Signal” on page 1-19.



To build the model, follow the instructions in “Eye Diagram of a Modulated Signal” on page 1-19 but replace the Eye Diagram block with the following block:

- Constellation Diagram , in the Communications Toolbox/Comm Sinks library
 - Set **Samples per symbol** to 8.
 - Set **Symbols to display** to 40. This specifies the number of symbols displayed in the signal trajectory. The total number of points displayed is the product of **Samples per symbol** and **Symbols to display**.

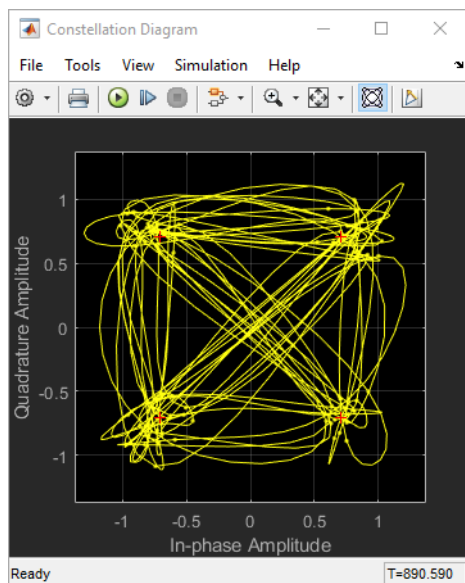
When you run the model, the Constellation Diagram displays a trajectory like the one below.



The plot displays 40 symbols. Because **Color fading** under **Rendering Properties** is selected, symbols are displayed less brightly the older they are.

See “Constellation Diagram of a Modulated Signal” on page 1-22 to compare the preceding signal trajectory to the scatter plot of the same signal. The Constellation Diagram block connects the points displayed by the Constellation Diagram block to display the signal trajectory.

If you increase **Symbols to display** to 100, the model produces a signal trajectory like the one below. The total number of points displayed at any instant is 800, which is the product of the parameters **Samples per symbol** and **Symbols to display**.



See Also

“Spreading Sequences” on page 1-26

Spreading Sequences

Spreading consists of multiplying input data bits by a pseudorandom or pseudonoise (PN) sequence. The ratio of the PN sequence bit rate to the data rate is called the *spreading factor*. When the PN sequence has a bit rate higher than the data bit rate, the spreading factor is greater than 1. When the spreading factor is greater than 1, spreading input data adds redundancy to the transmission signal.

Spreading input data by using spreading sequences with low cross-correlation properties enables the receiver to resolve individual user data after despreading the received signal. Using spreading sequences with low cross-correlation properties helps resolve individual user data in a multipath environment in the presence of interference signals.

After signal synchronization on the receiver side, the received signal is multiplied by the same PN that was used by the transmitter. This operation removes the spreading from the received signal. Ideally, after this despreading, the signal for the user of interest is recovered with no further contribution by the signals of interferers. In CDMA systems, each transmitter is assigned distinct spreading codes that have low cross-correlation properties, such as the ideal orthogonal codes or any one of the PN, Gold, or Kasami sequences.

Spread-spectrum communication systems spread the transmission signal over a wide frequency band, typically much wider than the minimum bandwidth required to transmit the data. The spreading uses a waveform that appears random to anyone except the intended receiver of the transmitted signal. The waveform is actually pseudorandom in the sense that it can be generated by precise rules, yet has the statistical properties of a truly random sequence.

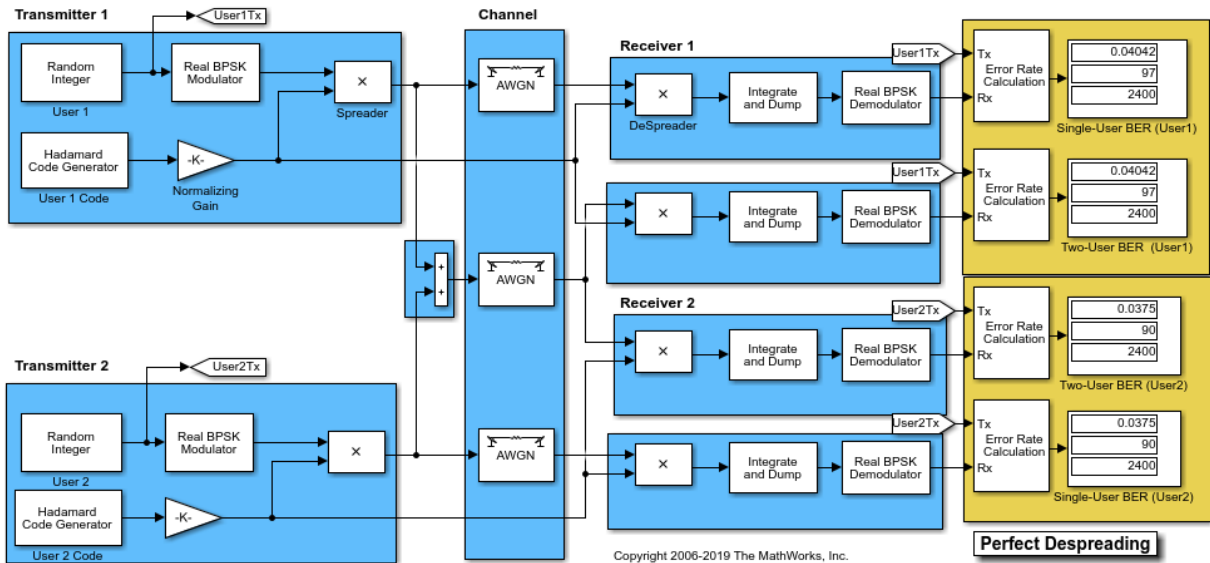
The following sections highlight various spreading sequences, their properties, and characteristic performance in single-user or multiuser and single-path or multipath transmission environments.

Orthogonal Spreading for Multiuser System in Single-Path Channel

This model compares data recovery for a single-user system versus a two-user system. Transmission data passes through a single-path AWGN channel in two data streams that are independently spread by different orthogonal codes.

The model uses random binary data, which is BPSK modulated (real), spread by orthogonal codes of length 64, and then transmitted over an AWGN channel. The receiver consists of a despreader followed by a BPSK demodulator.

Two-user Orthogonal Real Spreading with BPSK modulation



Using the same transmission data, the model calculates the BER performance for recovery of the single-user and two-user transmissions through identically configured AWGN channels.

The bit error rate results are exactly the same for the individual users in both cases. The matching error rates result from perfect despreading due to the ideal cross-correlation properties of the orthogonal codes selected.

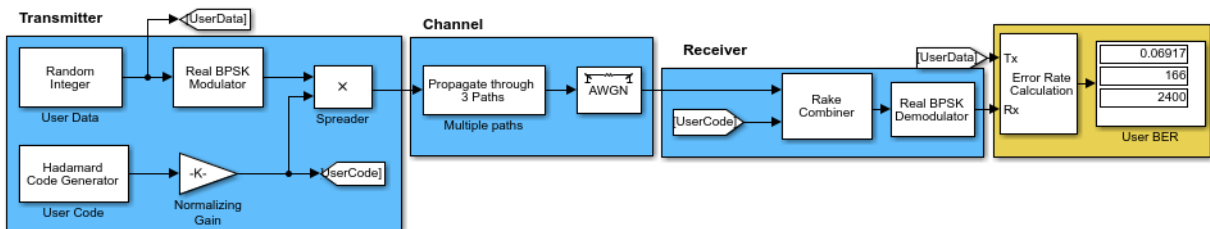
To experiment further, open the model. Modify the settings to see how the performance varies with different Hadamard codes for the individual users.

Orthogonal Spreading for Single-User System in Multipath Channel

This model simulates orthogonal spreading for a single-user system in a multipath transmission environment. This is similar to a mobile channel environment where the signals are received over multiple paths. Each path can have different amplitudes and delays. The receiver combines the independent paths coherently by using diversity reception to realize gains from the multipath transmissions received. The modeled system does not simulate fading effects and the receiver gets perfect knowledge of the number of paths and their respective delays.

The model uses random binary data, which is BPSK modulated (real), spread by orthogonal codes of length 64, and then transmitted over a multipath AWGN channel. The receiver consists of a despreader, a diversity combiner, and a BPSK demodulator.

Single-user Orthogonal Real Spreading with BPSK modulation and multiple paths



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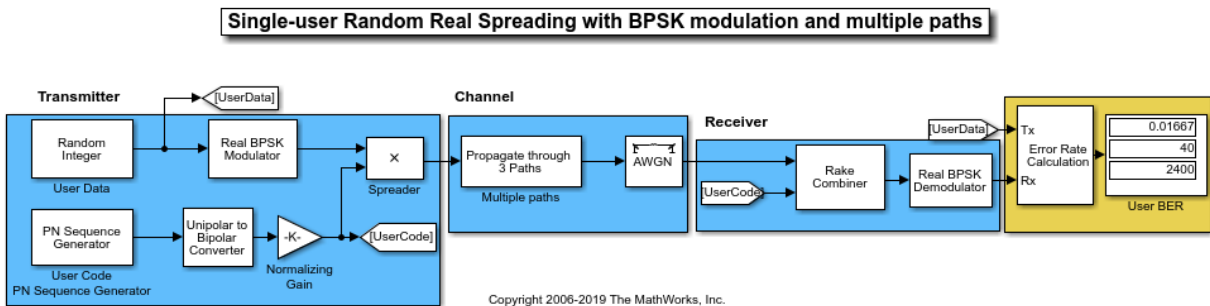
The non-ideal, auto-correlation values of the chosen orthogonal spreading codes prevent perfect resolution of the individual paths. As a consequence, BER performance is not improved by using diversity combining in the receiver. For a multipath example that uses PN sequences when spreading user data and uses diversity combining in the receiver, see “PN Spreading for Single-User System in Multipath Channel” on page 1-29.

To experiment further, open the model. Modify the settings to see how the performance varies for different path delays or with different Hadamard codes.

PN Spreading for Single-User System in Multipath Channel

This model simulates pseudo-random spreading for a single-user system in a multipath transmission environment. This is similar to a mobile channel environment where the signals are received over multiple paths. Each path can have different amplitudes and delays. The receiver combines the independent paths coherently by using diversity reception to realize gains from the multipath transmissions received. The modeled system does not simulate fading effects and the receiver gets perfect knowledge of the number of paths and their respective delays.

The model uses random binary data, which is BPSK modulated (real), spread by PN sequences, and then transmitted over a multipath AWGN channel. The receiver consists of a despreader, a diversity combiner, and a BPSK demodulator. The receiver achieves gains from diversity combining due to the ideal auto-correlation properties of the PN sequences used when spreading the data.



To experiment further, open the model. Modify the settings to see how the performance varies for different path delays or adjust the PN sequence generator parameters.

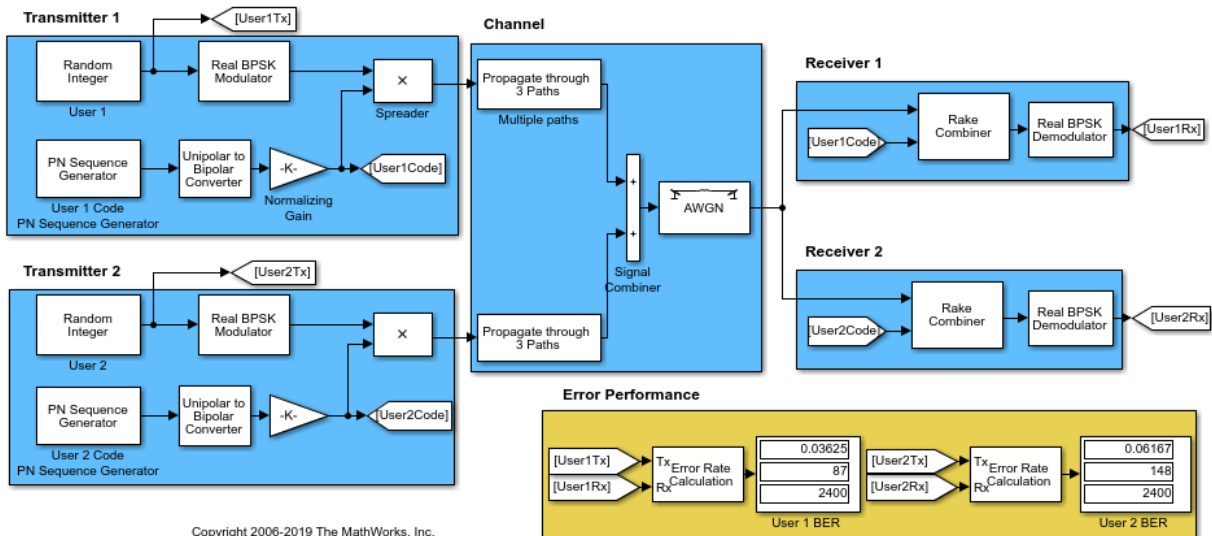
PN Spreading for Multiuser System in Multipath Channel

This model simulates pseudo-random spreading for two users in a multipath transmission environment. This is similar to a mobile channel environment where the signals are received over multiple paths. Each path can have different amplitudes and delays. The receiver combines the independent paths coherently using diversity reception to realize

gains from the multipath transmissions received. The modeled system does not simulate fading effects and the receiver gets perfect knowledge of the number of paths and their respective delays.

The model uses random binary data, which is BPSK modulated (real), spread by PN sequences, and then transmitted over a multipath AWGN channel. The receiver consists of a despreader, a diversity combiner, and a BPSK demodulator.

Two-user Random Real Spreading with BPSK modulation and multiple paths



Using the same transmission data, the model calculates the performance for two-user transmissions through identically configured, multipath AWGN channels.

Because the transmissions for the individual users were spread using different PN sequences, the error rate computed for the users are different. Due to the higher cross-correlation properties of the nonorthogonal PN sequences used to spread the data, BER performance is degraded in a multipath environment. Sequences with high orthogonality, such as Hadamard and Kasami, are a better choice for multipath environments. For a multipath example that uses Hadamard code sequences when spreading user data, see “Orthogonal Spreading for Multiuser System in Single-Path Channel” on page 1-26. For a multipath example that uses Kasami code sequences when spreading user data, see “Kasami Spreading for Multiuser System in Multipath Channel” on page 1-31.

To experiment further, open the model. Modify the settings to see how the performance varies for different path delays or with different PN sequences for the individual users.

Benefits of Diversity Combining for Nonorthogonal Sequence Spreading

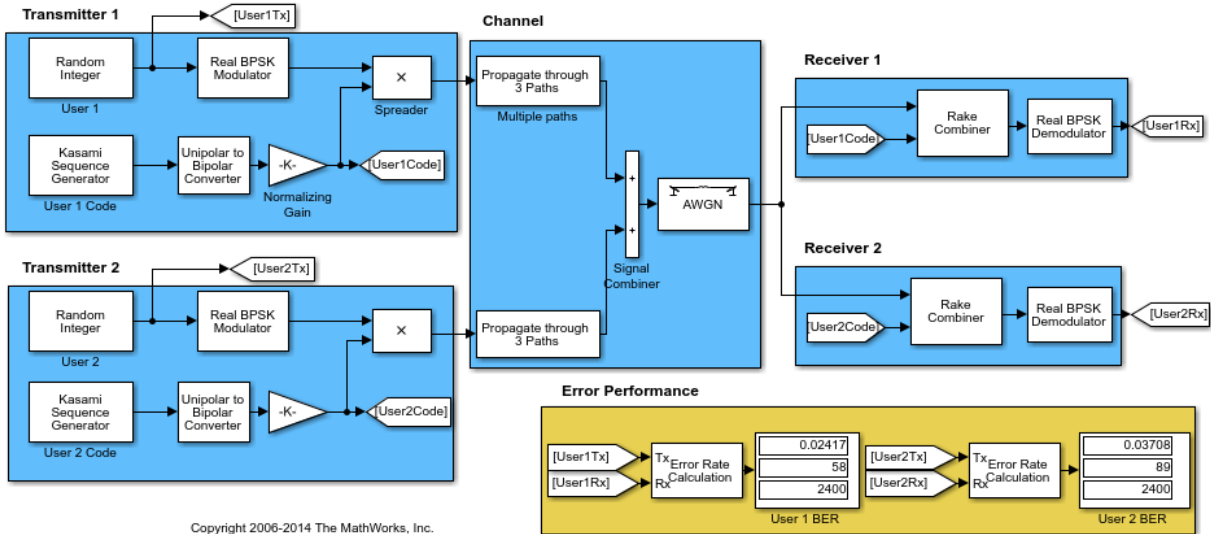
For the “PN Spreading for Multiuser System in Multipath Channel” on page 1-29 example, the individual user performance is degraded for the same channel conditions that were used in the “PN Spreading for Single-User System in Multipath Channel” on page 1-29 example. This is primarily due to the higher cross-correlation values between the two sequences, which prevent ideal separation. However, there are still advantages to diversity combining when using nonorthogonal sequence spreading, because the error rate for a multipath AWGN channel received using RAKE with diversity combining is nearly as good as the AWGN-only case in the “Orthogonal Spreading for Multiuser System in Single-Path Channel” on page 1-26 example.

Kasami Spreading for Multiuser System in Multipath Channel

This model simulates Kasami sequence spreading for two users in a multipath transmission environment. This is similar to a mobile channel environment where the signals are received over multiple paths. Each path can have different amplitudes and delays. The receiver combines the independent paths coherently using diversity reception to realize gains from the multipath transmissions received. The modeled system does not simulate fading effects and the receiver gets perfect knowledge of the number of paths and their respective delays.

The model uses random binary data, which is BPSK modulated (real), spread by Kasami sequences, and then transmitted over a multipath AWGN channel. The receiver consists of a despreader, a diversity combiner, and a BPSK demodulator.

Two-user Random Real Spreading with BPSK modulation and multiple paths



Using the same transmission data, the model calculates the performance for two-user transmissions through identically configured multipath AWGN channels.

The computed BER indicates transmission data spread using Kasami sequences exhibit low cross-correlation. The Kasami sequences provide a balance between the ideal cross-correlation properties of orthogonal codes and the ideal auto-correlation properties of PN sequences.

To experiment further, open the model. Modify the settings to see how the performance varies for different path delays or with different Kasami sequence generator settings for the individual users.

See Also

Read Signals From Hardware Devices

Communications Toolbox software can read a signal from external hardware devices using the Communications Toolbox support packages for software-defined radio (SDR). You can design, prototype and test SDR applications in MATLAB and Simulink with live radio signals. Use the supported hardware as a radio peripheral with the supplied bit stream and also run your own design in the FPGA with the automated targeting workflow using HDL Coder™.

Generate both fixed and user-provided bit streams with hardware support packages that support HDL targeting. For example, the Communications Toolbox Support Package for Xilinx® Zynq®-Based Radio and Communications Toolbox Support Package for USRP® Embedded Series Radio can be used for this purpose.

See “Supported Hardware - Software-Defined Radio” for more information on the SDR hardware support packages.

For more information about SDRs used with MATLAB and Simulink, visit Software-Defined Radio (SDR) on the MathWorks® web site.

Data and Signal Management

- “Matrices, Vectors, and Scalars” on page 2-2
- “Sample-Based and Frame-Based Processing” on page 2-4
- “Floating-Point and Fixed-Point Data Types” on page 2-5
- “Delays” on page 2-6

Matrices, Vectors, and Scalars

Simulink supports matrix signals, one-dimensional arrays, sample-based processing, and frame-based processing. This section describes how Communications Toolbox processes certain kinds of matrices and signals.

This documentation uses the unqualified words *scalar* and *vector* in ways that emphasize a signal's number of elements, not its strict dimension properties:

- A *scalar* signal contains a single element. The signal could be a one-dimensional array with one element, or a matrix of size 1-by-1.
- A *vector* signal contains one or more elements, arranged in a series. The signal could be a one-dimensional array, a matrix that has exactly one column, or a matrix that has exactly one row. The number of elements in a vector is called its *length* or, sometimes, its *width*.

In cases when it is important for a description or schematic to distinguish among different types of scalar signals or different types of vector signals, this document mentions the distinctions explicitly. For example, the terms *one-dimensional array*, *column vector*, and *row vector* distinguish among three types of vector signals.

The *size* of a matrix is the pair of numbers that indicate how many rows and columns the matrix has. The *orientation* of a two-dimensional vector is its status as either a row vector or column vector. A one-dimensional array has no orientation – this is sometimes called an unoriented vector.

A matrix signal that has more than one row and more than one column is called a *full matrix* signal.

Processing Rules

The following rules indicate how the blocks in the Communications Toolbox process scalar, vector, and matrix signals.

- In their numerical computations, blocks that process scalars do not distinguish between one-dimensional scalars and one-by-one matrices. If the block produces a scalar output from a scalar input, the block preserves dimension.
- For vector input signals:
 - The numerical computations do not distinguish between one-dimensional arrays and M-by-1 matrices.

- Most blocks do not process row vectors and do not support multichannel functionality.
- The block output preserves dimension and orientation.
- The block treats elements of the input vector as a collection that arises naturally from the block's operation (for example, a collection of symbols that jointly represent a codeword) or as successive samples from a single time series.
- Most blocks do not process matrix signals that have more than one row and more than one column. For blocks that do, a signal in the shape of an N -by- M matrix represents a series of N successive samples from M channels. An **Input processing** parameter on the block determines whether each element or column of the input signal is a channel.
- Some blocks, such as the digital baseband modulation blocks, can produce multiple output values for each value of a scalar input signal. A **Rate options** parameter on the block determines if the additional samples are output by increasing the rate of the output signal or by increasing the size of the output signal.
- Blocks that process continuous-time signals do not process frame-based inputs. Such blocks include the analog phase-locked loop blocks.

To learn which blocks processes scalar signals, vector signals, or matrices, refer to each block's individual Help page.

Sample-Based and Frame-Based Processing

In frame-based processing, blocks process data one frame at a time. Each frame of data contains sequential samples from an independent channel. For more information, see “Sample- and Frame-Based Concepts” (DSP System Toolbox).

In sample-based processing, blocks process signals one sample at a time. Each element of the input signal represents one sample of a distinct channel. For more information, see “What Is Sample-Based Processing?” (DSP System Toolbox).

Floating-Point and Fixed-Point Data Types

The inputs and outputs of the communications blocks support various data types. For some blocks, changing to single outputs can lead to different results when compared with double outputs for the same set of parameters. Some blocks may naturally output different data types than what they receive (e.g. digital modulators) a signal. Refer to the individual block reference pages for details.

For more information, see “Floating-Point Numbers” (Fixed-Point Designer) and “Fixed-Point Signal Processing” (DSP System Toolbox).

Access the Block Support Table

The Simulink Block Data Type Support for Communications Toolbox table provides details regarding capabilities and limitations pertaining to code generation, variable-sizing, and supported data types for all Communications Toolbox blocks. To access the table, type `showcommblockdatatypetable` at the MATLAB command line.

Delays

In this section...
“Section Overview” on page 2-6
“Sources of Delays” on page 2-7
“ADSL Example Model” on page 2-7
“Punctured Coding Model” on page 2-9
“Use the Find Delay Block” on page 2-12

Section Overview

Some models require you to know how long it takes for data in one portion of a model to influence a signal in another portion of a model. For example, when configuring an error rate calculator, you must indicate the delay between the transmitter and the receiver. If you miscalculate the delay, the error rate calculator processes mismatched pairs of data and consequently returns a meaningless result.

This section illustrates the computation of delays in multirate models and in models where the total delay in a sequence of blocks comprises multiple delays from individual blocks. This section also indicates how to use the Find Delay and Delay blocks to help deal with delays in a model.

Other References for Delays

Other parts of this documentation set also discuss delays. For information about dealing with delays or about delays in specific types of blocks, see

- “Group Delay” on page 16-5
- Find Delay block reference page
- Delay block reference page
- Viterbi Decoder block reference page
- Derepeat block reference page

For discussions of delays in simpler examples than the ones in this section, see

- Example: A Rate 2/3 Feedforward Encoder. on page 8-57.

- Example: Soft-Decision Decoding on page 8-62. (See Delay in Received Data on page 8-67.)
- Example: Delays from Demodulation on page 8-194.

Sources of Delays

While some blocks can determine their current output value using only the current input value, other blocks need input values from multiple time steps to compute the current output value. In the latter situation, the block incurs a delay. An example of this case is when the Derepeat block must average five samples from a scalar signal. The block must delay computing the average until it has received all five samples.

In general, delays in your model might come from various sources:

- Digital demodulators
- Convolutional interleavers or deinterleavers
- Equalizers
- Viterbi Decoder block
- Buffering, downsampling, derepeating, and similar signal operations
- Explicit delay blocks, such as Delay and Variable Integer Delay
- Filters

The following discussions include some of these sources of delay.

ADSL Example Model

This section examines the “256-Channel ADSL” example and shows how to compute the correct **Receive delay** parameter value in one of the Error Rate Calculation blocks in the model. The model includes delays from convolutional interleaving and an explicit delay block. To open the ADSL example model, enter `commads1` in the MATLAB Command Window.

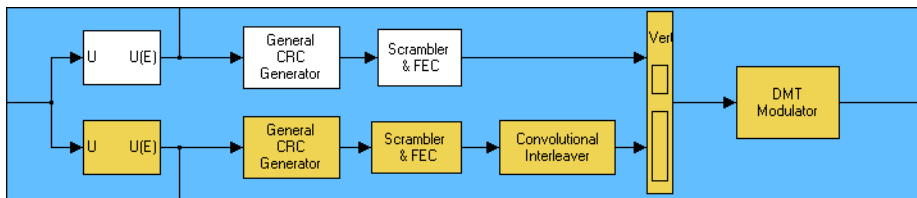
In the ADSL example, data follows one of two parallel paths, one with a nonzero delay and the other with a delay of zero. One path includes a convolutional interleaver and deinterleaver, while the other does not. Near the end of each path is an Error Rate Calculation block, whose **Receive delay** parameter must reflect the delay of the given path. The rest of the discussion makes an observation about frame periods in the model on page 2-8 and then considers the path for interleaved data on page 2-8.

Frame Periods in the Model

Before searching for individual delays, first observe that most signal lines throughout the model share the same frame period. To see this, select **Display > Sample Time**. This option colors blocks and signals according to their frame periods (or sample periods, in the case of sample-based signals). All signal lines at the top level of the model are the same color, which means that they share the same frame period. As a consequence, frames are a convenient unit for measuring delays in the blocks that process these signals. In the computation of the cumulative delay along a path, the weighted average (of numbers of frames, weighted by each frame's period) reduces to a sum.

Path for Interleaved Data

In the transmitter portion of the model, the interleaved path is the lower branch, shown in yellow below. Similarly, the interleaved path in the receiver portion of the model is the lower branch. Near the end of the interleaved path is an Error Rate Calculation block that computes the value labeled Interleaved BER.



The following table summarizes the delays in the path for noninterleaved data. Subsequent paragraphs explain the delays in more detail and explain why the total delay on page 2-9 relative to the Error Rate Calculation block is one frame, or 776 samples.

Block	Delay, in Output Samples from Individual Block	Delay, in Frames	Delay, in Input Samples to Error Rate Calculation Block
Convolutional Interleaver and Convolutional Deinterleaver pair	40	1 (combined)	776 (combined)
Delay	800		
<i>Total</i>	N/A	1	776

Interleaving

Unlike the noninterleaved path, the interleaved path contains a Convolutional Interleaver block in the transmitter and a Convolutional Deinterleaver block in the receiver. The delay of the interleaver/deinterleaver pair is the product of the **Rows of shift registers** parameter, the **Register length step** parameter, and one less than the **Rows of shift registers** parameter. In this case, the delay of the interleaver/deinterleaver pair turns out to be $5 \times 2 \times 4 = 40$ samples.

Delay Block

The receiver portion of the interleaved path also contains a Delay block. This block explicitly causes a delay of 800 samples having the same sample time as the 40 samples of delay from the interleaver/deinterleaver pair. Therefore, the total delay from interleaving, deinterleaving, and the explicit delay is 840 samples. These 840 samples make up one frame of data leaving the Delay block.

Summing the Delays

No other blocks in the interleaved path of the example cause any delays. Adding the delays from the interleaver/deinterleaver pair and the Delay block indicates that the total delay in the interleaved path is one frame.

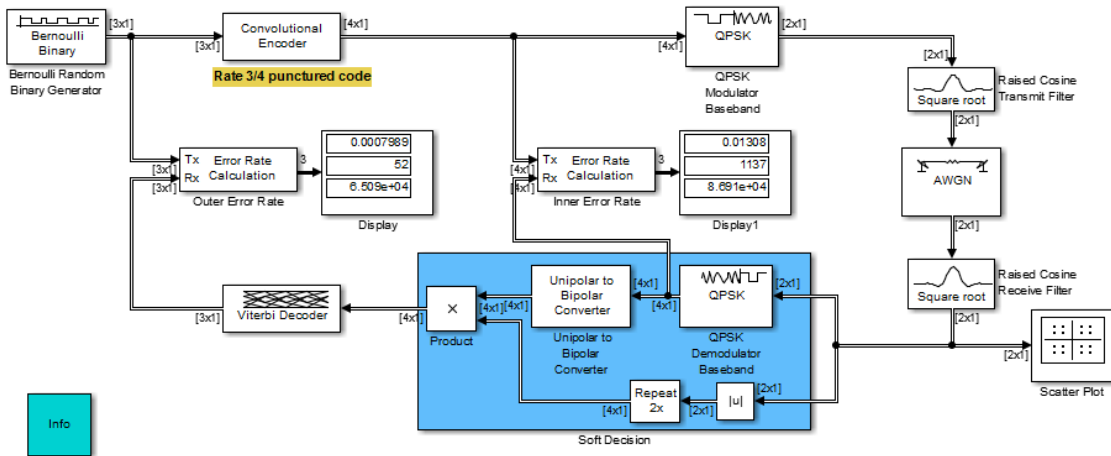
Total Delay Relative to Error Rate Calculation Block

The Error Rate Calculation block that computes the value labeled Interleaved BER requires a **Receive delay** parameter value that is equivalent to one frame. The **Receive delay** parameter is measured in samples and each input frame to the Error Rate Calculation block contains 776 samples. Also, the frame rate at the output ports of all delay-causing blocks in the interleaved path equals the frame rate at the input of the Error Rate Calculation block. Therefore, the correct value for the **Receive delay** parameter is 776 samples.

Punctured Coding Model

This section discusses a punctured coding model that includes delays from decoding, downsampling, and filtering. Two Error Rate Calculation blocks in the model work correctly if and only if their **Receive delay** parameters accurately reflect the delays in the model. To open the model, enter `doc_punct` in the MATLAB Command Window.

Punctured Coding Model



Frame Periods in the Model

Before searching for individual delays, select **Display>Sample Time>All**. Only the rightmost portion of the model differs in color from the rest of the model. This means that all signals and blocks in the model except those in the rightmost edge share the same frame period. Consequently, frames at this predominant frame rate are a convenient unit for measuring delays in the blocks that process these signals. In the computation of the cumulative delay along a path, the weighted average (of numbers of frames, weighted by each frame's period) reduces to a sum.

The yellow blocks represent multirate systems, while the AWGN Channel block runs at a higher frame rate than all the other blocks in the model.

Inner Error Rate Block

The block labeled Inner Error Rate, located near the center of the model, is a copy of the Error Rate Calculation block from the Comm Sinks library. It computes the bit error rate for the portion of the model that excludes the punctured convolutional code. In the portion of the model between this block's two input signals, delays come from the Tx Filter and the Rx Filter. This section explains why the Inner Error Rate block's **Receive delay** parameter is the total delay on page 2-11 value of 16.

Tx Filter Block

The block labeled Tx Filter is a copy of the Raised Cosine Transmit Filter block. It interpolates the input signal by a factor of 8 and applies a square-root raised cosine filter. The value of the block's **Filter span in symbols** parameter is 6, which means its group delay is 3 symbols. Since this block's sample rate increases from input port to output port, it must output an initial frame of zeros at the beginning of the simulation. Since its input frame size is 2, the block's total delay is $2 + 3 = 5$ symbols. This corresponds to 5 samples at the block's input port.

Rx Filter Block

The block labeled Rx Filter is a copy of the Raised Cosine Receive Filter block. It decimates its input signal by a factor of 8 and applies another square-root raised cosine filter. The value of this block's **Filter span in symbols** parameter is 6, which means its group delay is 3 symbols. At the block's output, the 3 symbols correspond to 3 samples.

QPSK Demodulator Block

The block labeled QPSK Demodulator Baseband receives complex QPSK signals and outputs 2 bits for each complex input. This conversion to output bits doubles the cumulative delay at the input of the block.

Summing the Delays

No other blocks in the portion of the model between the Inner Error Rate block's two input signals cause any delays. The total delay is then $(2 + 3 + 3) * 2 = 16$ samples. This value can be used as the **Receive Delay** parameter in the Inner Error Rate block.

Outer Error Rate Block

The block labeled Outer Error Rate, located at the left of the model, is a copy of the Error Rate Calculation block from the Comm Sinks library. It computes the bit error rate for the entire model, including the punctured convolutional code. Delays come from the Tx Filter, Rx Filter, and Viterbi Decoder blocks. This section explains why the Outer Error Rate block's **Receive delay** parameter is the total delay on page 2-12 value of 108.

Filter and Downsample Blocks

The Tx Filter, Rx Filter, and Downsample blocks have a combined delay of 16 samples. For details, see "Inner Error Rate Block" on page 2-10.

Viterbi Decoder Block

Because the Viterbi Decoder block decodes a rate 3/4 punctured code, it actually reduces the delay seen at its input. This reduction is given as $16 * 3/4 = 12$ samples.

The Viterbi Decoder block decodes the convolutional code, and the algorithm's use of a traceback path causes a delay. The block processes a frame-based signal and has **Operation mode** set to Continuous. Therefore, the delay, measured in output samples, is equal to the **Traceback depth** parameter value of 96. (The delay amount is stated on the reference page for the Viterbi Decoder block.) Because the output of the Viterbi Decoder block is precisely one of the inputs to the Outer Error Rate block, it is easier to consider the delay to be 96 samples rather than to convert it to an equivalent number of frames.

Total Delay Relative to Outer Error Rate Block

The Outer Error Rate block requires a **Receive delay** parameter value that is the sum of all delays in the system. This total delay is $12 + 96 = 108$ samples.

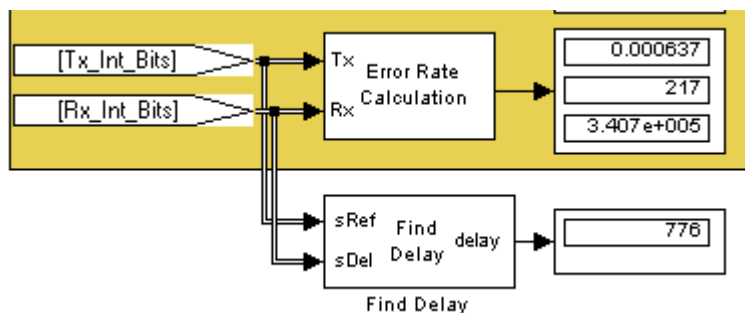
Use the Find Delay Block

The preceding discussions explained why certain Error Rate Calculation blocks in the models had specific **Receive delay** parameter values. You could have arrived at those numbers independently by using the Find Delay block. This section explains how to find the signal delay using the ADSL example model, `commads1`, as an example. Applying the technique to the punctured convolutional coding example, discussed in "Punctured Coding Model" on page 2-9, would be similar.

Using the Find Delay Block to Determine the Correct Receive Delay

Recall from "Path for Interleaved Data" on page 2-8 that the delay in the path for interleaved data is 776 samples. To have the Find Delay block compute that value for you, use this procedure:

- 1 Insert a Find Delay block and a Display block in the model near the Error Rate Calculation block that computes the value labeled Interleaved BER.
- 2 Connect the blocks as shown below.



- 3 Set the Find Delay block's **Correlation window length** parameter to a value substantially larger than 776, such as 2000.

Note You must use a sufficiently large correlation window length or else the values produced by the Find Delay block do not stabilize at a correct value.

- 4 Run the simulation.

The new Display block now shows the value 776, as expected.

Manipulate Delays

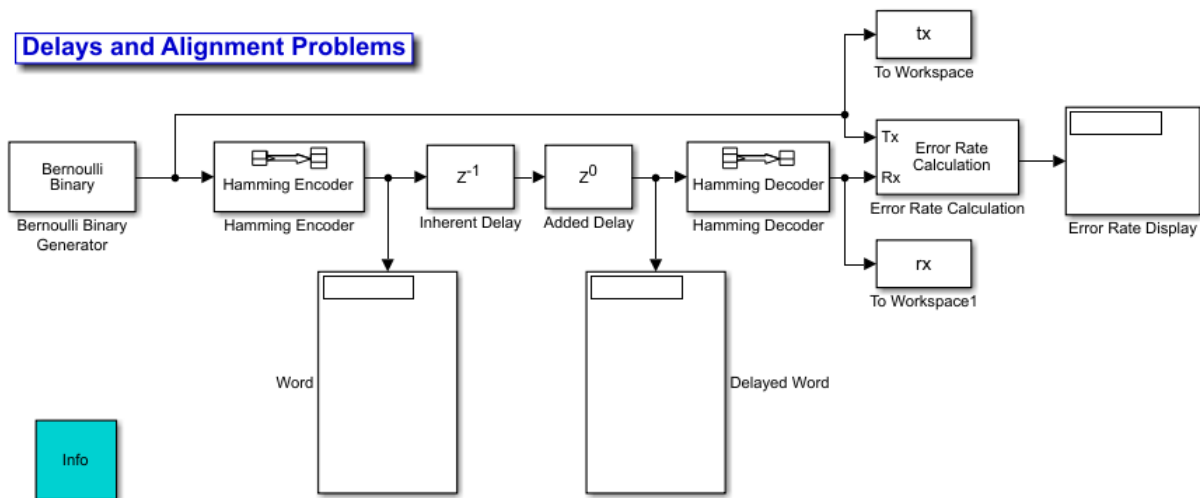
- “Delays and Alignment Problems” on page 2-13
- “Observing the Problem” on page 2-15
- “Aligning Words of a Block Code” on page 2-16
- “Aligning Words for Interleaving” on page 2-19
- “Aligning Words of a Concatenated Code” on page 2-22
- “Aligning Words for Nonlinear Digital Demodulation” on page 2-24

Delays and Alignment Problems

Some models require you not only to compute delays but to manipulate them. For example, if a model incurs a delay between a block encoder and its corresponding decoder, the decoder might misinterpret the boundaries between the codewords that it receives and, consequently, return meaningless results. More generally, such a situation can arise when the path between paired components of a block-oriented operation (such as interleaving, block coding, or bit-to-integer conversions) includes a delay-causing operation (such as those listed in “Sources of Delays” on page 2-7).

To avoid this problem, you can insert an additional delay of an appropriate amount between the encoder and decoder. If the model also computes an error rate, then the additional delay affects that process, as described in “Delays” on page 2-6. This section uses examples to illustrate the purpose, methods, and implications of manipulating delays in a variety of circumstances.

This section illustrates the sensitivity of block-oriented operations to delays, using a small model that aims to capture the essence of the problem in a simple form. Open the model by entering `doc_alignment` in the MATLAB Command Window. Then run the simulation so that the Display blocks show relevant values.



In this model, two coding blocks create and decode a block code. Two copies of the Delay block create a delay between the encoder and decoder. The two Delay blocks have different purposes in this illustrative model:

- The **Inherent Delay** block represents any delay-causing blocks that might occur in a model between the encoder and decoder. See “Sources of Delays” on page 2-7 for a list of possibilities that might occur in a more realistic model.
- The **Added Delay** block is an explicit delay that you insert to produce an appropriate amount of total delay between the encoder and decoder. For example, the `commads1` model contains a Delay block that serves this purpose.

Observing the Problem

By default, the **Delay** parameters in the Inherent Delay and Added Delay blocks are set to 1 and 0, respectively. This represents the situation in which some operation causes a one-bit delay between the encoder and decoder, but you have not yet tried to compensate for it. The total delay between the encoder and decoder is one bit. You can see from the blocks labeled Word and Delayed Word that the codeword that leaves the encoder is shifted downward by one bit by the time it enters the decoder. The decoder receives a signal in which the boundary of the codeword is at the second bit in the frame, instead of coinciding with the beginning of the frame. That is, the codewords and the frames that hold them are not aligned with each other.

This nonalignment is problematic because the Hamming Decoder block assumes that each frame begins a new codeword. As a result, it tries to decode a word that consists of the last bit of one output frame from the encoder followed by the first six bits of the next output frame from the encoder. You can see from the Error Rate Display block that the error rate from this decoding operation is close to 1/2. That is, the decoder rarely recovers the original message correctly.

To use an analogy, suppose someone corrupts a paragraph of prose by moving each period symbol from the end of the sentence to the end of the first word of the next sentence. If you try to read such a paragraph while assuming that a new sentence begins after a period, you misunderstand the start and end of each sentence. As a result, you might fail to understand the meaning of the paragraph.

To see how delays of different amounts affect the decoder's performance, vary the values of the **Delay** parameter in the Added Delay block and the **Receive delay** parameter in the Error Rate Calculation block and then run the simulation again. Many combinations of parameter values produce error rates that are close to 1/2. Furthermore, if you examine the transmitted and received data by entering

```
[tx rx]
```

in the MATLAB Command Window, you might not detect any correlation between the transmitted and received data.

Correcting the Delays

Some combinations of parameter values produce error rates of zero because the delays are appropriate for the system. For example:

- In the Added Delay block, set **Delay** to 6.

- In the Error Rate Calculation block, set **Receive delay** to 4.
- Run the simulation.
- Enter [tx rx] in the MATLAB Command Window.

The top number in the Error Rate Display block shows that the error rate is zero. The decoder recovered each transmitted message correctly. However, the Word and Displayed Word blocks do not show matching values. It is not immediately clear how the encoder's output and the decoder's input are related to each other. To clarify the matter, examine the output in the MATLAB Command Window. The sequence along the first column (tx) appears in the second column (rx) four rows later. To confirm this, enter

```
isequal(tx(1:end-4),rx(5:end))
```

in the MATLAB Command Window and observe that the result is 1 (true). This last command tests whether the first column matches a shifted version of the second column. Shifting the MATLAB vector rx by four rows corresponds to the Error Rate Calculation block's behavior when its **Receive delay** parameter is set to 4.

To summarize, these special values of the **Delay** and **Receive delay** parameters work for these reasons:

- Combined, the Inherent Delay and Added Delay blocks delay the encoded signal by a full codeword rather than by a partial codeword. Thus the decoder is correct in its assumption that a codeword boundary falls at the beginning of an input frame and decodes the words correctly. However, the delay in the encoded signal causes each recovered message to appear one word later, that is, four bits later.
- The Error Rate Calculation block compensates for the one-word delay in the system by comparing each word of the transmitted signal with the data four bits later in the received signal. In this way, it correctly concludes that the decoder's error rate is zero.

Note These are not the only parameter values that produce error rates of zero. Because the code in this model is a (7, 4) block code and the inherent delay value is 1, you can set the **Delay** and **Receive delay** parameters to $7k-1$ and $4k$, respectively, for any positive integer k . It is important that the sum of the inherent delay (1) and the added delay ($7k-1$) is a multiple of the codeword length (7).

Aligning Words of a Block Code

The ADSL example, discussed in “ADSL Example Model” on page 2-7, illustrates the need to manipulate the delay in a model so that each frame of data that enters a block decoder

has a codeword boundary at the beginning of the frame. The need arises because the path between a block encoder and block decoder includes a delay-causing convolutional interleaving operation. This section explains why the model uses a Delay block to manipulate the delay between the convolutional deinterleaver and the block decoder, and why the Delay block is configured as it is. To open the ADSL example model, enter `commads1` in the MATLAB Command Window.

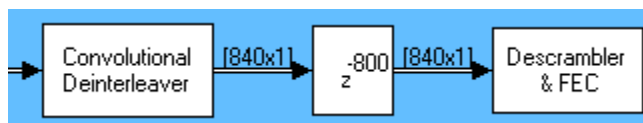
Misalignment of Codewords

In the ADSL example, the Convolutional Interleaver and Convolutional Deinterleaver blocks appear after the Scrambler & FEC subsystems but before the Descrambler & FEC subsystems. These two subsystems contain blocks that perform Reed-Solomon coding, and the coding blocks expect each frame of input data to start on a new word rather than in the middle of a word.

As discussed in “Path for Interleaved Data” on page 2-8, the delay of the interleaver/deinterleaver pair is 40 samples. However, the input to the Descrambler & FEC subsystem is a frame of size 840, and 40 is not a multiple of 840. Consequently, the signal that exits the Convolutional Deinterleaver block is a frame whose first entry does *not* represent the beginning of a new codeword. As described in “Observing the Problem” on page 2-15, this misalignment, between codewords and the frames that contain them, prevents the decoder from decoding correctly.

Inserting a Delay to Correct the Alignment

The ADSL example solves the problem by moving the word boundary from the 41st sample of the 840-sample frame to the first sample of a successive frame. Moving the word boundary is equivalent to delaying the signal. To this end, the example contains a Delay block between the Convolutional Deinterleaver block and the Descrambler & FEC subsystem.



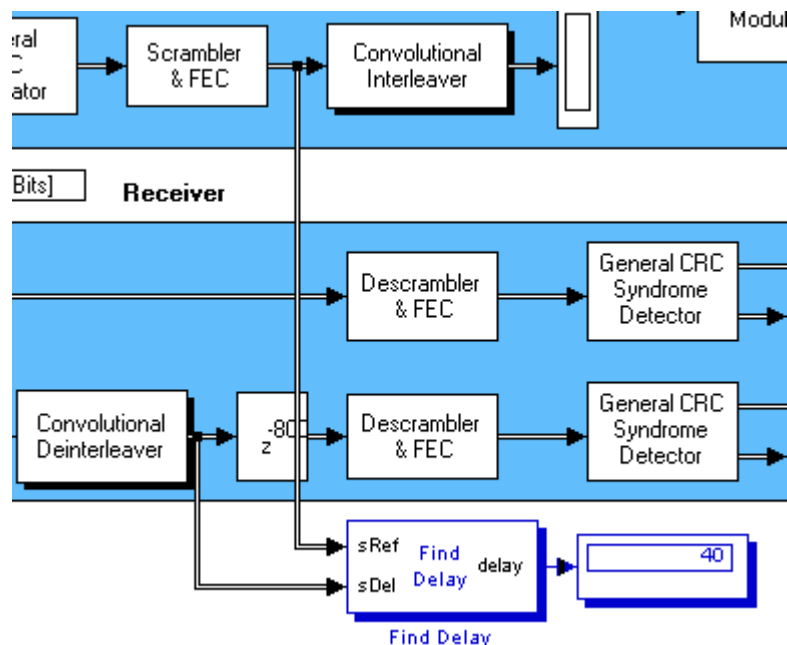
The **Delay** parameter in the Delay block is 800 because that is the minimum number of samples required to shift the 41st sample of one 840-sample frame to the first sample of the next 840-sample frame. In other words, the sum of the inherent 40-sample delay (from the interleaving/deinterleaving process) and the artificial 800-sample delay is a full frame of data, not a partial frame.

This 800-sample delay has implications for other parts of the model, specifically, the **Receive delay** parameter in one of the Error Rate Calculation blocks. For details about how the delay influences the value of that parameter, see “Path for Interleaved Data” on page 2-8.

Using the Find Delay Block

The preceding discussion explained why an 800-sample delay is necessary to correct the misalignment between codewords and the frames that contain them. Knowing that the Descrambler & FEC subsystem requires frame boundaries to occur on word boundaries, you could have arrived at the number 800 independently by using the Find Delay block. Use this procedure:

- 1 Insert a Find Delay block and a Display block in the model.
- 2 Create a branch line that connects the input of the Convolutional Interleaver block to the `sRef` input of the Find Delay block.
- 3 Create another branch line that connects the output of the Convolutional Deinterleaver block to the `sDel` input of the Find Delay block.
- 4 Connect the `delay` output of the Find Delay block to the new Display block. The modified part of the model now looks like the following image (which also shows drop shadows on key blocks to emphasize the modifications).



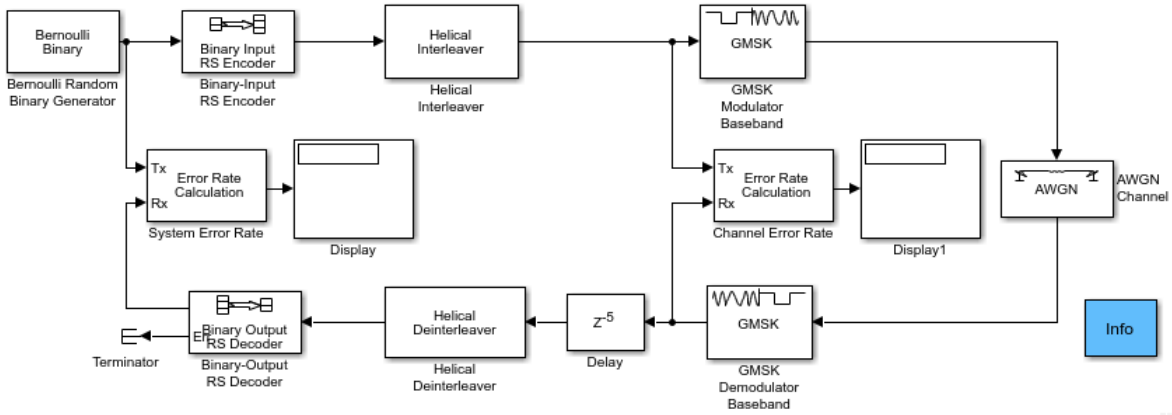
- 5 Show the dimensions of each signal in the model by selecting enabling **Display > Signals & Ports > Signal Dimensions**.
- 6 Run the simulation.

The new Display block now shows the value 40. Also, the display of signal dimensions shows that the output from the Convolutional Deinterleaver block is a frame of length 840. These results indicate that the sequence of blocks between the Convolutional Interleaver and Convolutional Deinterleaver, inclusive, delays an 840-sample frame by 40 samples. An additional delay of 800 samples brings the total delay to 840. Because the total delay is now a multiple of the frame length, the delayed deinterleaved data can be decoded.

Aligning Words for Interleaving

This section describes an example that manipulates the delay before a deinterleaver, because the path between the interleaver and deinterleaver includes a delay from demodulation. To open the model, enter `doc_gmskint` in the MATLAB Command Window.

Aligning Words for Interleaving



The model includes block coding, helical interleaving, and GMSK modulation. The table below summarizes the individual block delays in the model.

Block	Delay, in Output Samples from Individual Block	Reference
GMSK Demodulator Baseband	16	“Delays in Digital Modulation” on page 8-193
Helical Deinterleaver	42	“Delays of Convolutional Interleavers” on page 8-162
Delay	5	Delay reference page

Misalignment of Interleaved Words

The demodulation process in this model causes a delay between the interleaver and deinterleaver. Because the deinterleaver expects each frame of input data to start on a new word, it is important to ensure that the total delay between the interleaver and deinterleaver includes one or more full frames but no partial frames.

The delay of the demodulator is 16 output samples. However, the input to the Helical Deinterleaver block is a frame of size 21, and 16 is not a multiple of 21. Consequently, the signal that exits the GMSK Demodulator Baseband block is a frame whose first entry does *not* represent the beginning of a new word. As described in “Observing the Problem” on

page 2-15, this misalignment between words and the frames that contain them hinders the deinterleaver.

Inserting a Delay to Correct the Alignment

The model moves the word boundary from the 17th sample of the 21-sample frame to the first sample of the next frame. Moving the word boundary is equivalent to delaying the signal by five samples. The Delay block between the GMSK Demodulator Baseband block and the Helical Deinterleaver block accomplishes such a delay. The Delay block has its **Delay** parameter set to 5.

Combining the effects of the demodulator and the Delay block, the total delay between the interleaver and deinterleaver is a full 21-sample frame of data, not a partial frame.

Checking Alignment of Block Codewords

The interleaver and deinterleaver cause a combined delay of 42 samples measured at the output from the Helical Deinterleaver block. Because the delayed output from the deinterleaver goes next to a Reed-Solomon decoder, and because the decoder expects each frame of input data to start on a new word, it is important to ensure that the total delay between the encoder and decoder includes one or more full frames but no partial frames.

In this case, the 42-sample delay is exactly two frames. Therefore, it is not necessary to insert a Delay block between the Helical Deinterleaver block and the Binary-Output RS Decoder block.

Computing Delays to Configure the Error Rate Calculation Blocks

The model contains two Error Rate Calculation blocks, labeled Channel Error Rate and System Error Rate. Each of these blocks has a **Receive delay** parameter that must reflect the delay of the path between the block's Tx and Rx signals. The following table explains the **Receive delay** values in the two blocks.

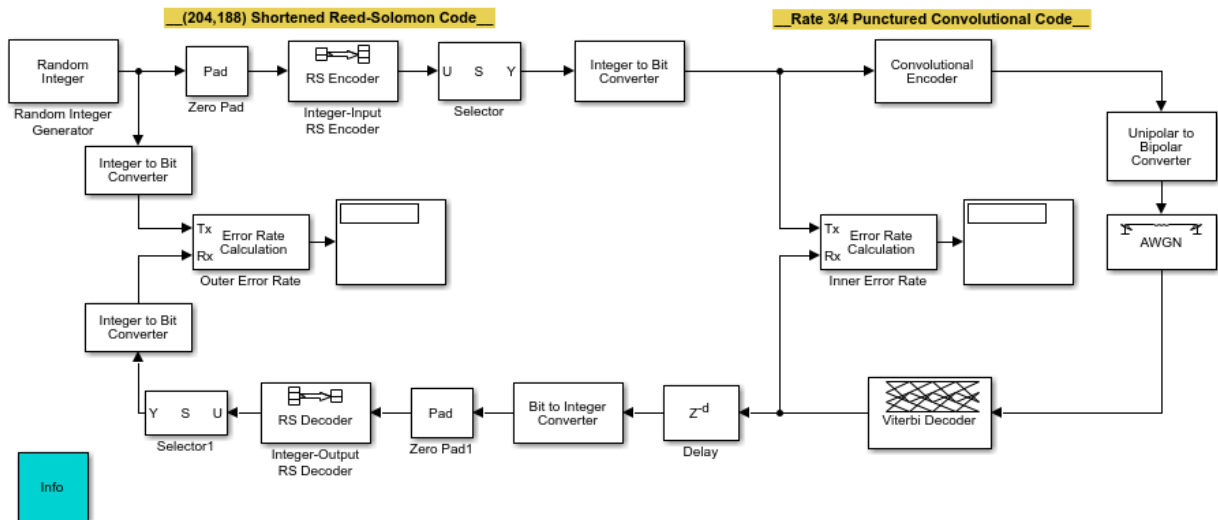
Block	Receive Delay Value	Reason
Channel Error Rate	16	Delay of GMSK Demodulator Baseband block, in samples

Block	Receive Delay Value	Reason
System Error Rate	15*3	Three fifteen-sample frames: one frame from the GMSK Demodulator Baseband and Delay blocks, and two frames from the interleaver/deinterleaver pair

Aligning Words of a Concatenated Code

This section describes an example that manipulates the delay between the two portions of a concatenated code decoder, because the first portion includes a delay from Viterbi decoding while the second portion expects frame boundaries to coincide with word boundaries. To open the model, enter `doc_concat` in the MATLAB Command Window. It uses the block and convolutional codes from the `commdvbt` example, but simplifies the overall design a great deal.

Aligning Words of a Concatenated Code



The model includes a shortened block code and a punctured convolutional code. All signals and blocks in the model share the same frame period. The following table summarizes the individual block delays in the model.

Block	Delay, in Output Samples from Individual Block
Viterbi Decoder	136
Delay	1496 (that is, 1632 - 136)

Misalignment of Block Codewords

The Viterbi decoding process in this model causes a delay between the Integer to Bit Converter block and the Bit to Integer Converter block. Because the latter block expects each frame of input data to start on a new 8-bit word, it is important to ensure that the total delay between the two converter blocks includes one or more full frames but no partial frames.

The delay of the Viterbi Decoder block is 136 output samples. However, the input to the Bit to Integer Converter block is a frame of size 1632. Consequently, the signal that exits the Viterbi Decoder block is a frame whose first entry does *not* represent the beginning of a new word. As described in “Observing the Problem” on page 2-15, this misalignment between words and the frames that contain them hinders the converter block.

Note The outer decoder in this model (Integer-Output RS Decoder) also expects each frame of input data to start on a new codeword. Therefore, the misalignment issue in this model affects many concatenated code designs, not just those that convert between binary-valued and integer-valued signals.

Inserting a Delay to Correct the Alignment

The model moves the word boundary from the 137th sample of the 1632-sample frame to the first sample of the next frame. Moving the word boundary is equivalent to delaying the signal by 1632-136 samples. The Delay block between the Viterbi Decoder block and the Bit to Integer Converter block accomplishes such a delay. The Delay block has its **Delay** parameter set to 1496.

Combining the effects of the Viterbi Decoder block and the Delay block, the total delay between the interleaver and deinterleaver is a full 1632-sample frame of data, not a partial frame.

Computing Delays to Configure the Error Rate Calculation Blocks

The model contains two Error Rate Calculation blocks, labeled Inner Error Rate and Outer Error Rate. Each of these blocks has a **Receive delay** parameter that must reflect

the delay of the path between the block's Tx and Rx signals. The table below explains the **Receive delay** values in the two blocks.

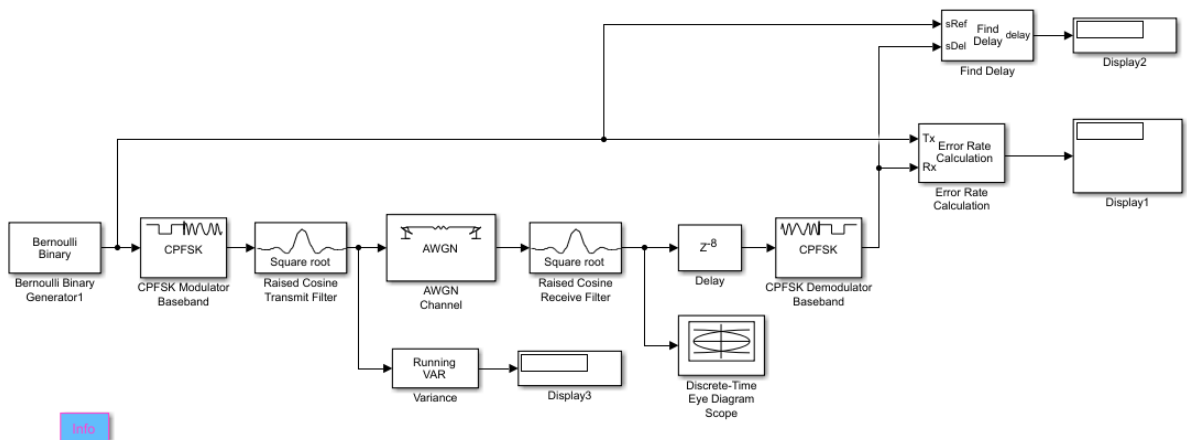
Block	Receive Delay Value	Reason
Inner Error Rate	136	Delay of Viterbi Decoder block, in samples
Outer Error Rate	1504 (188*8 bits)	One 188-sample frame, from the combination of the inherent delay of the Viterbi Decoder block and the added delay of the Delay block

Aligning Words for Nonlinear Digital Demodulation

This example manipulates delay in order obtain the correct symbol synchronization of a signal so that symbol boundaries correctly align before demodulation occurs.

To open this model, type `doc_nonlinear_digital_demod` at the MATLAB command line.

Aligning Words for Nonlinear Digital Demodulation



This model includes a CPFSK modulation scheme and pulse shaping filter. For the demodulation to work properly, the input signal to the CPFSK demodulator block must have the correct alignment. Various blocks in this model introduce processing delays.

Because of these delays, the input signal to the CPFSK demodulator block is not in the correct alignment.

Both the Raised Cosine Transmit and Receive Filter blocks introduce a delay. The delay is defined as: $GroupDelay \cdot Ts$

where Ts represents the input sample time of the Raised Cosine Transmit Filter block.

The input sample time of the Raised Cosine Transmit Filter block equals the output sample time of the Raised Cosine Receive Filter block. Therefore, the total delay at the output of the Raised Cosine Receive Filter is:

$$2 \cdot GroupDelay \cdot Ts$$

$$\text{or } 8 \cdot Ts$$

$$\text{as } GroupDelay = 4$$

The CPFSK demodulator block receives this delayed signal, and then it processes each collection of 8 samples per symbol to compute 1 output symbol. You must ensure that the CPFSK demodulator receives input samples in the correct collection of samples. For binary CPFSK with a **Modulation index** of 1/2, the demodulator input must align along even numbers of symbols. Note that this requirement applies only to binary CPFSK with a modulation index of 1/2. Other CPM schemes with different M-ary values and modulation indexes have different requirements.

To ensure that the CPFSK demodulator in this model receives the correct collection of input samples with the correct alignment, introduce a delay of 8 samples (in this example, $8 \cdot Ts$). The total delay at the input of the CPFSK demodulator is $16 \cdot Ts$, which equates to two symbol delays ($2 \cdot T$, where T is the symbol period).

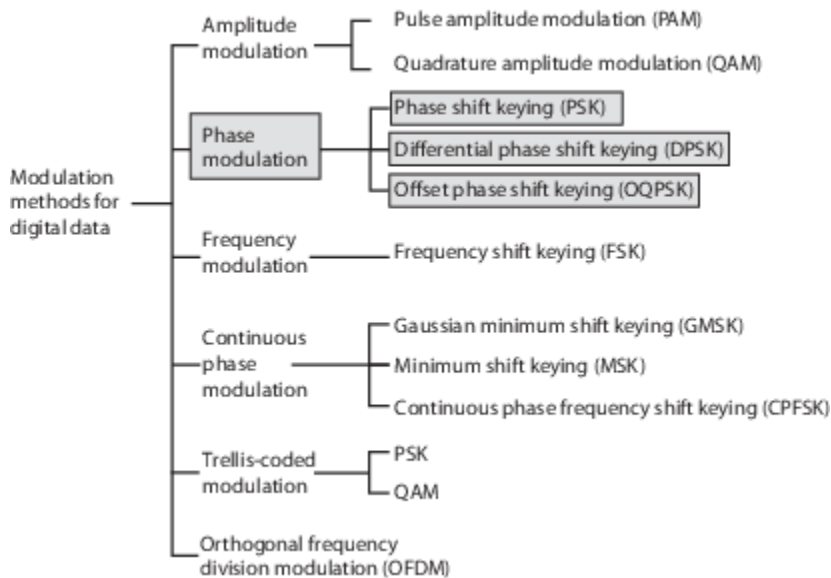
In sample-based mode, the CPFSK demodulator introduces a delay of **Traceback length** + 1 samples at its output. In this example, **Traceback length** equals 16. Therefore, the total **Receiver delay** in the Error rate calculation block equals 17+2 or 19. For more information, see “Delays in Digital Modulation” on page 8-193 in the Communications Toolbox *User's Guide*.

Digital Modulation

Phase Modulation

Phase modulation is a linear baseband modulation technique in which the message modulates the phase of a constant amplitude signal. Communications Toolbox provides modulators and demodulators for these phase modulation techniques:

- Phase shift keying (PSK) — Binary, quadrature, and general PSK
- Differential phase shift keying (DPSK) — Binary, quadrature, and general DPSK
- Offset QPSK (OQPSK)



To modulate input data with these techniques, you can use MATLAB functions, System objects, or Simulink blocks.

Modulation Scheme	MATLAB functions	System objects	Simulink blocks
Binary PSK (BPSK)		<ul style="list-style-type: none"> • comm.BPSKModulator • comm.BPSKDemodulator 	<ul style="list-style-type: none"> • BPSK Modulator Baseband • BPSK Demodulator Baseband
Quadrature PSK (QPSK)		<ul style="list-style-type: none"> • comm.QPSKModulator • comm.QPSKDemodulator 	<ul style="list-style-type: none"> • QPSK Modulator Baseband • QPSK Demodulator Baseband
General PSK	<ul style="list-style-type: none"> • pskmod • pskdemod 	<ul style="list-style-type: none"> • comm.PSKModulator • comm.PSKDemodulator 	<ul style="list-style-type: none"> • M-PSK Modulator Baseband • M-PSK Demodulator Baseband
Differential BPSK (DBPSK)		<ul style="list-style-type: none"> • comm.DBPSKModulator • comm.DBPSKDemodulator 	<ul style="list-style-type: none"> • DBPSK Modulator Baseband • DBPSK Demodulator Baseband
Differential QPSK (DQPSK)		<ul style="list-style-type: none"> • comm.DQPSKModulator • comm.DQPSKDemodulator 	<ul style="list-style-type: none"> • DQPSK Modulator Baseband • DQPSK Demodulator Baseband
General DPSK	<ul style="list-style-type: none"> • dpskmod • dpskdemod 	<ul style="list-style-type: none"> • comm.DPSKModulator • comm.DPSKDemodulator 	<ul style="list-style-type: none"> • M-DPSK Modulator Baseband • M-DPSK Demodulator Baseband
OQPSK		<ul style="list-style-type: none"> • comm.OQPSKModulator • comm.OQPSKDemodulator 	<ul style="list-style-type: none"> • OQPSK Modulator Baseband • OQPSK Demodulator Baseband

Baseband and Passband Simulation

Communications Toolbox supports *baseband* and *passband* simulation methods; however, the phase shift keying techniques support baseband simulation only.

A general passband waveform can be represented as

$$Y_1(t)\cos(2\pi f_c t + \theta) - Y_2(t)\sin(2\pi f_c t + \theta),$$

where f_c is the carrier frequency and θ is the initial phase of the carrier signal. This equation is equal to the real part of

$$[(Y_1(t) + jY_2(t))e^{j\theta}]\exp(j2\pi f_c t).$$

In a baseband simulation, only the expression within the square brackets is modeled. The vector y is a sampling of the complex signal

$$(Y_1(t) + jY_2(t))e^{j\theta}.$$

BPSK

In binary phase shift keying (BPSK), the phase of a constant amplitude signal switches between two values corresponding to binary 1 and binary 0. The passband waveform of a BPSK signal is

$$s_n(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t + \pi(1 - n)); n \in \{0, 1\},$$

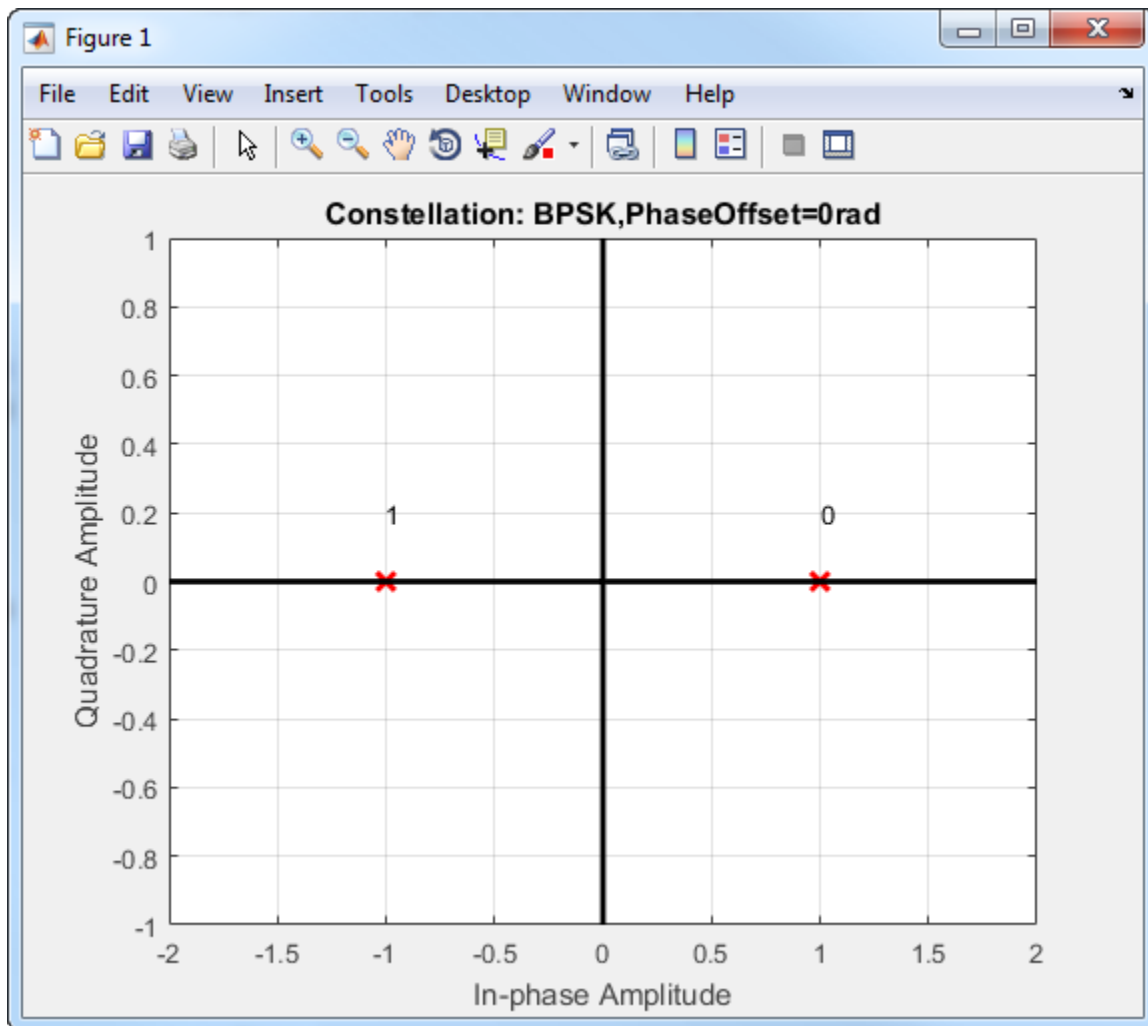
where:

- E_b is the energy per bit.
- T_b is the bit duration.
- f_c is the carrier frequency.

In MATLAB, the baseband representation of a BPSK signal is

$$s_n(t) = \cos(\pi n); n \in \{0, 1\}.$$

The BPSK signal has two phases: 0 and π .



The probability of a bit error in an AWGN channel is

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right),$$

where N_0 is the noise power spectral density.

QPSK

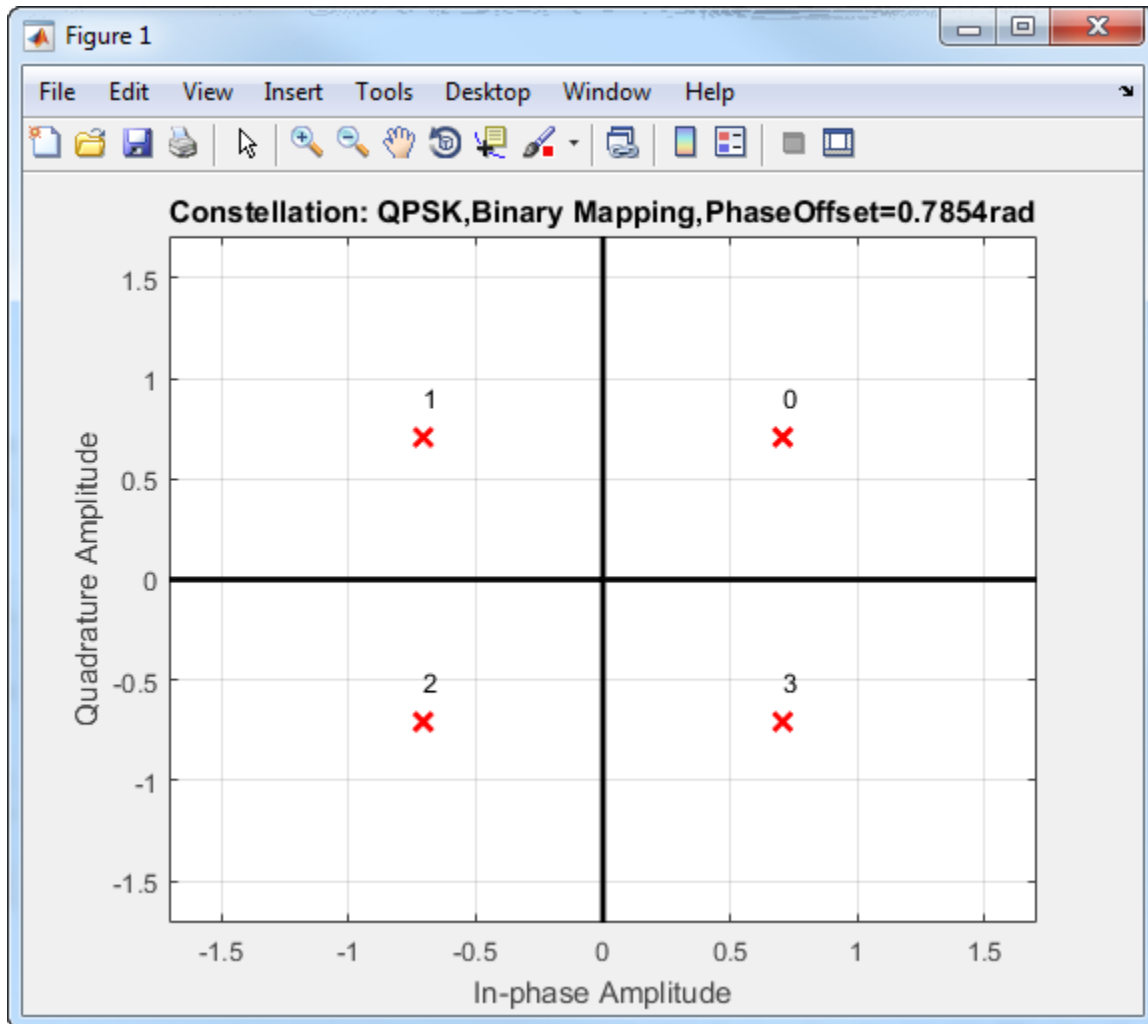
In quadrature phase shift keying, the message bits are grouped into 2-bit symbols, which are transmitted as one of four phases of a constant amplitude baseband signal. This grouping provides a bandwidth efficiency that is twice as great as the efficiency of BPSK. The general QPSK signal is expressed as

$$s_n(t) = \sqrt{\frac{2E_s}{T_s}} \cos\left(2\pi f_c t + (2n + 1)\frac{\pi}{4}\right); \quad n \in \{0, 1, 2, 3\},$$

where E_s is the energy per symbol and T_s is the symbol duration. The complex baseband representation of a QPSK signal is

$$s_n(t) = \exp\left(j\pi\left(\frac{2n + 1}{4}\right)\right); \quad n \in \{0, 1, 2, 3\}.$$

In this QPSK constellation diagram, each 2-bit sequence is mapped to one of four possible states. The states correspond to phases of $\pi/4$, $3\pi/4$, $5\pi/4$, and $7\pi/4$.

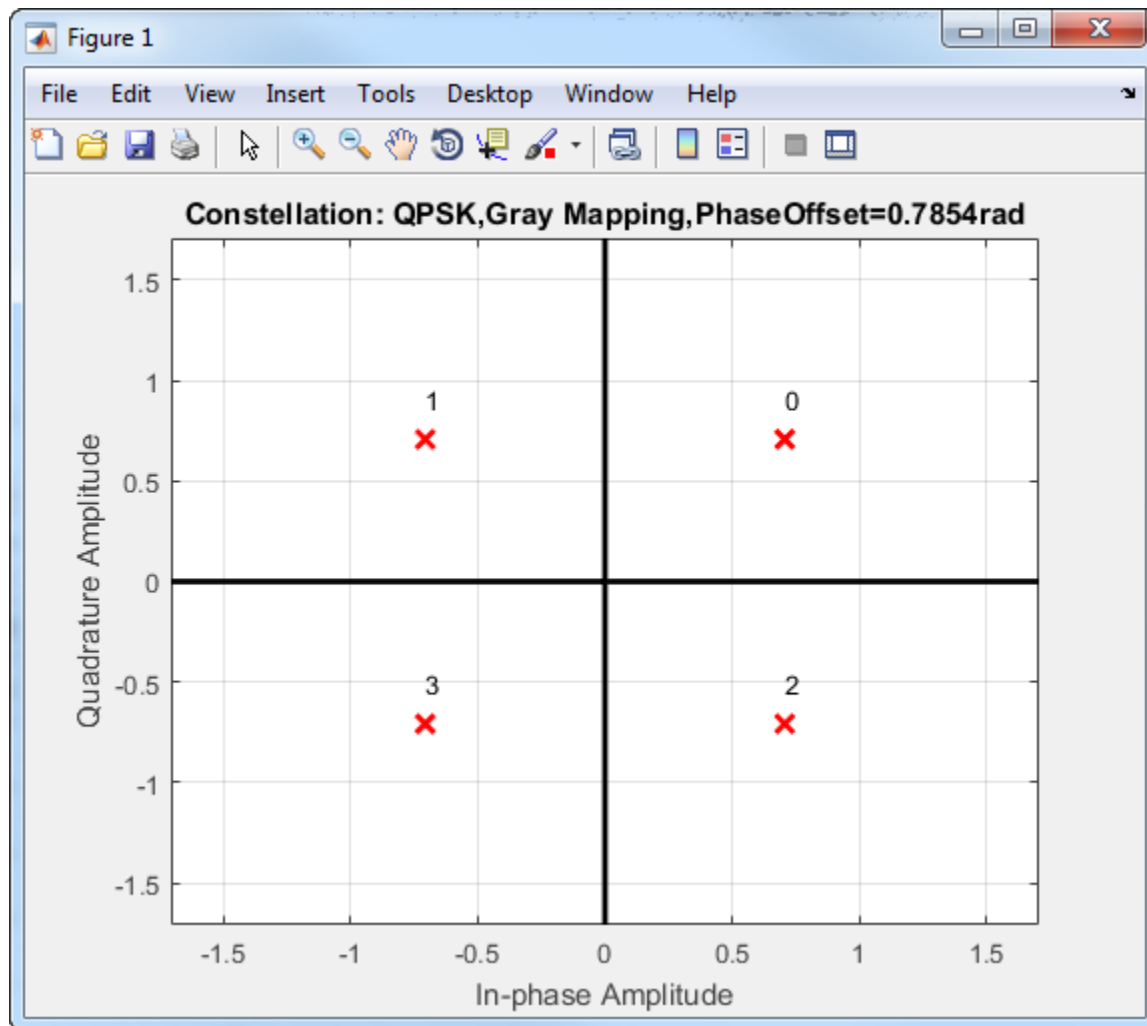


To improve bit error rate performance, the incoming bits can be mapped to a Gray-coded ordering.

Binary-to-Gray Mapping

Binary Sequence	Gray-coded Sequence
00	00
01	01
10	11
11	10

The primary advantage of the Gray code is that only one of the two bits changes when moving between adjacent constellation points. Gray codes can be applied to higher-order modulations, as shown in this Gray-coded QPSK constellation.



The bit error probability for QPSK in AWGN with Gray coding is

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right),$$

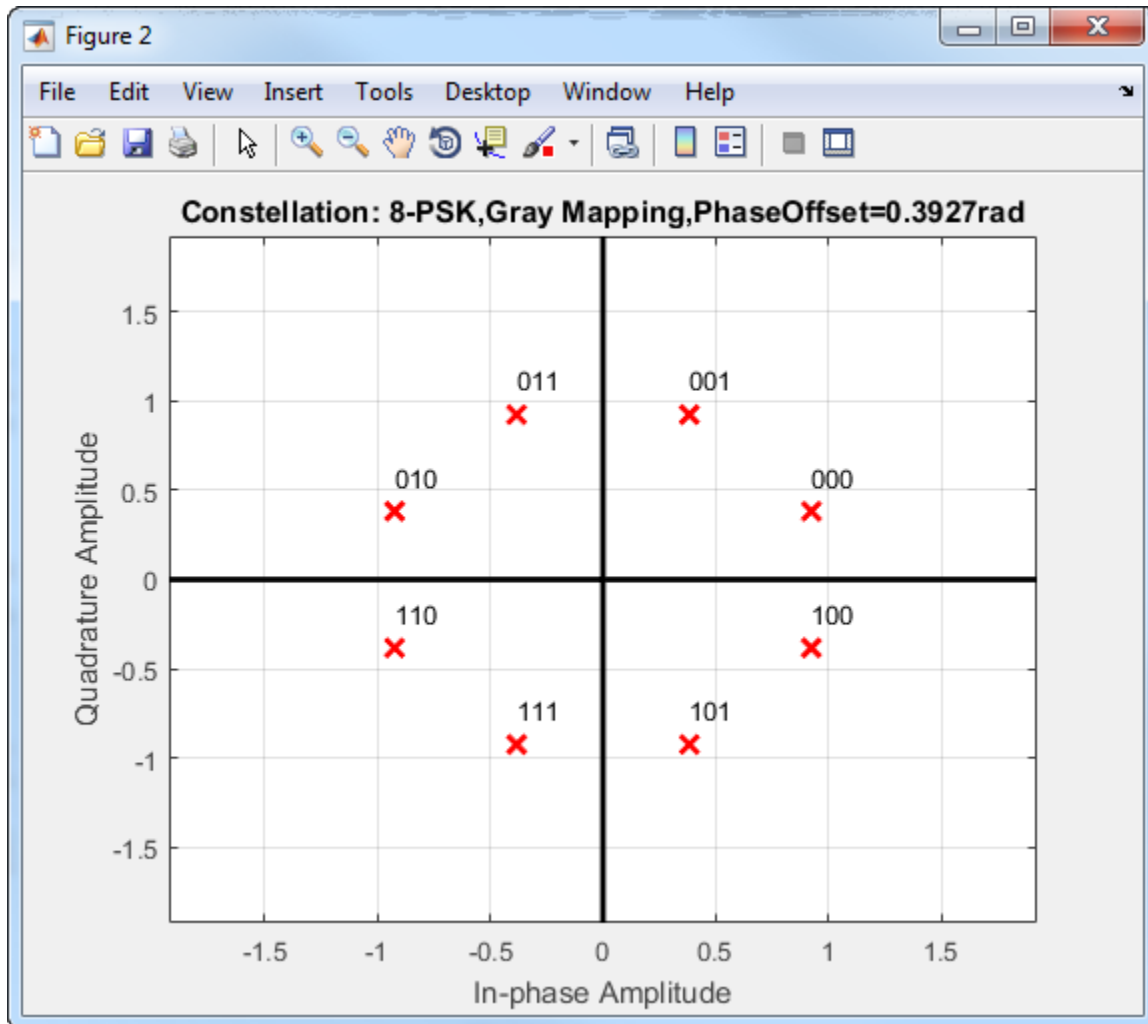
which is the same as the expression for BPSK. As a result, QPSK provides the same performance with twice the bandwidth efficiency.

Higher-Order PSK

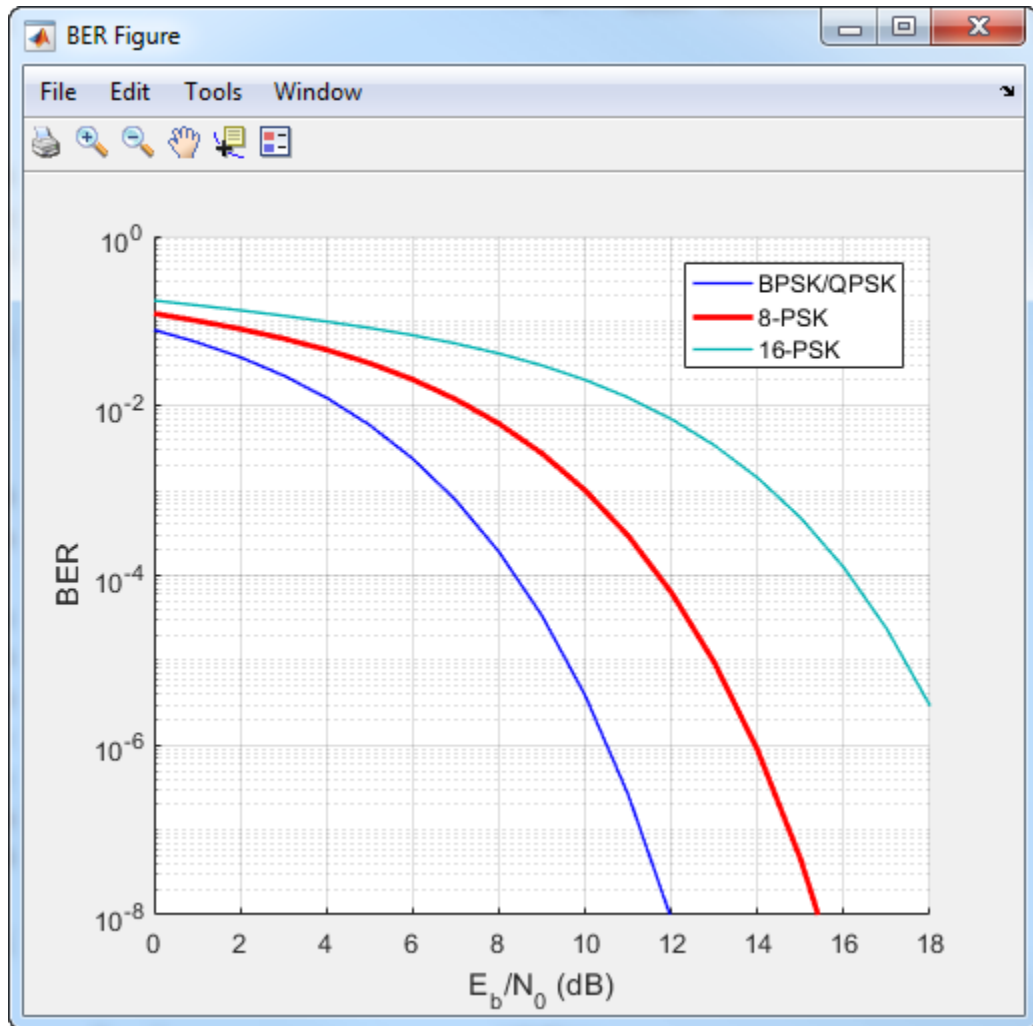
In MATLAB, you can modulate and demodulate higher-order PSK constellations. The complex baseband form for an M-ary PSK signal using natural binary-ordered symbol mapping is

$$s_n(t) = \exp\left(j\pi\left(\frac{2n+1}{M}\right)\right); \quad n \in \{0, 1, \dots, M-1\}.$$

This 8-PSK constellation uses Gray-coded symbol mapping.



For modulation orders beyond 4, the bit error rate performance of PSK in AWGN worsens. In the following figure, the QPSK and BPSK curves overlap one another.



DPSK

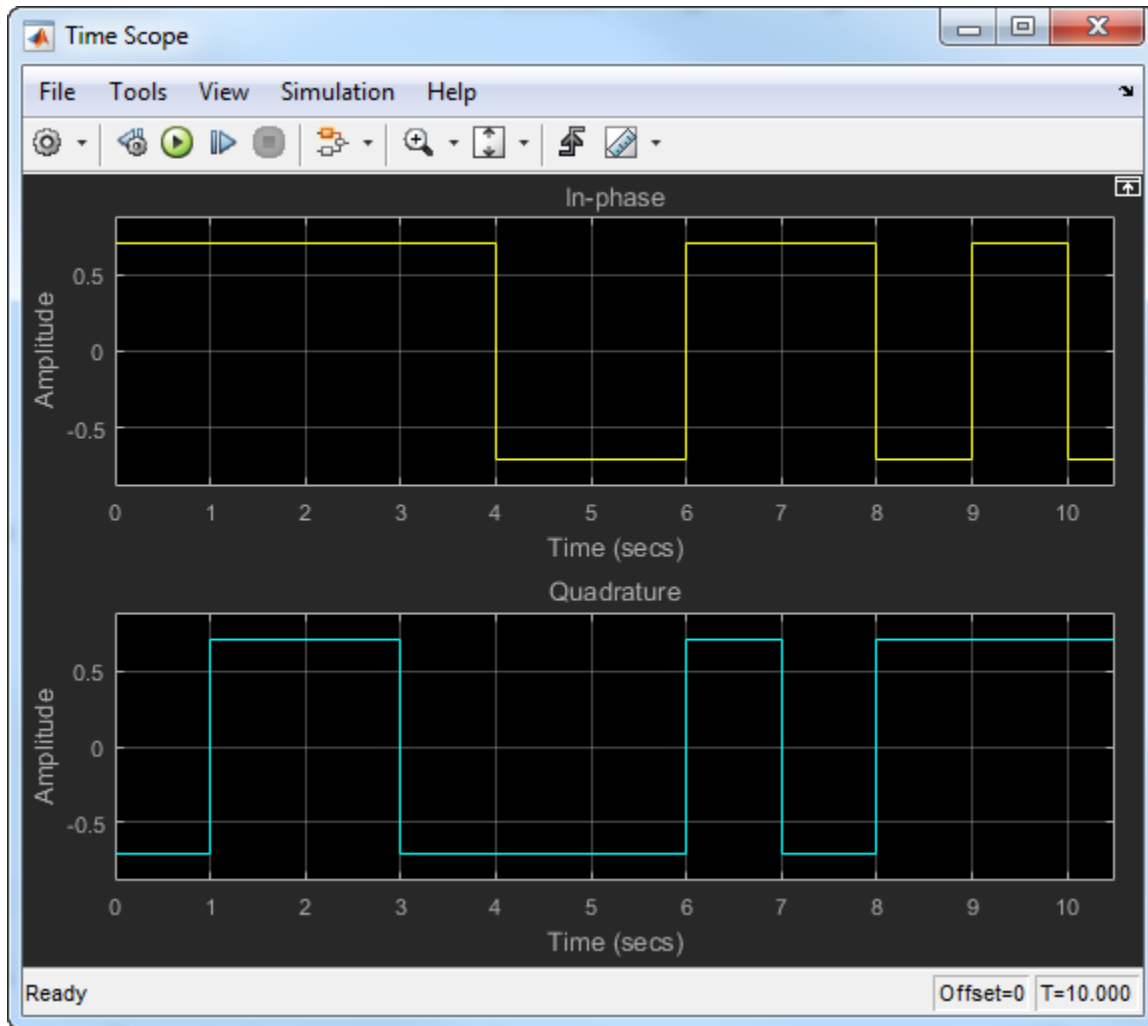
DPSK is a noncoherent form of phase shift keying that does not require a coherent reference signal at the receiver. With DPSK, the difference between successive input symbols is mapped to a specific phase. As an example, for binary DPSK (DBPSK), the modulation scheme operates such that the difference between successive bits is mapped

to a binary 0 or 1. When the input bit is 1, the differentially encoded symbol remains the same as the previous symbol, while an incoming 0 toggles the output symbol.

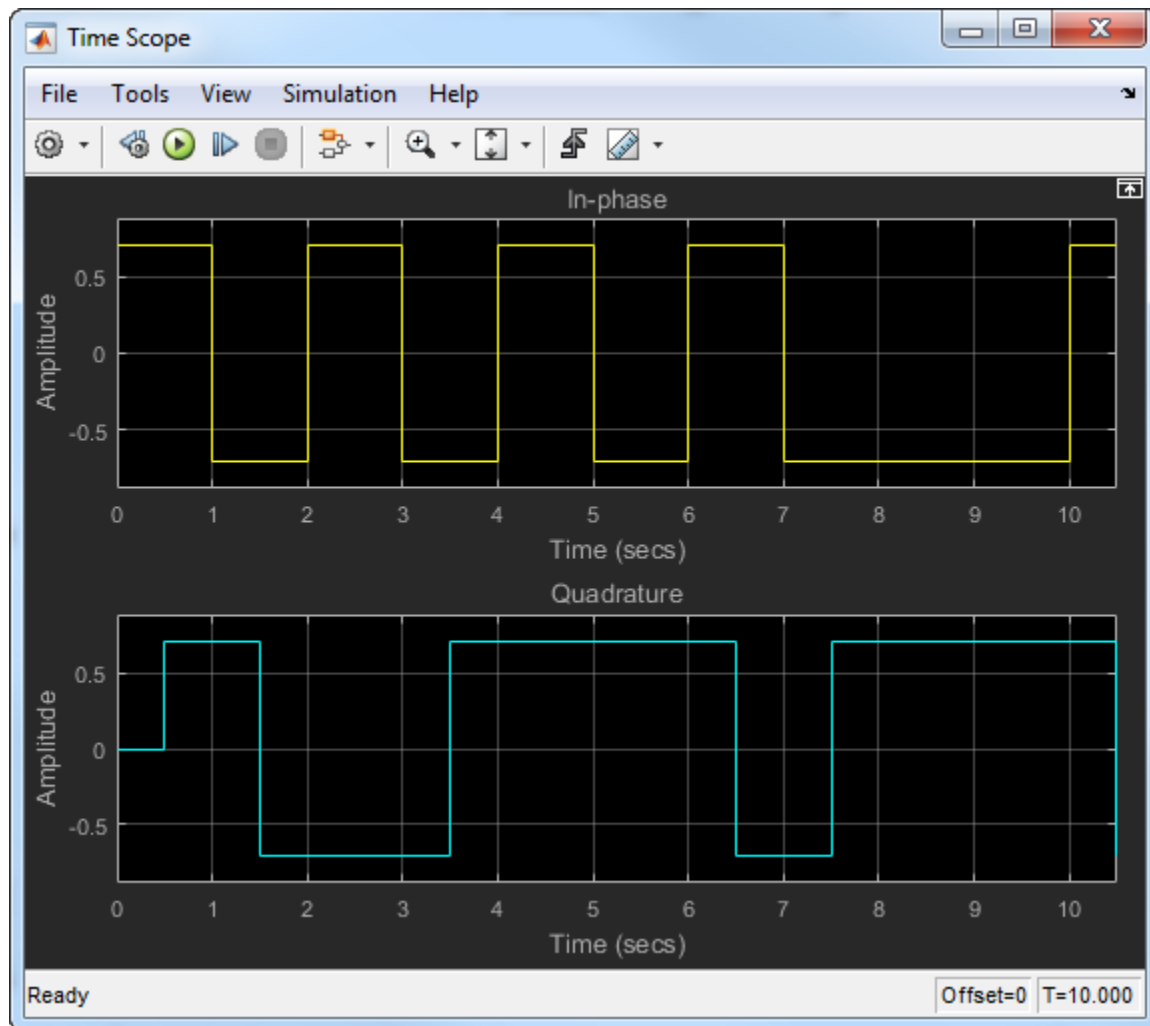
The disadvantage of DPSK is that it is approximately 3 dB less energy efficient than coherent PSK. The bit error probability for DBPSK in AWGN is $P_b = 1/2 \exp(-E_b/N_0)$.

OQPSK

Offset QPSK is similar to QPSK except that the time alignment of the in-phase and quadrature bit streams differs. In QPSK, the in-phase and quadrature bit streams transition at the same time. In OQPSK, the transitions have an offset of a half-symbol period as shown.



The in-phase and quadrature signals transition only on boundaries between symbols. These transitions occur at 1-second intervals because the sample rate is 1 Hz. The following figure shows the in-phase and quadrature signals for an OQPSK signal.



For OQPSK, the quadrature signal has a 1/2 symbol period offset (0.5 s).

The BER for an OQPSK signal in AWGN is identical to that of a QPSK signal. The BER is

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right),$$

where E_b is the energy per bit and N_0 is the noise power spectral density.

Soft-Decision Demodulation

All Communications Toolbox demodulator functions, System objects and blocks can demodulate binary data using either hard decisions or soft decisions. Two soft-decision algorithms are available: exact log-likelihood ratio (LLR) and approximate LLR. Exact LLR provides the greatest accuracy but is slower, while approximate LLR is less accurate but more efficient.

Exact LLR Algorithm

The log-likelihood ratio (LLR) is the logarithm of the ratio of probabilities of a 0 bit being transmitted versus a 1 bit being transmitted for a received signal. The LLR for a bit, b , is defined as:

$$L(b) = \log\left(\frac{\Pr(b = 0 | r = (x, y))}{\Pr(b = 1 | r = (x, y))}\right)$$

Assuming equal probability for all symbols, the LLR for an AWGN channel can be expressed as:

$$L(b) = \log\left(\frac{\sum_{s \in S_0} e^{-\frac{1}{\sigma^2}((x - s_x)^2 + (y - s_y)^2)}}{\sum_{s \in S_1} e^{-\frac{1}{\sigma^2}((x - s_x)^2 + (y - s_y)^2)}}\right)$$

Variable	Description
r	Received signal with coordinates (x, y)
b	Transmitted bit (one of the K bits in an M -ary symbol, assuming all M symbols are equally probable)
S_0	Ideal symbols or constellation points with bit 0, at the given bit position
S_1	Ideal symbols or constellation points with bit 1, at the given bit position
s_x	In-phase coordinate of ideal symbol or constellation point

Variable	Description
s_y	Quadrature coordinate of ideal symbol or constellation point
σ^2	Noise variance of baseband signal
σ_x^2	Noise variance along in-phase axis
σ_y^2	Noise variance along quadrature axis

Note Noise components along the in-phase and quadrature axes are assumed to be independent and of equal power, that is, $\sigma_x^2 = \sigma_y^2 = \sigma^2/2$.

Approximate LLR Algorithm

Approximate LLR is computed by using only the nearest constellation point to the received signal with a 0 (or 1) at that bit position, rather than all the constellation points as done in exact LLR. It is defined in [2] as:

$$L(b) = -\frac{1}{\sigma^2} \left(\min_{s \in S_0} \left((x - s_x)^2 + (y - s_y)^2 \right) - \min_{s \in S_1} \left((x - s_x)^2 + (y - s_y)^2 \right) \right)$$

References

- [1] Rappaport, Theodore S. *Wireless Communications: Principles and Practice*. Upper Saddle River, NJ: Prentice Hall, 1996, pp. 238-248.
- [2] Viterbi, A. J. "An Intuitive Justification and a Simplified Implementation of the MAP Decoder for Convolutional Codes," *IEEE Journal on Selected Areas in Communications*. Vol. 16, No. 2, Feb. 1998, pp. 260-264

See Also

Related Examples

- "Estimate BER of 8-PSK in AWGN with Reed-Solomon Coding" on page 11-32
- "Gray Coded 8-PSK" on page 9-15

- “16-PSK with Custom Symbol Mapping” on page 9-56
- “LLR vs. Hard Decision Demodulation”

Various UG Topic Examples

- “Create a Standalone GSM Waveform Explorer Application with MATLAB Compiler” on page 4-2
- “GSM TDMA Frame Parameterization for Waveform Generation” on page 4-5
- “Compensate for Frequency Offset Using Coarse and Fine Compensation” on page 4-24
- “Correct Symbol Timing and Doppler Offsets” on page 4-30
- “Estimate Turbo Code BER Performance in AWGN” on page 4-35
- “Random Noise Generators” on page 4-39
- “Visualize Effects of Frequency-Selective Fading” on page 4-45
- “Correct Phase and Frequency Offset for 16-QAM Using Coarse and Fine Synchronization” on page 4-64
- “Adjust Carrier Synchronizer Damping Factor to Correct Frequency Offset” on page 4-69
- “Modulate and Demodulate 8-PSK Signal” on page 4-74
- “Binary to Gray Conversion in Simulink” on page 4-77
- “Read Baseband Signal from File” on page 4-78
- “Write Baseband Signal to File” on page 4-81

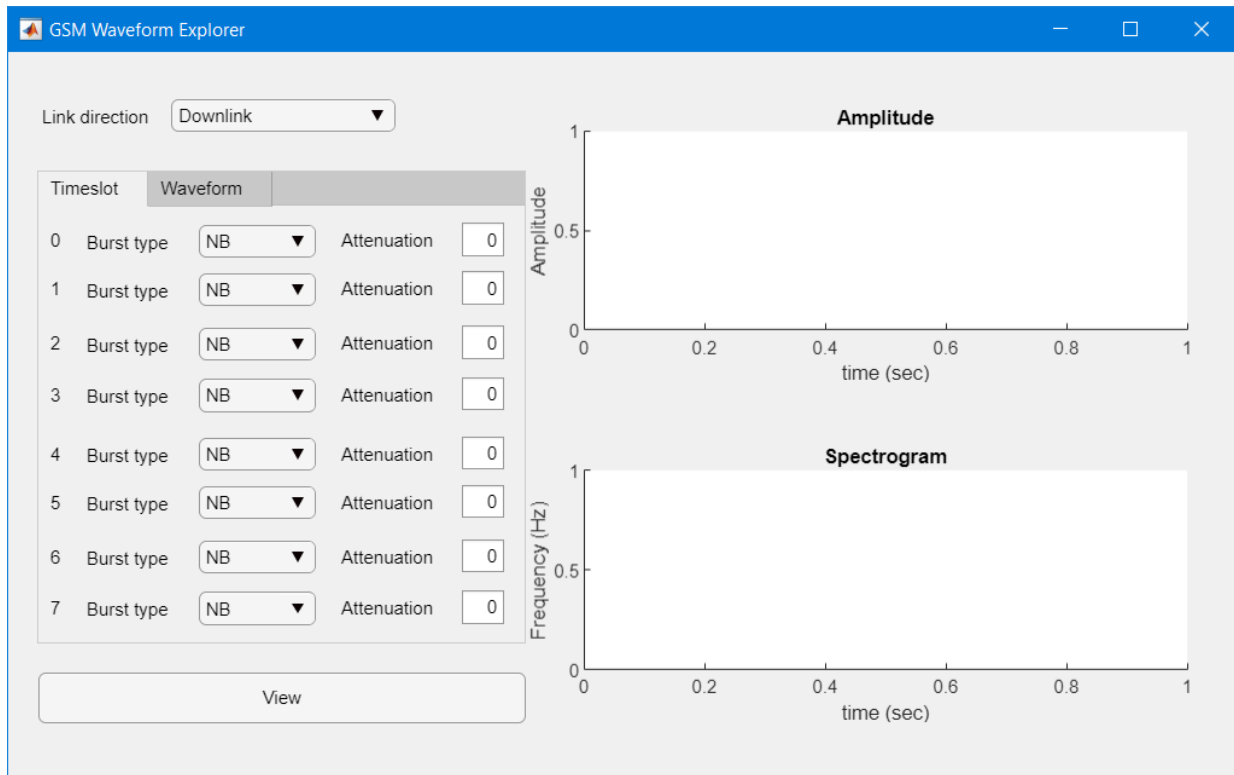
Create a Standalone GSM Waveform Explorer Application with MATLAB Compiler

This example shows how to use MATLAB Compiler™ to create a standalone application from the custom MATLAB™ app, GSMWaveformExplorer, which was created by using App Designer. By installing “MATLAB Runtime” (MATLAB Compiler) you can run standalone applications on systems that do not have MATLAB installed. For more information, see “Create and Run a Simple App Using App Designer” (MATLAB).

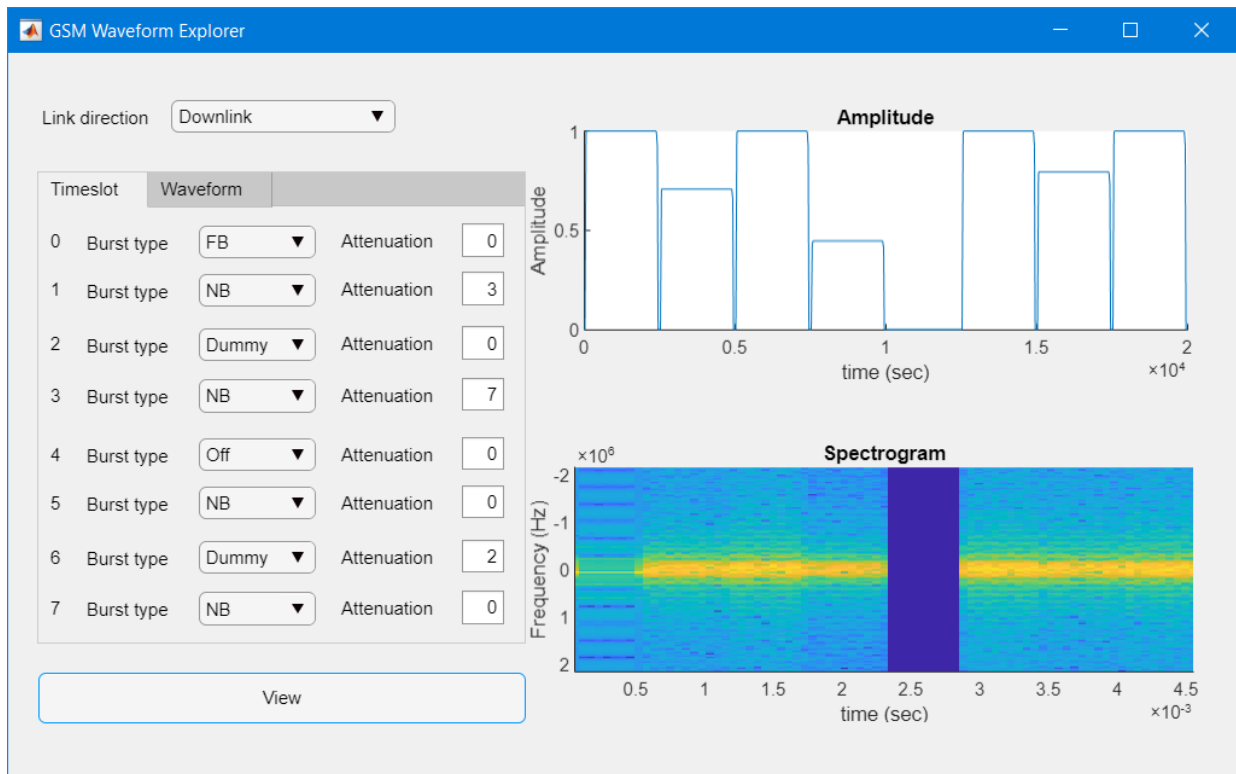
MATLAB Simulation

Open the GSMWaveformExplorer app in MATLAB by entering:

```
GSMWaveformExplorer
```



The `GSMWaveformExplorer` app allows you to explore GSM TDMA frame configurations by using the `gsmUplinkConfig` and `gsmDownlinkConfig` objects and the `gsmFrame` function. You can select the **Link direction** as **Uplink** or **Downlink**. In the **Timeslot** tab, you can adjust the burst type and attenuation of individual timeslots. In the **Waveform** tab, you can adjust the samples per symbol and burst shape. Select **View** to visualize the time domain and spectrogram plots of the waveform.



Compile the MATLAB Function into a Standalone Application

Compile `GSMWaveformExplorer` into a standalone application by using the `mcc` function and specifying the `'-m'` option. This step takes a few minutes to complete. The first message shown below appears only if you have a network installation and the second message appears only if you are running MATLAB Compiler with a demo license.

```
mcc('-m', 'GSMWaveformExplorer');
```

```
Disregard cmd.exe warnings about UNC directory pathnames.  
DEMO Compiler license.
```

```
The generated application will expire 30 days from today,  
on Sun Jul 21 12:05:58 2019.
```

You can also use the interactive `applicationCompiler` app to generate the standalone application.

Run the Standalone Application

Before deploying the standalone app, you can test it on a machine that has MATLAB installed by running commands in the MATLAB Command Window. You can run the standalone `GSMWaveformExplorer` app on a machine that has MATLAB installed by using the `system` command.

- If you are running in the Windows or Linux operating system, type:

```
status = system(fullfile(pwd, 'GSMWaveformExplorer'));
```

- If you are running in the Mac operating system, type:

```
status =  
system(fullfile('GSMWaveformExplorer.app', 'Contents', 'MacOS', 'GSMWaveformExplorer'));
```

Running the standalone application with the `system` command uses the MATLAB environment and any library files needed from this installation of MATLAB. As with running the app in MATLAB, running the standalone version of the `GSMWaveformExplorer` app opens an GSM Waveform Explorer window that allows you to adjust the GSM TDMA frame configurations and view the waveform.

To deploy this application on a machine that does not have MATLAB installed, see “MATLAB Runtime” (MATLAB Compiler).

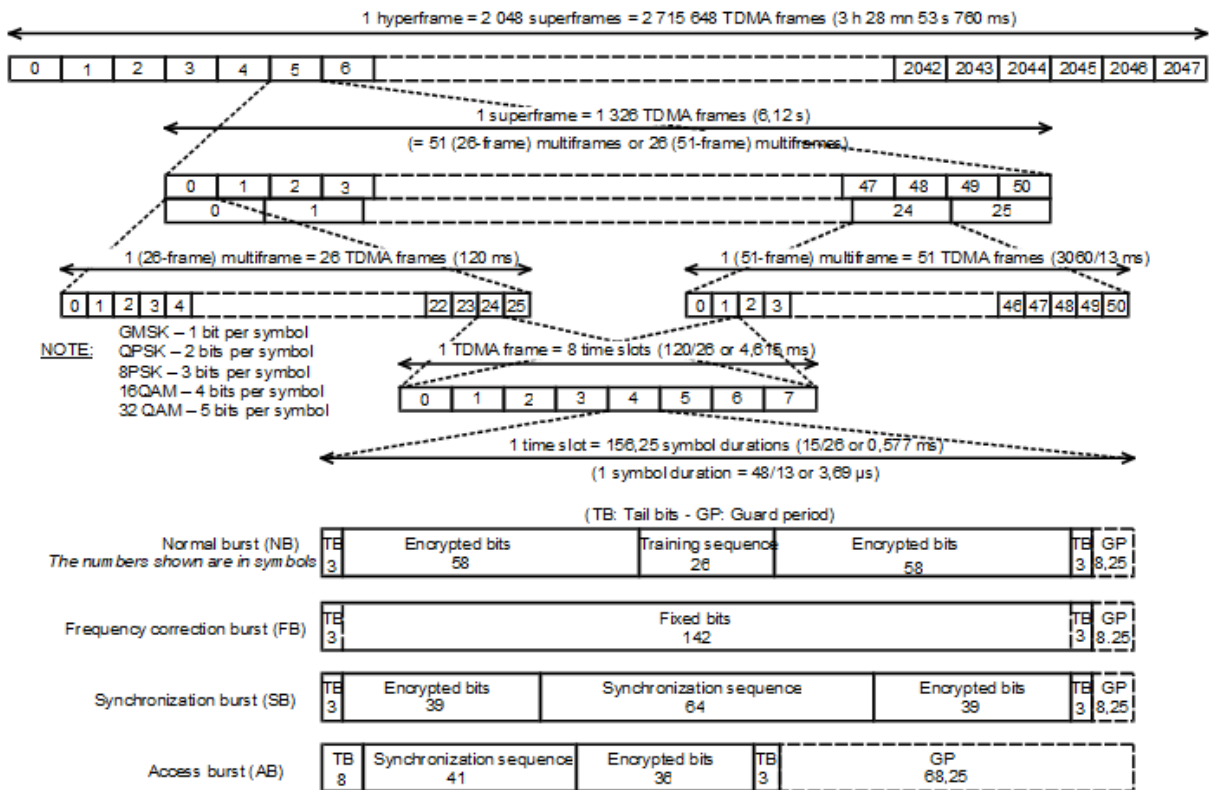
GSM TDMA Frame Parameterization for Waveform Generation

This example shows how to parameterize and generate different GSM TDMA frames and multiframe structures.

Introduction

The GSM standard [1 on page 4-0] specifies a TDMA frame as a combination of 8 time slots. Each time slot has a duration of $3/5200$ seconds (about 0.577 ms) and a time slot number (TN) from 0 to 7. GSM frames use GMSK modulation, where one symbol is equivalent to one bit. Each time slot is 156.25 bits long. The content of a time slot is called a burst. The transmission timing of a burst within a time slot is defined in terms of the bit number. The bit number (BN) refers to a particular bit period within a time slot. The bit with the lowest bit number is transmitted first. BN0 is the first bit period and BN156 is the last quarter-bit period. This figure shows time frames, time slots, and bursts for a GSM system [1 on page 4-0].

4 Various UG Topic Examples



A TDMA contains eight time slots with each timeslot separated by a guard period. Each time slot can carry only one type of burst. Available burst type are: normal burst (NB), frequency correction burst (FB), synchronization burst (SB), access burst (AB), or dummy burst [2 on page 4-0]. The different burst types and the guard period are described in these next sections.

Normal Burst (NB)

The normal burst consists of these bit fields and can appear in uplink or downlink frames. All tail bits are zero. Based on the specified training sequence code (TSC), the training sequence field contains one of eight possible training sequences.

normalBurstDescription()

ans=6x3 table

BitNumber	LengthOfField	ContentsOfField
"0 - 2"	{[3]}	"tail bits"
"3 - 60"	{[58]}	"encrypted bits"
"61 - 86"	{[26]}	"training sequence bits"
"87 - 144"	{[58]}	"encrypted bits"
"145 - 147"	{[3]}	"tail bits"
"148 - 156"	{[8.2500]}	"guard period (bits)"

Access Burst (AB)

The access burst consists of these bit fields and can appear in uplink frames only. All tail bits are zero.

accessBurstDescription()

ans=5x3 table

BitNumber	LengthOfField	ContentsOfField
"0 - 7"	{[8]}	"extended tail bits"
"8 - 48"	{[41]}	"synch. sequence bits"
"49 - 84"	{[36]}	"encrypted bits"
"85 - 87"	{[3]}	"tail bits"
"88 - 156"	{[68.2500]}	"guard period (bits)"

Frequency Correction Burst (FB)

The frequency correction burst consists of these bit fields and can appear in downlink frames only. All tail bits and fixed bits are zero. Modulating all zeros with the GMSK modulator results in a constant phase rotation of -90 degrees for each symbol duration. Therefore, this burst generates an unmodulated carrier with a positive frequency offset of 1625/24 kHz.

frequencyCorrectionBurstDescription()

ans=4x3 table

BitNumber	LengthOfField	ContentsOfField
"0 - 2"	{[3]}	"tail bits"

"3 - 144"	{[142]}	"fixed bits"
"145 - 147"	{[3]}	"tail bits"
"148 - 156"	{[8.2500]}	"guard period (bits)"

Synchronization Burst (SB)

The synchronization burst consists of these bit fields and can appear in downlink frames only. All tail bits are zero.

synchronizationBurstDescription()

ans=6x3 table

BitNumber	LengthOfField	ContentsOfField
"0 - 2"	{[3]}	"tail bits"
"3 - 41"	{[39]}	"encrypted bits"
"42 - 105"	{[64]}	"extended training sequence bits"
"106 - 144"	{[39]}	"encrypted bits"
"145 - 147"	{[3]}	"tail bits"
"148 - 156"	{[8.2500]}	"guard period (bits)"

Dummy Burst

The dummy burst burst consists of these bit fields and can appear in downlink frames only. All tail bits are zero. Mixed bits contain a predetermined sequence of ones and zeros.

dummyBurstDescription()

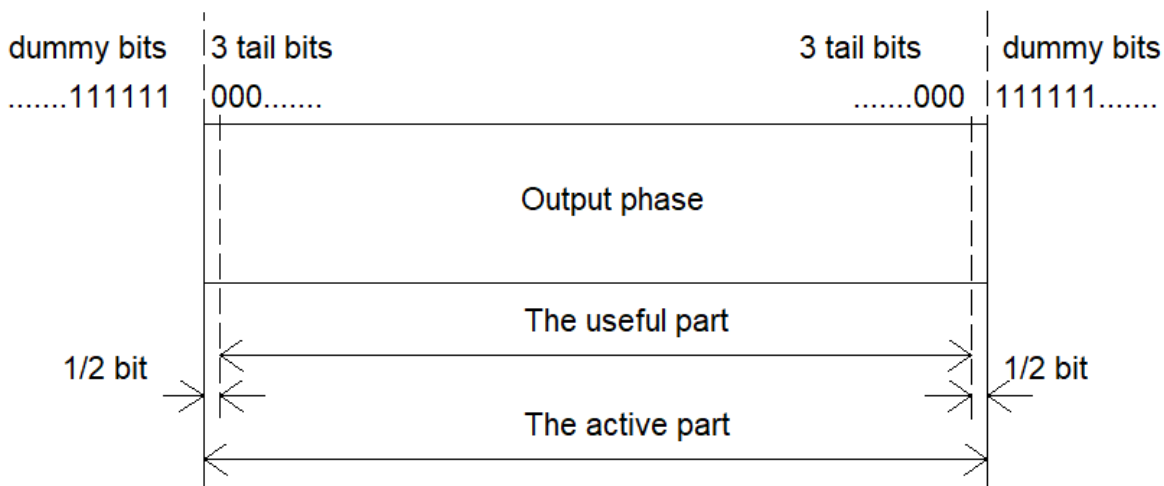
ans=4x3 table

BitNumber	LengthOfField	ContentsOfField
"0 - 2"	{[3]}	"tail bits"
"3 - 144"	{[142]}	"mixed bits"
"145 - 147"	{[3]}	"tail bits"
"148 - 156"	{[8.2500]}	"guard period (bits)"

Guard Period

The GSM standard, [3 on page 4-0], requires mobile stations to attenuate their transmission during the period between bursts. The ramp-up and ramp-down of the signal

power level occurs during the guard periods. The useful part of a burst starts half way through the bit number 0. The useful part ends halfway through BN87 for ABs and BN147 for NBs, FBs, SBs, and dummy bursts. This figure shows the useful and active parts of a burst.



Generate Single Uplink Frame

Configure an uplink GSM TDMA frame using the `gsmUplinkConfig` object.

```
cfg = gsmUplinkConfig()
```

```
cfg =
```

```
gsmUplinkConfig with properties:
```

```

    BurstType: [NB    NB    NB    NB    NB    NB    NB    NB]
SamplesPerSymbol: 16
      TSC: [0 1 2 3 4 5 6 7]
Attenuation: [0 0 0 0 0 0 0 0]
    RiseTime: 2
    RiseDelay: 0
    FallTime: 2
    FallDelay: 0

```

Set time slots 2 and 5 to carry access bursts. Since MATLAB array indices start from 1, but time slots start from 0, set the third and sixth elements of the `BurstType` to "AB".

```
cfg.BurstType([2 5] +1) = "AB"
```

```
cfg =  
  gsmUplinkConfig with properties:  
  
      BurstType: [NB    NB    AB    NB    NB    AB    NB    NB]  
SamplesPerSymbol: 16  
      TSC: [0 1 2 3 4 5 6 7]  
Attenuation: [0 0 0 0 0 0 0 0]  
      RiseTime: 2  
      RiseDelay: 0  
      FallTime: 2  
      FallDelay: 0
```

Assign training sequence codes 3, 5, 1, 7, 0, and 2 to time slots 0, 1, 3, 4, 6, and 7, respectively.

```
cfg.TSC([0 1 3 4 6 7] +1) = [3 5 1 7 0 2]
```

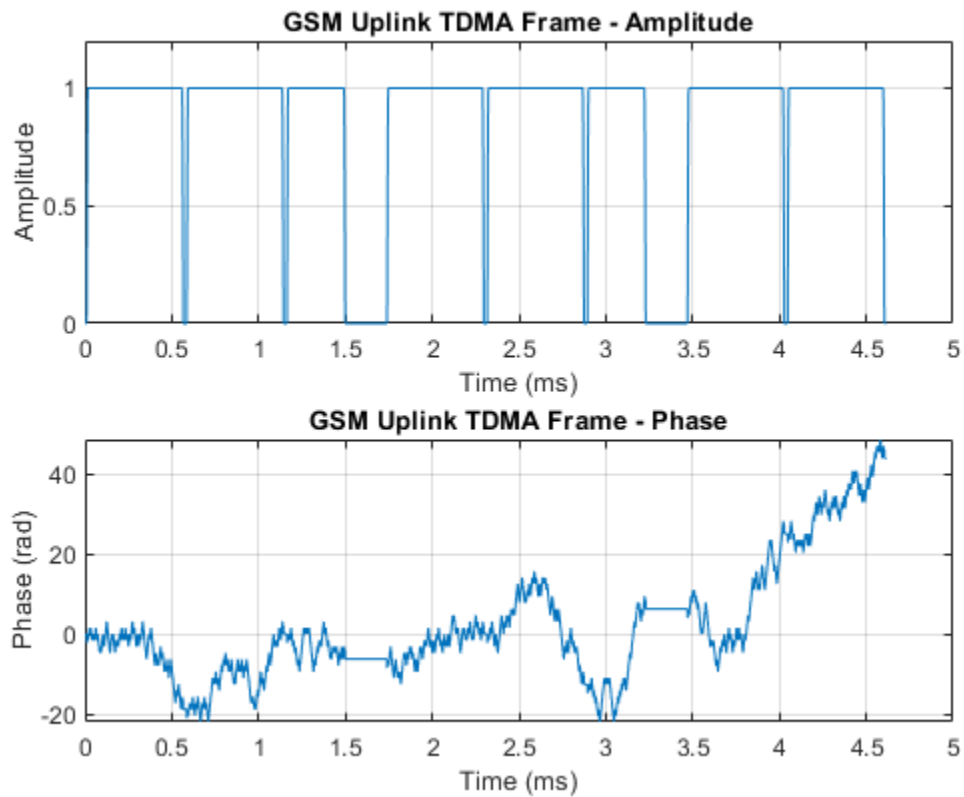
```
cfg =  
  gsmUplinkConfig with properties:  
  
      BurstType: [NB    NB    AB    NB    NB    AB    NB    NB]  
SamplesPerSymbol: 16  
      TSC: [3 5 2 1 7 5 0 2]  
Attenuation: [0 0 0 0 0 0 0 0]  
      RiseTime: 2  
      RiseDelay: 0  
      FallTime: 2  
      FallDelay: 0
```

Generate the baseband samples of the frame using the `gsmFrame` function.

```
x = gsmFrame(cfg);
```

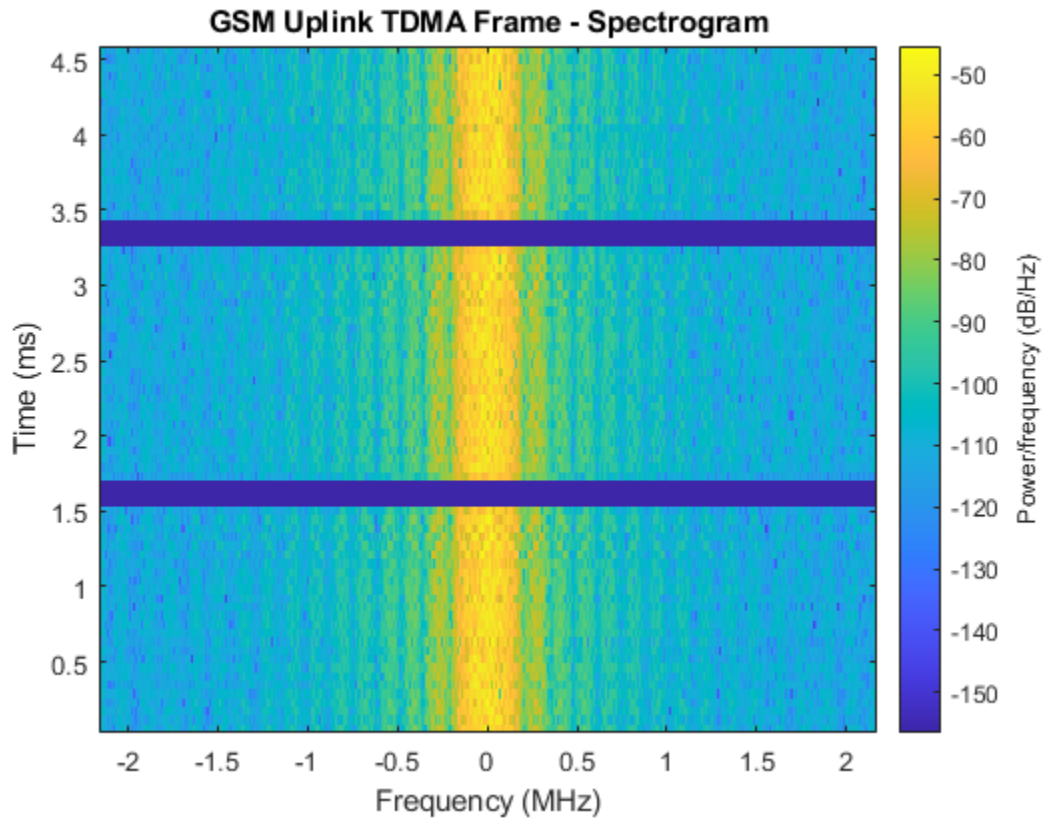
Plot the frame. Get the sample rate of the generated waveform by using the `gsmInfo` function, and then calculate the time axis values in ms. The plot shows 8 bursts in the frame, with guard periods between each burst. As described in the Access Burst (AB) on page 4-0 section, ABs are short burst and have a wider guard period than other bursts.

```
wfInfo = gsmInfo(cfg);  
Rs = wfInfo.SampleRate;  
t = (0:length(x) - 1)/Rs*1e3;  
subplot(2,1,1)  
plot(t,abs(x))  
grid on  
axis([0 5 0 1.2])  
title('GSM Uplink TDMA Frame - Amplitude')  
xlabel('Time (ms)')  
ylabel('Amplitude')  
subplot(2,1,2)  
plot(t,unwrap(angle(x)))  
grid on  
title('GSM Uplink TDMA Frame - Phase')  
xlabel('Time (ms)')  
ylabel('Phase (rad)')
```



Plot the spectrogram of the frame.

```
figure  
spectrogram(x,500,[],[],Rs,'centered')  
title('GSM Uplink TDMA Frame - Spectrogram')
```



Generate Single Downlink Frame

Configure a downlink GSM TDMA frame using the `gsmDownlinkConfig` object.

```
cfg = gsmDownlinkConfig
```

```
cfg =
```

```
gsmDownlinkConfig with properties:
```

```

    BurstType: [NB    NB    NB    NB    NB    NB    NB    NB]
SamplesPerSymbol: 16
          TSC: [0 1 2 3 4 5 6 7]
Attenuation: [0 0 0 0 0 0 0 0]
          RiseTime: 2

```

```
RiseDelay: 0
FallTime: 2
FallDelay: 0
```

Set time slots 0 to carry a frequency correction burst, set time slots 4 and 6 to carry dummy bursts, and set time slot 2 to be empty.

```
cfg.BurstType(0 +1) = "FB";
cfg.BurstType([4 6] +1) = "Dummy";
cfg.BurstType(2 +1) = "Off"
```

```
cfg =
    gsmDownlinkConfig with properties:
```

```
        BurstType: [FB    NB    Off    NB    Dummy    NB    Dummy    NB]
SamplesPerSymbol: 16
          TSC: [0 1 2 3 4 5 6 7]
Attenuation: [0 0 0 0 0 0 0 0]
        RiseTime: 2
        RiseDelay: 0
        FallTime: 2
        FallDelay: 0
```

Generate the baseband samples of the frame using the `gsmFrame` function. This function inserts random bits instead of encrypted bits.

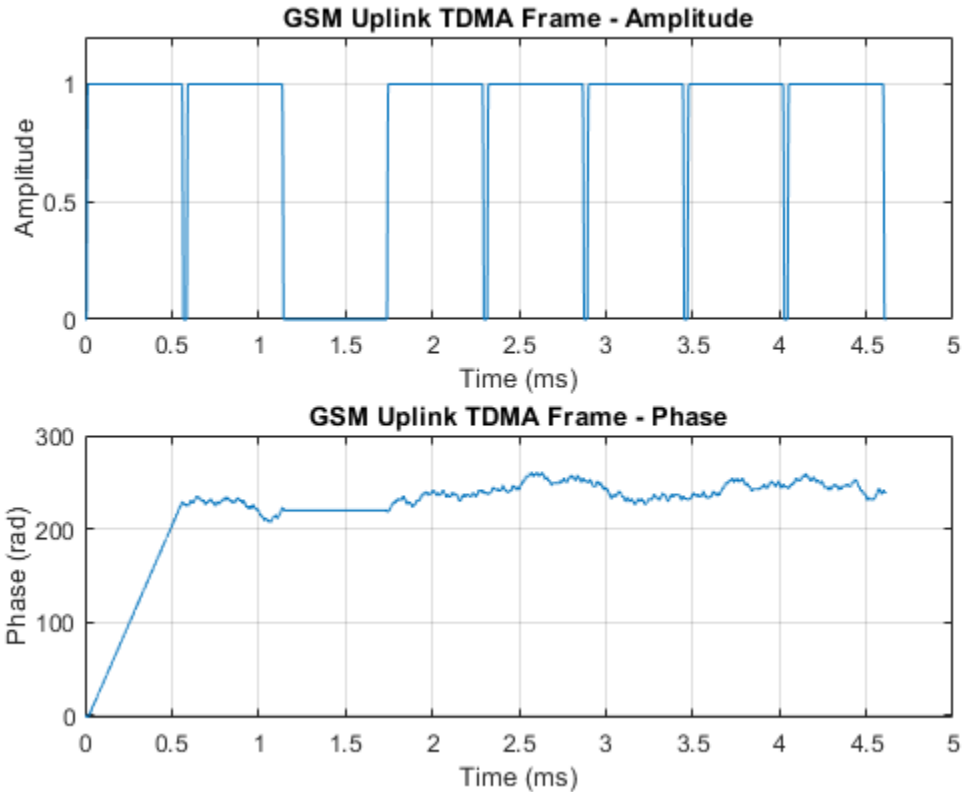
```
x = gsmFrame(cfg);
```

Plot the frame.

```
wfInfo = gsmInfo(cfg);
Rs = wfInfo.SampleRate;
t = (0:length(x) - 1)/Rs*1e3;
subplot(2,1,1)
plot(t,abs(x))
grid on
axis([0 5 0 1.2])
title('GSM Uplink TDMA Frame - Amplitude')
xlabel('Time (ms)');ylabel('Amplitude')
subplot(2,1,2)
plot(t,unwrap(angle(x)))
grid on
title('GSM Uplink TDMA Frame - Phase')
```

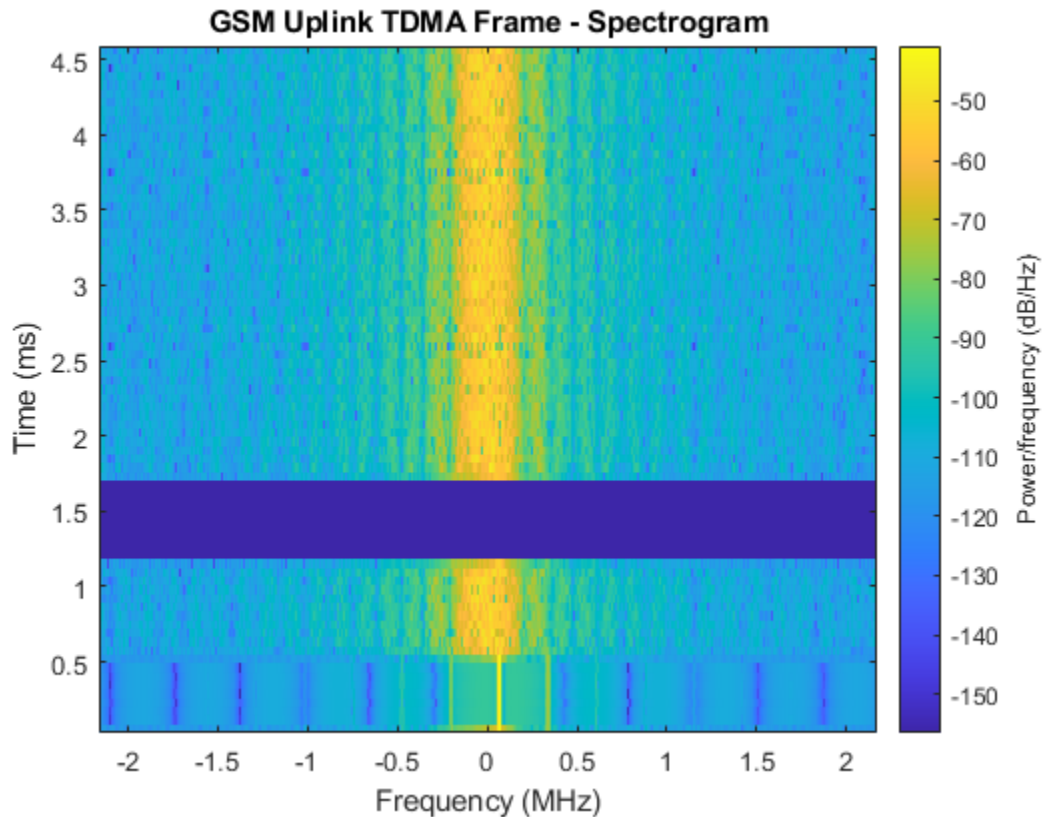


```
xlabel('Time (ms)')
ylabel('Phase (rad)')
```



Plot the spectrogram of the frame. This plot shows the single tone during time slot 0 due to the FB.

```
figure
spectrogram(x,500,[],[],Rs,'centered')
title('GSM Uplink TDMA Frame - Spectrogram')
```



Generate Multiframe Structure

Create a 51-frame multiframe structure, as shown in the figure in the Introduction on page 4-0 section. Create three `gsmDownlinkConfig` objects with specified burst configurations. To assemble the 51-frame multiframe, use the first and second `gsmDownlinkConfig` objects once and repeat the third `gsmDownlinkConfig` objects for the next 49 frames. Repeat the multiframe structure 3 times.

```

cfg1 = gsmDownlinkConfig('BurstType',["FB" "NB" "NB" "NB" "NB" "Dummy" "NB" "NB"]);
cfg2 = gsmDownlinkConfig('BurstType',["SB" "NB" "NB" "NB" "NB" "Dummy" "NB" "NB"]);
cfg3 = gsmDownlinkConfig('BurstType',["NB" "NB" "NB" "NB" "NB" "Dummy" "NB" "NB"]);
wfInfo = gsmInfo(cfg);
frameLength = wfInfo.FrameLengthInSamples;
x = zeros(frameLength*51*3,1);
    
```

```

for p=1:3
    x1 = gsmFrame(cfg1);
    x2 = gsmFrame(cfg2);
    x3 = gsmFrame(cfg3,49);
    x((p-1)*frameLength*51+1:p*frameLength*51) = [x1;x2;x3];
end

```

Simulate Power Control and Propagation Loss Effects

Due to power control and unique propagation loss for each user, the power of each time slot within a frame might be different. Set the power attenuation for time slots 0, 3, and 7 to 2, 6, and 10 dB, respectively.

```

cfg = gsmUplinkConfig;
cfg.Attenuation([0 3 7] +1) = [2 6 10]

```

```

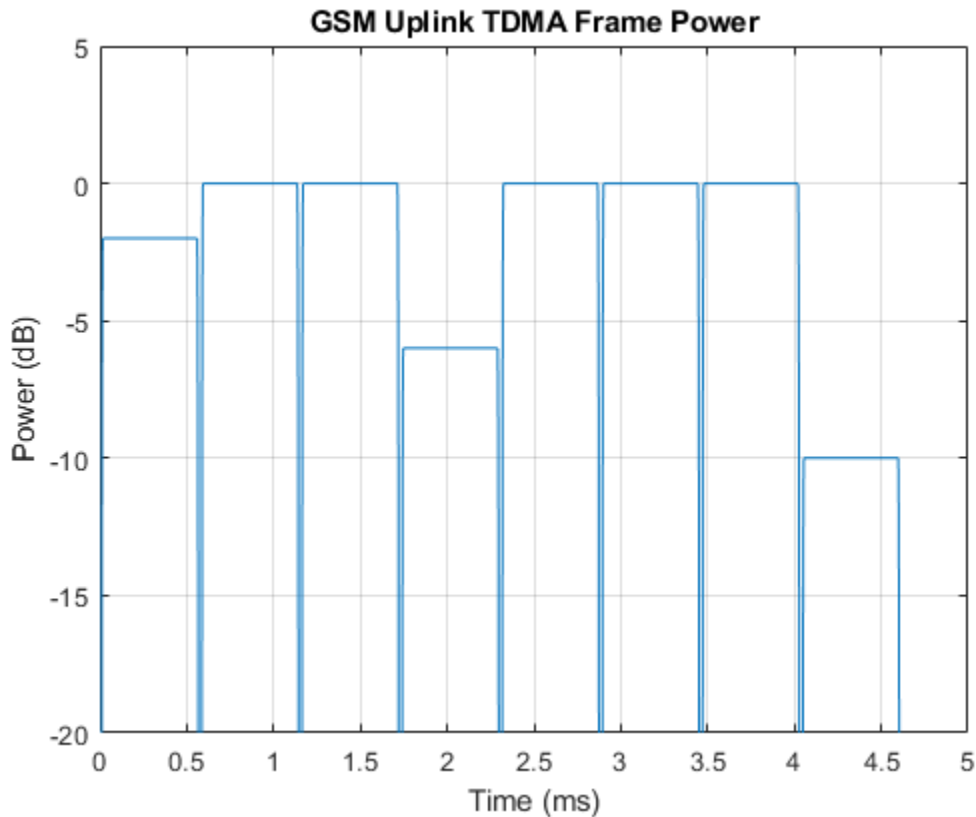
cfg =
    gsmUplinkConfig with properties:
        BurstType: [NB    NB    NB    NB    NB    NB    NB    NB]
        SamplesPerSymbol: 16
        TSC: [0 1 2 3 4 5 6 7]
        Attenuation: [2 0 0 6 0 0 0 10]
        RiseTime: 2
        RiseDelay: 0
        FallTime: 2
        FallDelay: 0

```

```

x = gsmFrame(cfg);
wfInfo = gsmInfo(cfg);
Rs = wfInfo.SampleRate;
t = (0:length(x) - 1)/Rs*1e3;
plot(t, 20*log10(abs(x)))
axis([0 5 -20 5])
grid on
title('GSM Uplink TDMA Frame Power')
xlabel('Time (ms)')
ylabel('Power (dB)')

```



Adjust Ramp-Up and Ramp-Down Behavior

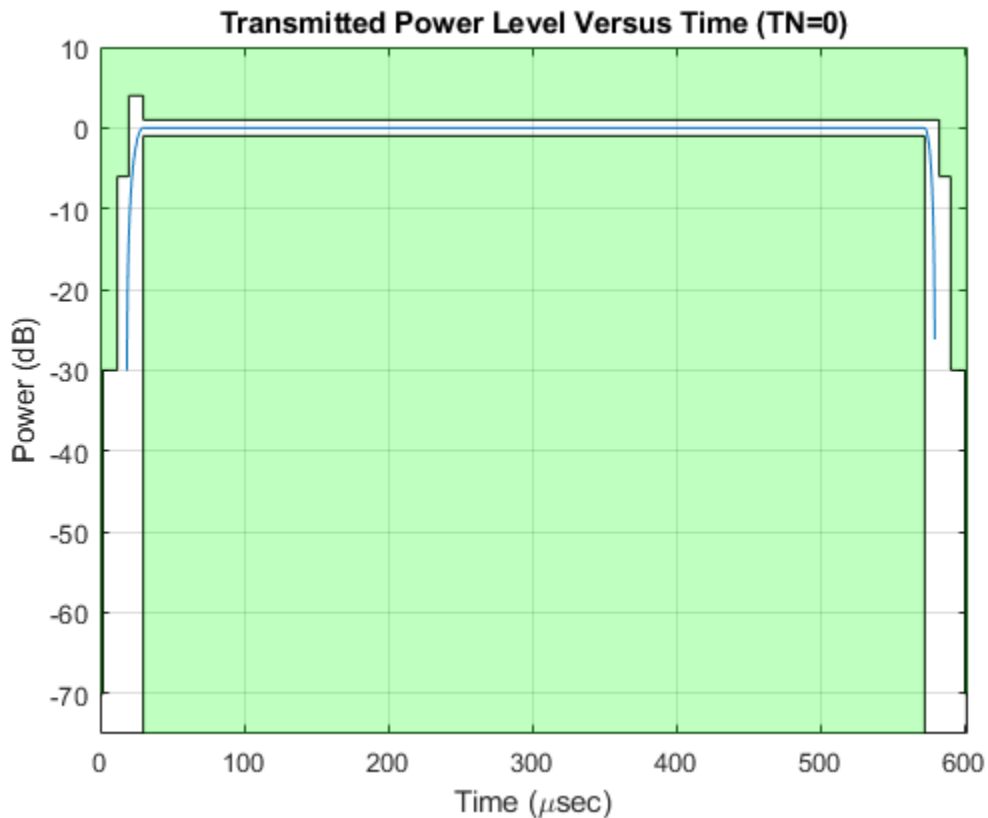
GSM bursts must ramp up and ramp down during guard periods [2] on page 4-0 . The `gsmFrame` function implements the rise and fall characteristics of the bursts as a sinusoid. The burst ramps up from zero to full amplitude in the number of symbol durations specified by the `RiseTime` property value. The resolution of `RiseTime` is $1/N_{\text{sps}}$, where N_{sps} represents the `SamplesPerSymbol` property value of the `gsmDownlinkConfig` object.

Adjust the ramp-up characteristics of the bursts. Since `SamplesPerFrame` is 16, you can specify `RiseTime` with a symbol duration resolution of 0.0625. Set `RiseTime` to a duration of 3.125 symbols.

```
cfg = gsmDownlinkConfig;  
cfg.RiseTime = 3.125;
```

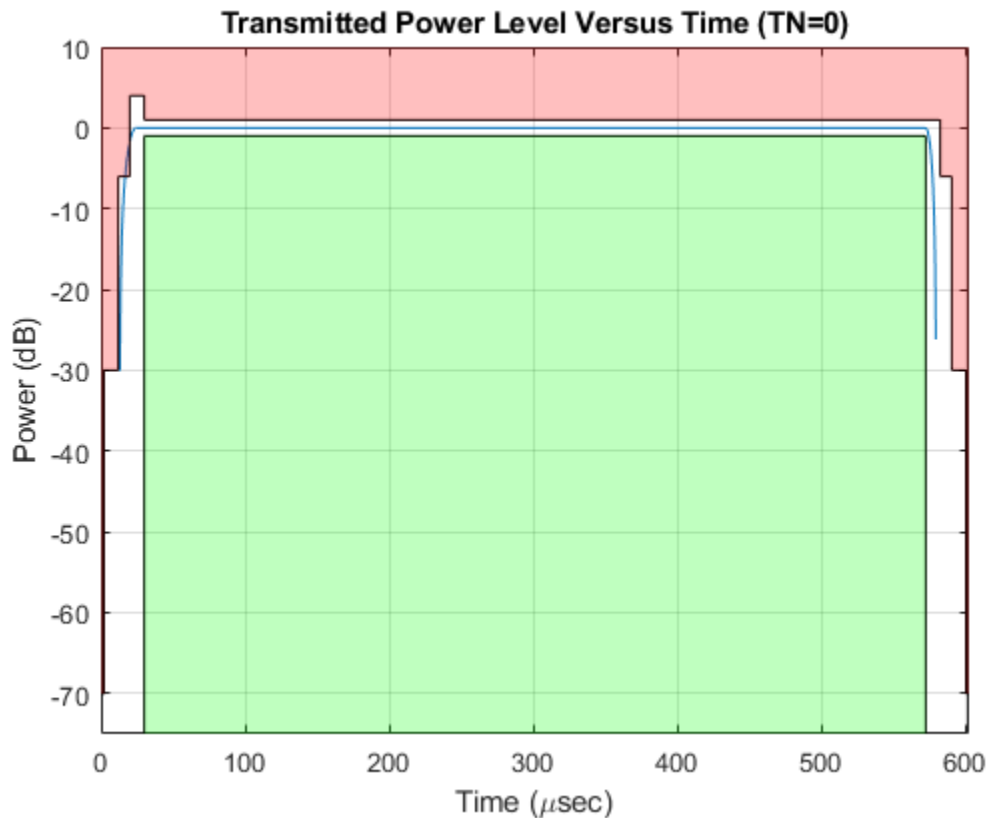
Visualize and check if the rise-time characteristics are within the GSM specifications by using the `gsmCheckTimeMask` function.

```
gsmCheckTimeMask(cfg)
```



Move the start of the rise time duration to the left by 1.5 symbols by setting the `RiseDelay` to -1.5. When `RiseDelay` is 0, the burst reaches full amplitude at the start of the useful part of the burst.

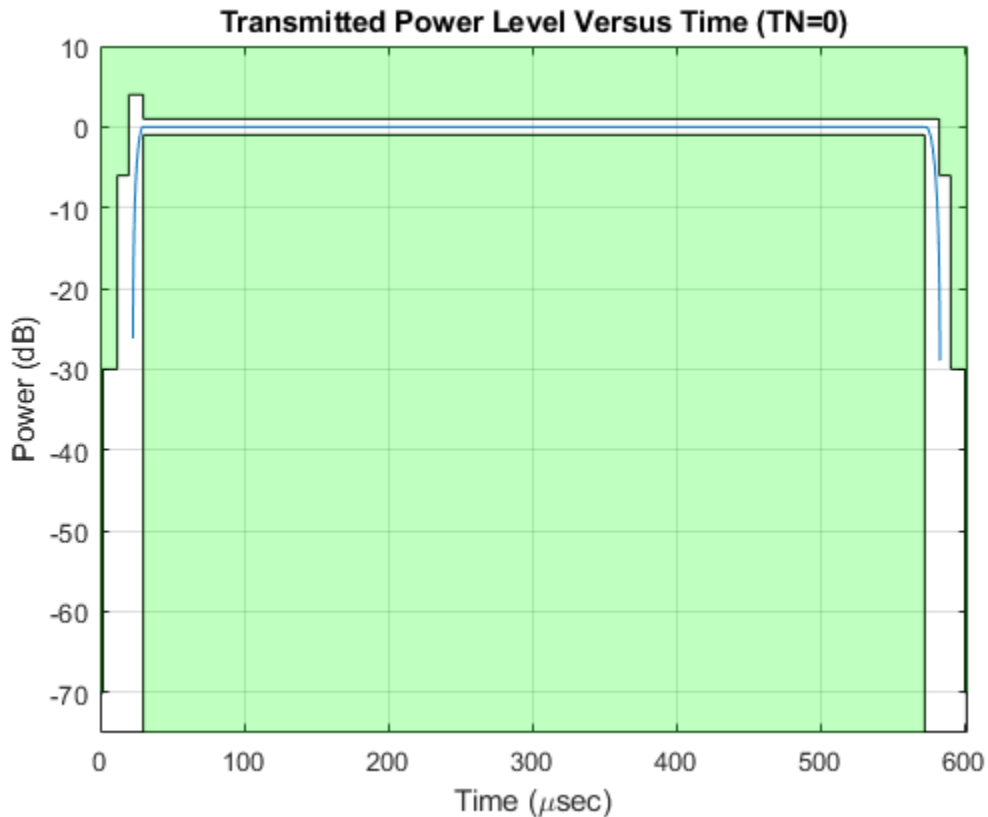
```
cfg.RiseDelay = -1.5;  
gsmCheckTimeMask(cfg)
```



The burst ramps down from full amplitude to zero in the number of symbol durations specified by the `FallTime` property. The resolution of `FallTime` is $1/N_{\text{sps}}$, where N_{sps} represents the `SamplesPerSymbol` property value of the `gsmDownlinkConfig` object. Set `FallTime` to a duration of 2.75 symbols.

Move the start of the fall time to the right by 0.25 symbols durations by setting the `FallDelay` to 0.25. When `FallDelay` is 0, the burst starts to ramp down from full amplitude at the end of the useful part of the burst.

```
cfg = gsmDownlinkConfig;  
cfg.FallTime = 2.75;  
cfg.FallDelay = 0.25;  
gsmCheckTimeMask(cfg)
```



References

- [1] 3GPP TS 45.001. "GSM/EDGE Physical layer on the radio path. *General description.*" *3rd Generation Partnership Project; Technical Specification Group Radio Access Network.*
- [2] 3GPP TS 45.002, "GSM/EDGE Multiplexing and multiple access on the radio path." *3rd Generation Partnership Project; Technical Specification Group Radio Access Network*
- [3] 3GPP TS 45.004, "GSM/EDGE Modulation." *General description.*" *3rd Generation Partnership Project; Technical Specification Group Radio Access Network*

Helper Functions

normalBurstDescription

This function formats a table to show information about normal burst fields.

```
function d = normalBurstDescription()
BitNumber = ["0 - 2";"3 - 60";"61 - 86";...
            "87 - 144";"145 - 147";"148 - 156"];
LengthOfField = {3;58;26;58;3;8.25};
ContentsOfField = [...
    "tail bits";...
    "encrypted bits";...
    "training sequence bits";...
    "encrypted bits";...
    "tail bits";...
    "guard period (bits)"...
    ];
d = table(BitNumber,LengthOfField,ContentsOfField);
end
```

frequencyCorrectionBurstDescription

This function formats a table to show information about frequency correction burst fields.

```
function d = frequencyCorrectionBurstDescription()
BitNumber = ["0 - 2";"3 - 144";"145 - 147";"148 - 156"];
LengthOfField = {3;142;3;8.25};
ContentsOfField = [...
    "tail bits";...
    "fixed bits";...
    "tail bits";...
    "guard period (bits)"...
    ];
d = table(BitNumber,LengthOfField,ContentsOfField);
end
```

synchronizationBurstDescription

This function formats a table to show information about synchronization burst fields.

```
function d = synchronizationBurstDescription()
BitNumber = ["0 - 2";"3 - 41";"42 - 105";...
            "106 - 144";"145 - 147";"148 - 156"];
LengthOfField = {3;39;64;39;3;8.25};
ContentsOfField = [...
    "tail bits";...
    "encrypted bits";...
    "extended training sequence bits";...
    ];
```



```

        "encrypted bits";...
        "tail bits";...
        "guard period (bits)"...
    ];
d = table(BitNumber,LengthOfField,ContentsOfField);
end

```

dummyBurstDescription

This function formats a table to show information about dummy burst fields.

```

function d = dummyBurstDescription()
BitNumber = ["0 - 2";"3 - 144";"145 - 147";"148 - 156"];
LengthOfField = {3;142;3;8.25};
ContentsOfField = [...
    "tail bits";...
    "mixed bits";...
    "tail bits";...
    "guard period (bits)"...
];
d = table(BitNumber,LengthOfField,ContentsOfField);
end

```

accessBurstDescription

This function formats a table to show information about access burst fields.

```

function d = accessBurstDescription()
BitNumber = ["0 - 7";"8 - 48";"49 - 84";...
    "85 - 87";"88 - 156"];
LengthOfField = {8;41;36;3;68.25};
ContentsOfField = [...
    "extended tail bits";...
    "synch. sequence bits";...
    "encrypted bits";...
    "tail bits";...
    "guard period (bits)"...
];
d = table(BitNumber,LengthOfField,ContentsOfField);
end

```

Compensate for Frequency Offset Using Coarse and Fine Compensation

Correct for a phase and frequency offset in a noisy QAM signal using a carrier synchronizer. Then correct for the offsets using both a carrier synchronizer and a coarse frequency compensator.

Set the example parameters.

```
fs = 10000;           % Symbol rate (Hz)
sps = 4;             % Samples per symbol
M = 16;             % Modulation order
k = log2(M);        % Bits per symbol
```

Create a QAM modulator and an AWGN channel.

```
channel = comm.AWGNChannel('EbNo',20,'BitsPerSymbol',k,'SamplesPerSymbol',sps);
```

Create a constellation diagram object to visualize the effects of the offset compensation techniques. Specify the constellation diagram to display only the last 4000 samples.

```
constdiagram = comm.ConstellationDiagram(...
    'ReferenceConstellation',qammod(0:M-1,M), ...
    'SamplesPerSymbol',sps, ...
    'SymbolsToDisplaySource','Property','SymbolsToDisplay',4000, ...
    'XLimits',[-5 5],'YLimits',[-5 5]);
```

Introduce a frequency offset of 400 Hz and a phase offset of 30 degrees.

```
phaseFreqOffset = comm.PhaseFrequencyOffset(...
    'FrequencyOffset',400,...
    'PhaseOffset',30,...
    'SampleRate',fs);
```

Generate random data symbols and apply 16-QAM modulation.

```
data = randi([0 M-1],10000,1);
modSig = qammod(data,M);
```

Create a raised cosine filter object and filter the modulated signal.

```
txfilter = comm.RaisedCosineTransmitFilter('OutputSamplesPerSymbol',sps, ...
    'Gain',sqrt(sps));
txSig = txfilter(modSig);
```

Apply the phase and frequency offset, and then pass the signal through the AWGN channel.

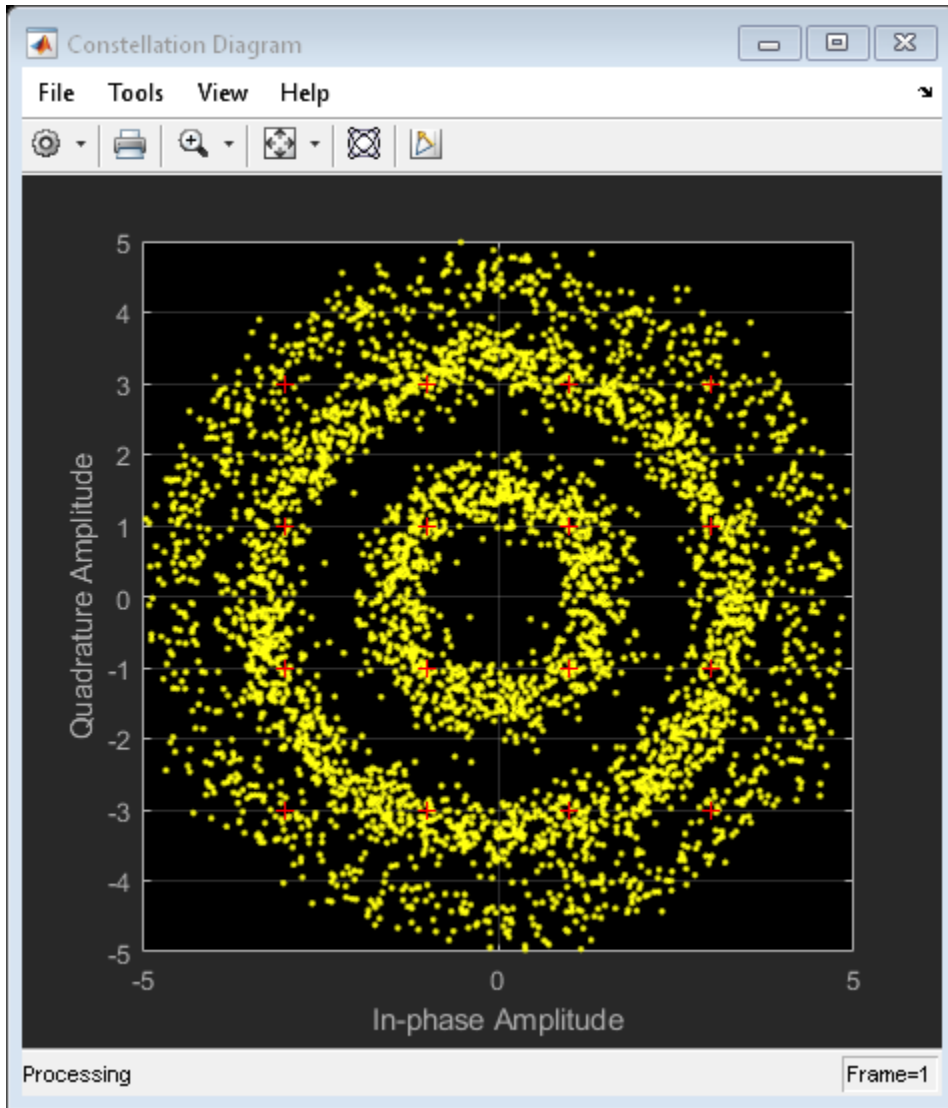
```
freqOffsetSig = phaseFreqOffset(txSig);  
rxSig = channel(freqOffsetSig);
```

Apply fine frequency correction to the signal by using the carrier synchronizer.

```
fineSync = comm.CarrierSynchronizer('DampingFactor',0.7, ...  
    'NormalizedLoopBandwidth',0.005, ...  
    'SamplesPerSymbol',sps, ...  
    'Modulation','QAM');  
rxData = fineSync(rxSig);
```

Display the constellation diagram of the last 4000 symbols.

```
constdiagram(rxData)
```



Even with time to converge, the spiral nature of the plot shows that the carrier synchronizer has not yet compensated for the large frequency offset. The 400 Hz offset is 1% of the sample rate.

Repeat the process with a coarse frequency compensator inserted before the carrier synchronizer.

Create a coarse frequency compensator to reduce the frequency offset to a manageable level.

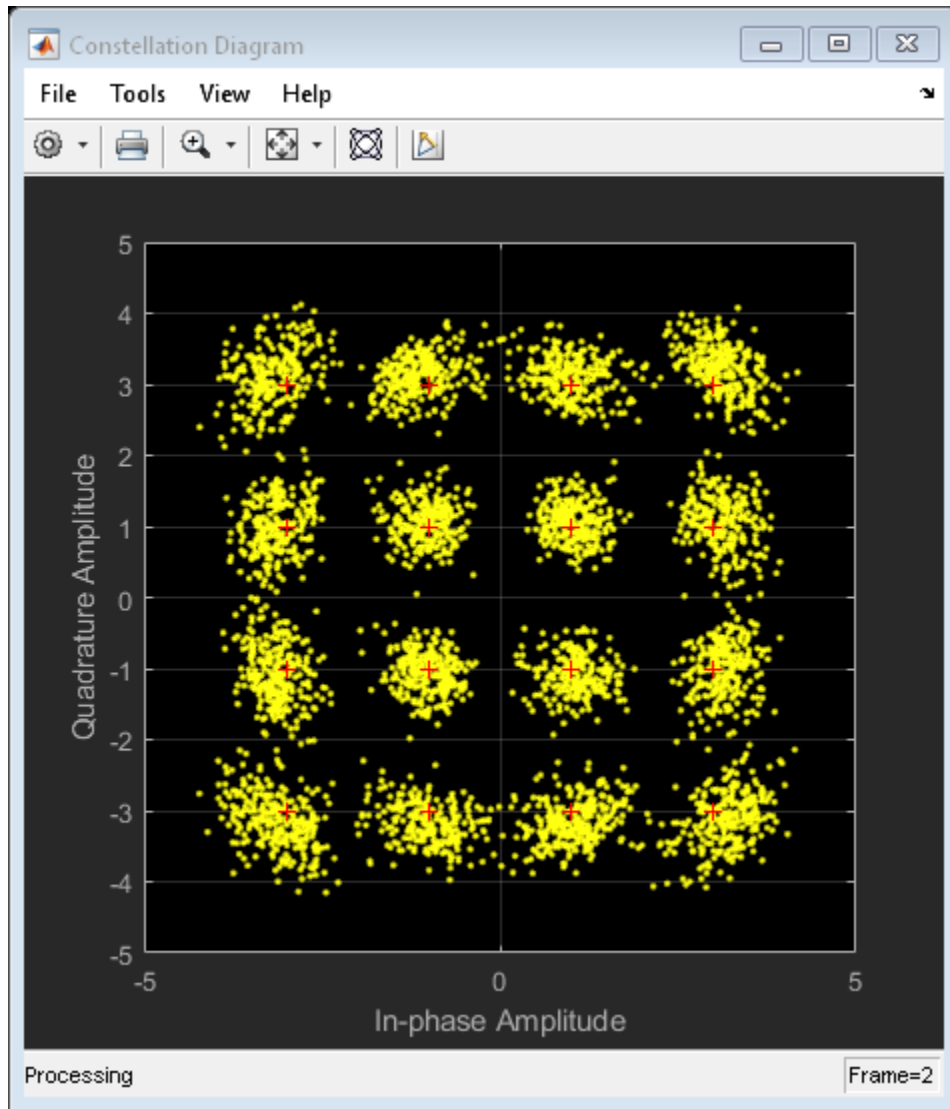
```
coarseSync = comm.CoarseFrequencyCompensator('Modulation','QAM','FrequencyResolution',3);
```

Pass the received signal to the coarse frequency compensator and then to the carrier synchronizer.

```
syncCoarse = coarseSync(rxSig);  
rxData = fineSync(syncCoarse);
```

Plot the constellation diagram of the signal after coarse and fine frequency compensation.

```
constdiagram(rxData)
```



The received data now aligns with the reference constellation.

See Also

`comm.CarrierSynchronizer` | `comm.CoarseFrequencyCompensator`

Correct Symbol Timing and Doppler Offsets

Correct symbol timing and frequency offset errors by using the `comm.SymbolSynchronizer` and `comm.CarrierSynchronizer` System objects.

Configuration

Initialize simulation parameters.

```
M = 16;           % Modulation order
nSym = 2000;     % Number of symbols in a packet
sps = 2;        % Samples per symbol
spsFilt = 8;    % Samples per symbol for filters and channel
spsSync = 2;   % Samples per symbol for synchronizers
lenFilt = 10;  % RRC filter length
```

Create a matched pair of root raised cosine (RRC) filter System objects for transmitter and receiver.

```
txfilter = comm.RaisedCosineTransmitFilter('FilterSpanInSymbols',lenFilt, ...
    'OutputSamplesPerSymbol',spsFilt,'Gain',sqrt(spsFilt));
rxfilter = comm.RaisedCosineReceiveFilter('FilterSpanInSymbols',lenFilt, ...
    'InputSamplesPerSymbol',spsFilt,'DecimationFactor',spsFilt/2,'Gain',sqrt(1/spsFilt));
```

Create a phase-frequency offset System object to introduce a 100 Hz Doppler shift.

```
doppler = comm.PhaseFrequencyOffset('FrequencyOffset',100, ...
    'PhaseOffset',45,'SampleRate',1e6);
```

Create a variable delay System object to introduce timing offsets.

```
varDelay = dsp.VariableFractionalDelay;
```

Create carrier and symbol synchronizer System objects to correct for Doppler shift and timing offset, respectively.

```
carrierSync = comm.CarrierSynchronizer('SamplesPerSymbol',spsSync);
symbolSync = comm.SymbolSynchronizer( ...
    'TimingErrorDetector','Early-Late (non-data-aided)', ...
    'SamplesPerSymbol',spsSync);
```

Create constellation diagram System objects to view the results.

```
refConst = qammod(0:M-1,M,'UnitAveragePower',true);
cdReceive = comm.ConstellationDiagram('ReferenceConstellation',refConst, ...
```



```

    'SamplesPerSymbol',spsFilt,'Title','Received Signal');
cdDoppler = comm.ConstellationDiagram('ReferenceConstellation',refConst, ...
    'SamplesPerSymbol',spsSync,'Title','Frequency Corrected Signal');
cdTiming = comm.ConstellationDiagram('ReferenceConstellation',refConst, ...
    'SamplesPerSymbol',spsSync,'Title','Frequency and Timing Synchronized Signal');

```

Main Processing Loop

The main processing loop:

- Generates random symbols and apply QAM modulation.
- Filters the modulated signal.
- Applies frequency and timing offsets.
- Passes the transmitted signal through an AWGN channel.
- Filters the received signal.
- Corrects the Doppler shift.
- Corrects the timing offset.

```

for k = 1:15
    data = randi([0 M-1],nSym,1);
    modSig = qammod(data,M,'UnitAveragePower',true);
    txSig = txfilter(modSig);

    txDoppler = doppler(txSig);
    txDelay = varDelay(txDoppler,k/15);

    rxSig = awgn(txDelay,25);

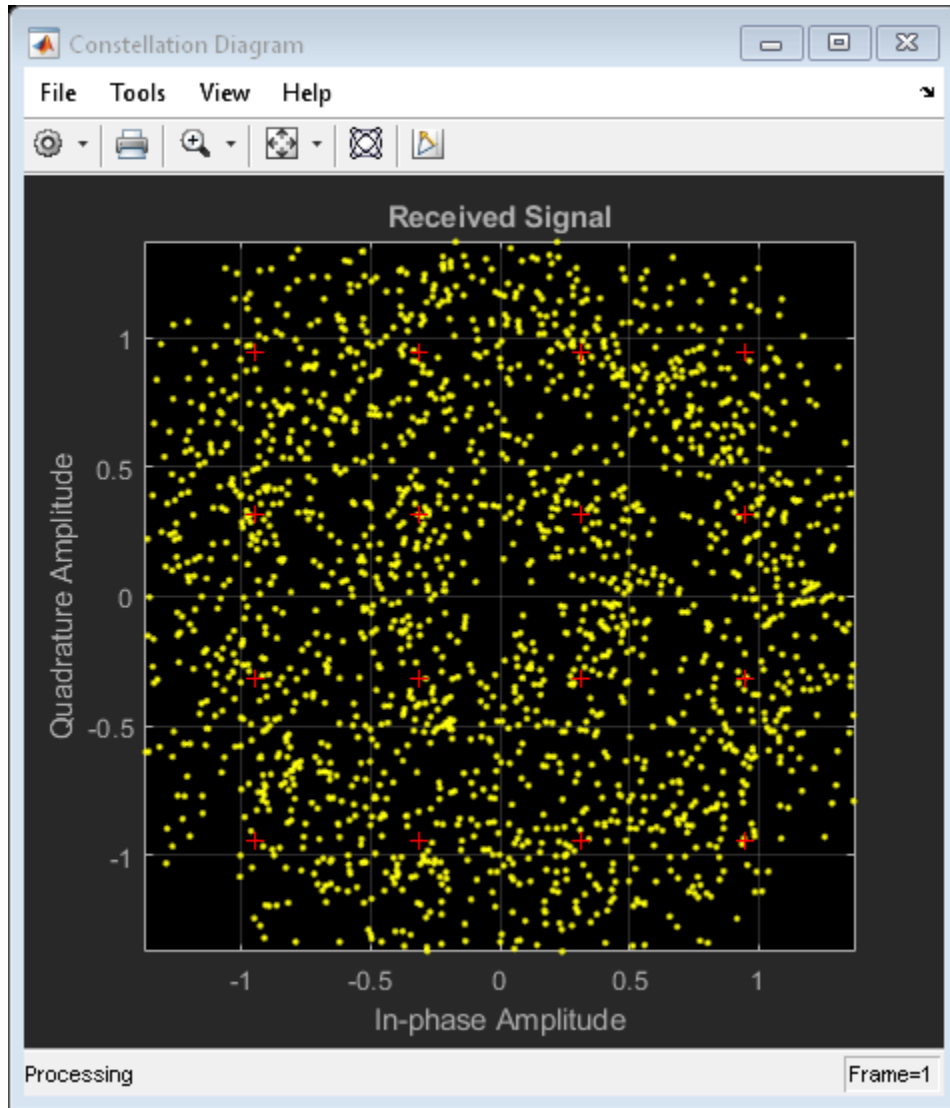
    rxFiltSig = rxfilter(rxSig);
    rxCorr = carrierSync(rxFiltSig);
    rxData = symbolSync(rxCorr);
end

```

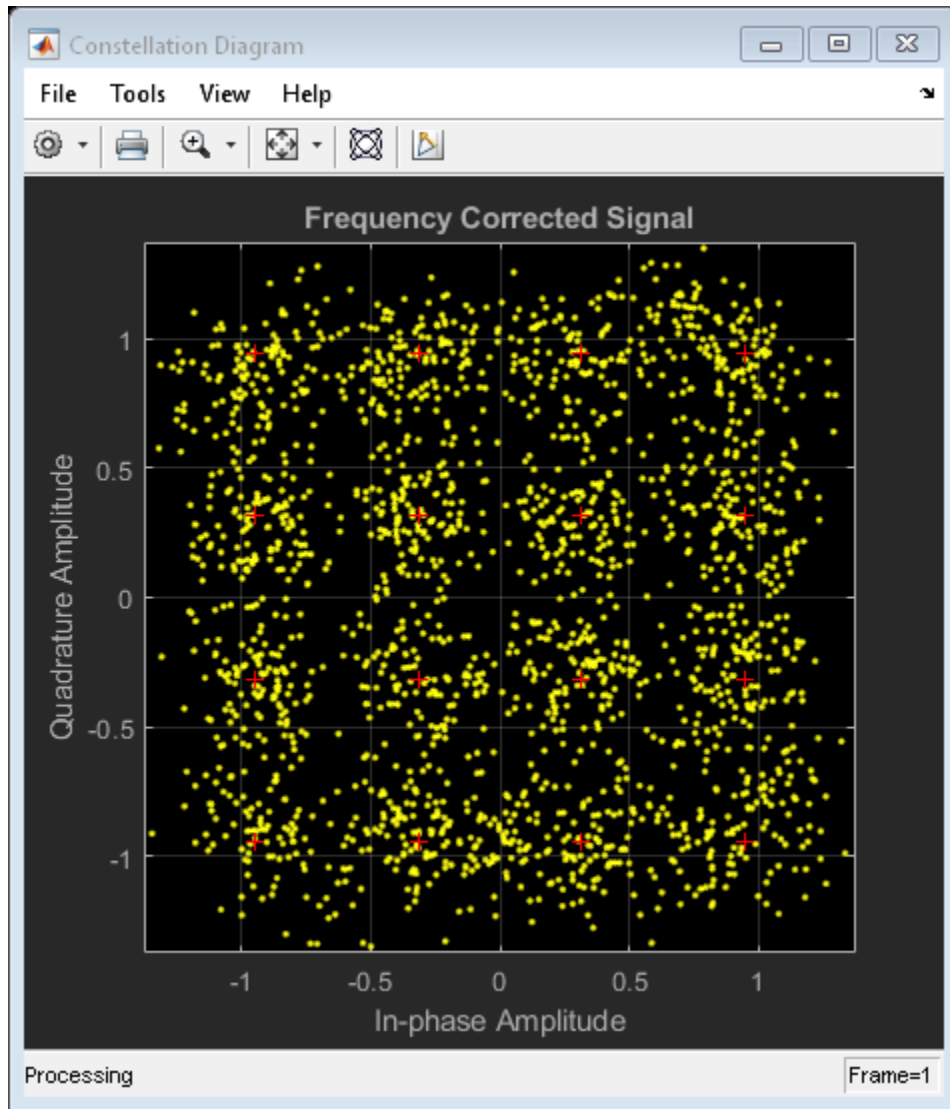
Visualization

Plot the constellation diagrams of the received signal, frequency corrected signal, and frequency and timing synchronized signal. Specific constellation points cannot be identified in the received signal and can be only partially identified in the frequency corrected signal. However, the timing and frequency synchronized signal aligns with the expected QAM constellation points.

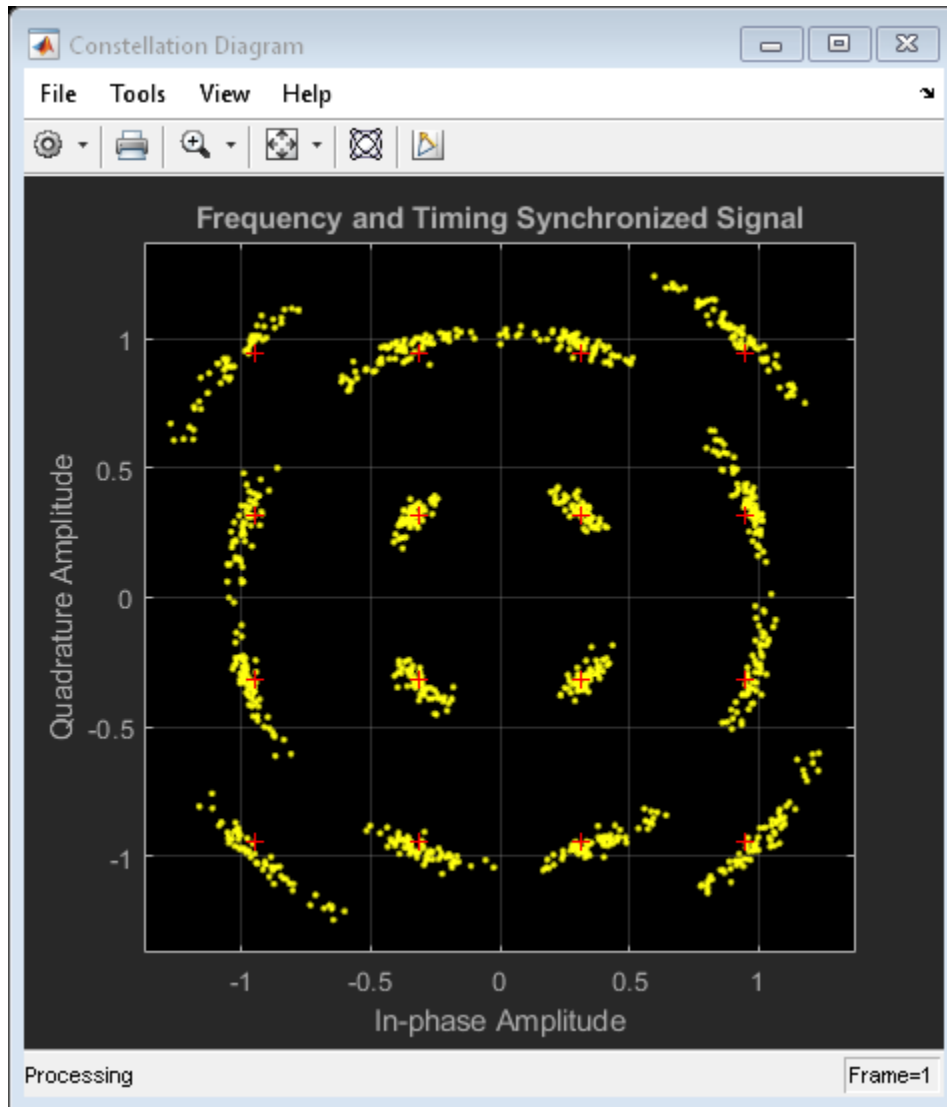
```
cdReceive(rxSig)
```



`cdDoppler(rxCorr)`



```
cdTiming(rxData)
```



See Also

`comm.CarrierSynchronizer` | `comm.SymbolSynchronizer`

Estimate Turbo Code BER Performance in AWGN

Simulate an end-to-end communication link employing 16-QAM using turbo codes in an AWGN channel. Estimate the bit error rate.

Initialize Simulation

Set the modulation order and the range of Eb/No values to evaluate. Set the packet length to 500.

```
rng(10, 'twister');
M = 16; % Modulation order
bps = log2(M); % Bits per symbol
EbNo = (2:0.5:4);
pktLen = 500;
```

Initialize the bit error rate vector.

```
ber = zeros(size(EbNo));
```

Create System objects for a turbo encoder and decoder pair, where the interleaver indices are supplied as input arguments.

```
turboEnc = comm.TurboEncoder('InterleaverIndicesSource', 'Input port');
turboDec = comm.TurboDecoder('InterleaverIndicesSource', 'Input port', ...
    'NumIterations', 4);
```

Create an AWGN channel System object and an error rate counter System object.

```
awgnChannel = comm.AWGNChannel('NoiseMethod', 'Variance', 'Variance', 1);
errorRate = comm.ErrorRate;
```

Use the packet length and turbo encoder settings to determine actual transmitted bit rate. The turbo-coding objects are initialized to use rate-1/2 trellis for their constituent convolutional codes, resulting in a turbo encoder output with **2** parity bit streams, (in addition to the systematic stream) and **12** tail bits for the input packet. The 12 tail bits are due to the specified constraint length of 4 per constituent encoder, which leads to 3-bit outputs per stream, for a total of 4 streams (S1 P1 S2 P2).

```
rate = pktLen/(3*pktLen+4*3);
```

Main Processing Loop

The processing loop performs the following steps:

- Generate random binary data
- Generate random interleaver indices
- Turbo encode the data
- Apply 16-QAM modulation
- Pass the modulated signal through an AWGN channel
- Demodulate the noisy signal using an LLR algorithm
- Turbo decode the data
- Calculate the error statistics

```
for k = 1:length(EbNo)
```

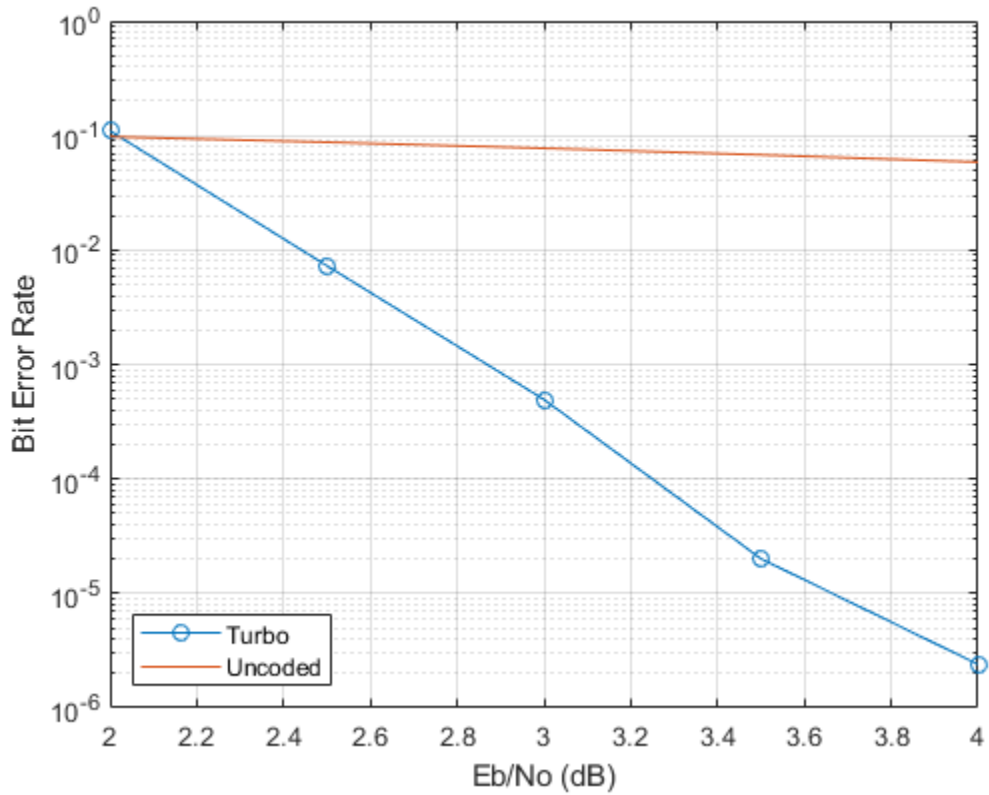
Initialize the error statistics vector, signal-to-noise ratio, and noise variance. Update the AWGN channel System object noise variance value.

```
    errorStats = zeros(1,3);  
    EsNo = EbNo(k) + 10*log10(bps);  
    snrdB = EsNo + 10*log10(rate); % in dB  
    noiseVar = 1./(10.^(snrdB/10));  
    awgnChannel.Variance = noiseVar;  
  
    while errorStats(2) < 100 && errorStats(3) < 1e7  
        % Generate random binary data  
        data = randi([0 1],pktLen,1);  
        % Interleaver indices  
        intrlvrInd = randperm(pktLen);  
        % Turbo encode the data  
        encodedData = turboEnc(data,intrlvrInd);  
        % Modulate the encoded data  
        modSignal = qammod(encodedData,M,'InputType','bit','UnitAveragePower',true);  
        % Pass the signal through the AWGN channel  
        rxSignal = awgnChannel(modSignal);  
        % Demodulate the received signal  
        demodSignal = qamdemod(rxSignal,M,'UnitAveragePower',true,'OutputType','llr','I');  
        % Turbo decode the demodulated signal. Because the bit mapping from the  
        % demodulator is opposite that expected by the turbo decoder, the  
        % decoder input must use the inverse of demodulated signal.  
        rxBits = turboDec(-demodSignal,intrlvrInd);
```

```
        % Calculate the error statistics
        errorStats = errorRate(data,rxBits);
    end
    % Save the BER data and reset the bit error rate object
    ber(k) = errorStats(1);
    reset(errorRate)
end
```

Plot the bit error rate and compare it to the uncoded bit error rate.

```
semilogy(EbNo,ber, '-o')
grid
xlabel('Eb/No (dB)')
ylabel('Bit Error Rate')
uncodedBER = berawgn(EbNo,'qam',M); % Estimate of uncoded BER
hold on
semilogy(EbNo,uncodedBER)
legend('Turbo','Uncoded','location','sw')
```

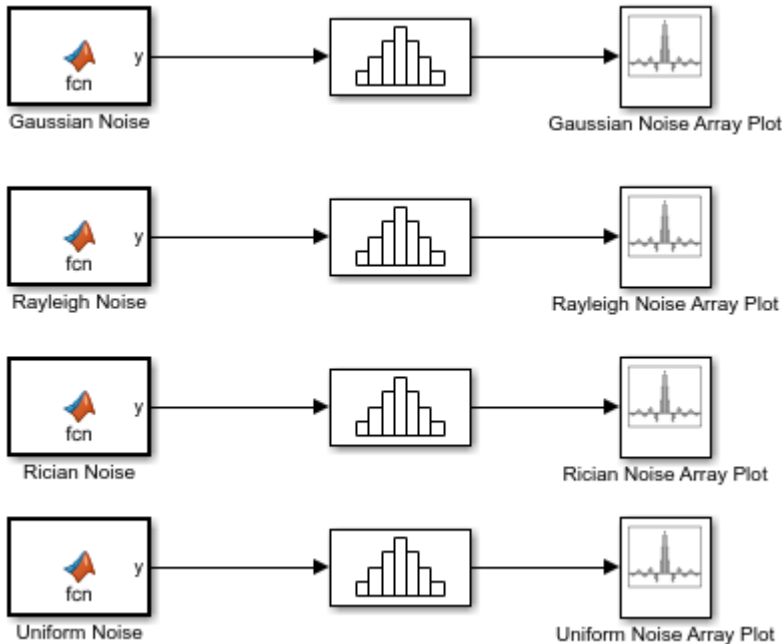


See Also

`comm.TurboDecoder` | `comm.TurboEncoder`

Random Noise Generators

You can generate noise for communication system modeling using the MATLAB Function block with a random number generator. This example generates and displays histogram plots of Gaussian, Rayleigh, Rician, and Uniform noise.



The noise generators output $1e5$ -by- 1 vectors every second, which is equivalent to a 0.00001 second sample time. In this model, each MATLAB Function block defines a specific noise generator using its underlying function. To view the underlying code for a MATLAB Function block in the MATLAB Editor, open the model, select the desired MATLAB Function block, and then press **Ctrl+u**. Each MATLAB function block contains block mask parameters that map to the function arguments in the underlying code.

For each MATLAB Function block the **Samples per frame** parameter maps to its underlying function argument `spf`. Similarly, **Seed** maps to `seed`.

The **Gaussian Noise** MATLAB Function block maps the **Power (dBW)** parameter to `p`, and defines the function

$$y = wgn(sp, 1, p)$$

The **Rayleigh Noise** MATLAB Function block maps the **Sigma** parameter to α , and defines the function

$$y = abs(alpha * (randn(sp, 1) + 1i * randn(sp, 1)))$$

The **Rician Noise** MATLAB Function block maps the **Rician K-factor** parameter to K and the **Sigma** parameter to s , and defines the function

$$m1 = sqrt(2 * K) * s$$

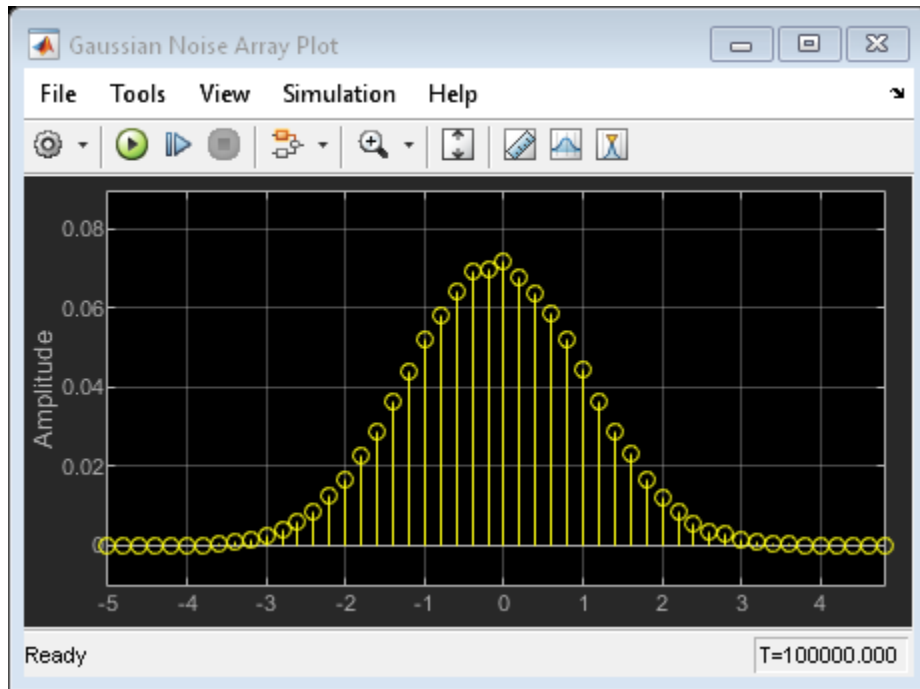
$$m2 = 0$$

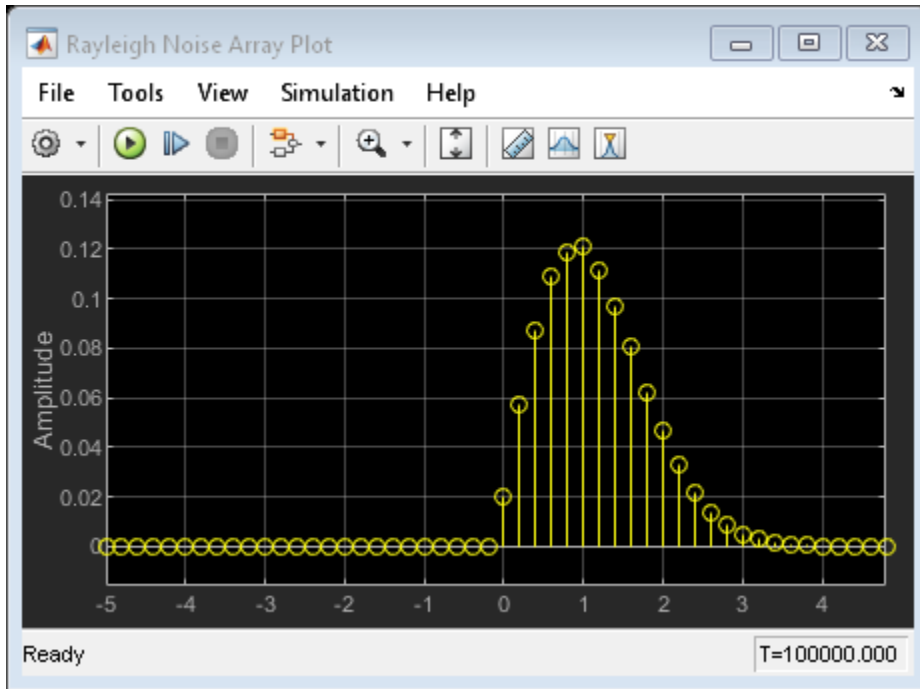
$$y = sqrt((s^2 * randn(sp, 1) + m1)^2 + (s^2 * randn(sp, 1) + m2)^2)$$

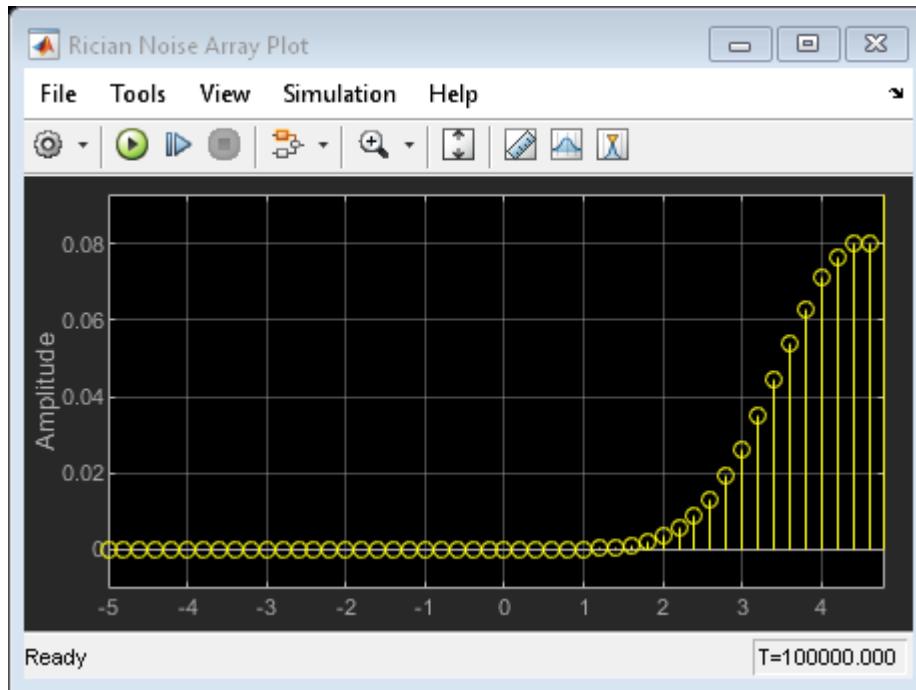
The **Uniform Noise** MATLAB Function block maps the **Noise lower bound** parameter to lb and the **Noise upper bound** parameter to ub , and defines the function

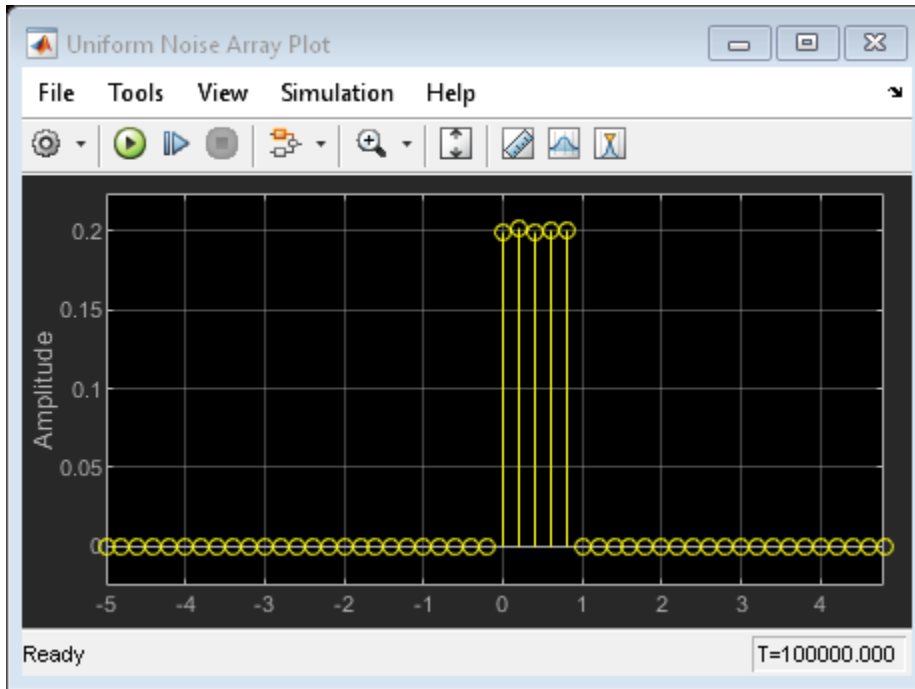
$$y = lb + (ub - lb) * rand(sp, 1)$$

The model generates these histogram plots to show the noise distribution across the spectrum for each noise generator.









For further exploration, open the model and adjust one of the noise generation settings. For example, the Rician noise generator has a K-factor of 10, which causes the mean value of the noise to be larger than that of the Rayleigh distributed noise. Double-click the Rician Noise MATLAB Function block to open the block mask and change the K-factor from 10 to 2. Rerun the model to see the noise spectrum shift.

Visualize Effects of Frequency-Selective Fading

FSK Modulation in Fading Channel

Pass an FSK signal through a Rayleigh multipath fading channel. Change the signal bandwidth to observe the impact of the fading channel on the FSK spectrum.

FSK Modulation in Flat Fading

Set modulation order to 4, the modulated symbol rate to 45 bps, and the frequency separation to 200 Hz.

```
M = 4;                % Modulation order
symbolRate = 45;     % Symbol rate (bps)
freqSep = 200;       % Frequency separation (Hz)
```

Calculate the samples per symbol parameter, `sampPerSym`, as a function of the modulation order, frequency separation, and symbol rate. To avoid output signal aliasing, the product of `sampPerSym` and `symbolRate` must be greater than the product of `M` and `freqSep`. Calculate the sample rate of the FSK output signal.

```
sampPerSym = ceil(M*freqSep/symbolRate);
fsamp = sampPerSym*symbolRate;
```

Create an FSK modulator.

```
fskMod = comm.FSKModulator(M, ...
    'FrequencySeparation', freqSep, ...
    'SamplesPerSymbol', sampPerSym, ...
    'SymbolRate', symbolRate);
```

Set the path delays and average path gains for the fading channel.

```
pathDelays = [0 3 10]*1e-6; % Discrete delays of three-path channel (s)
avgPathGains = [0 -3 -6]; % Average path gains (dB)
```

By convention, the delay of the first path is typically set to zero. For subsequent paths, a 1 microsecond delay corresponds to a 300 m difference in path length. The path delays and path gains specify the average delay profile of the channel.

Create a Rayleigh channel using the defined parameters. Set the `Visualization` property to display the impulse and frequency responses.

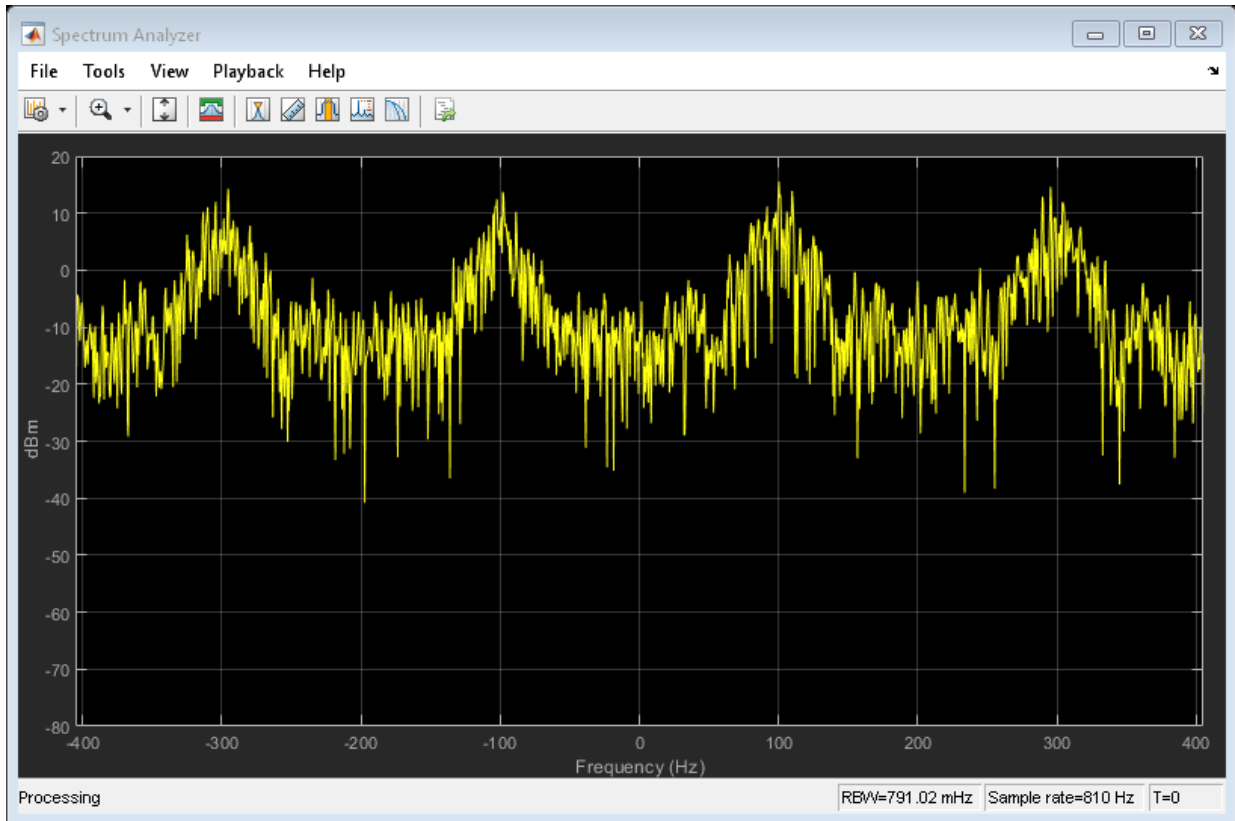
```
channel = comm.RayleighChannel(...  
    'SampleRate',fsamp, ...  
    'PathDelays',pathDelays, ...  
    'AveragePathGains',avgPathGains, ...  
    'MaximumDopplerShift',0.01, ...  
    'Visualization','Impulse and frequency responses', ...  
    'SamplesToDisplay','10%');
```

Generate random data symbols and apply FSK modulation.

```
data = randi([0 3],2000,1);  
modSig = fskMod(data);
```

Plot the spectrum of the FSK modulated signal.

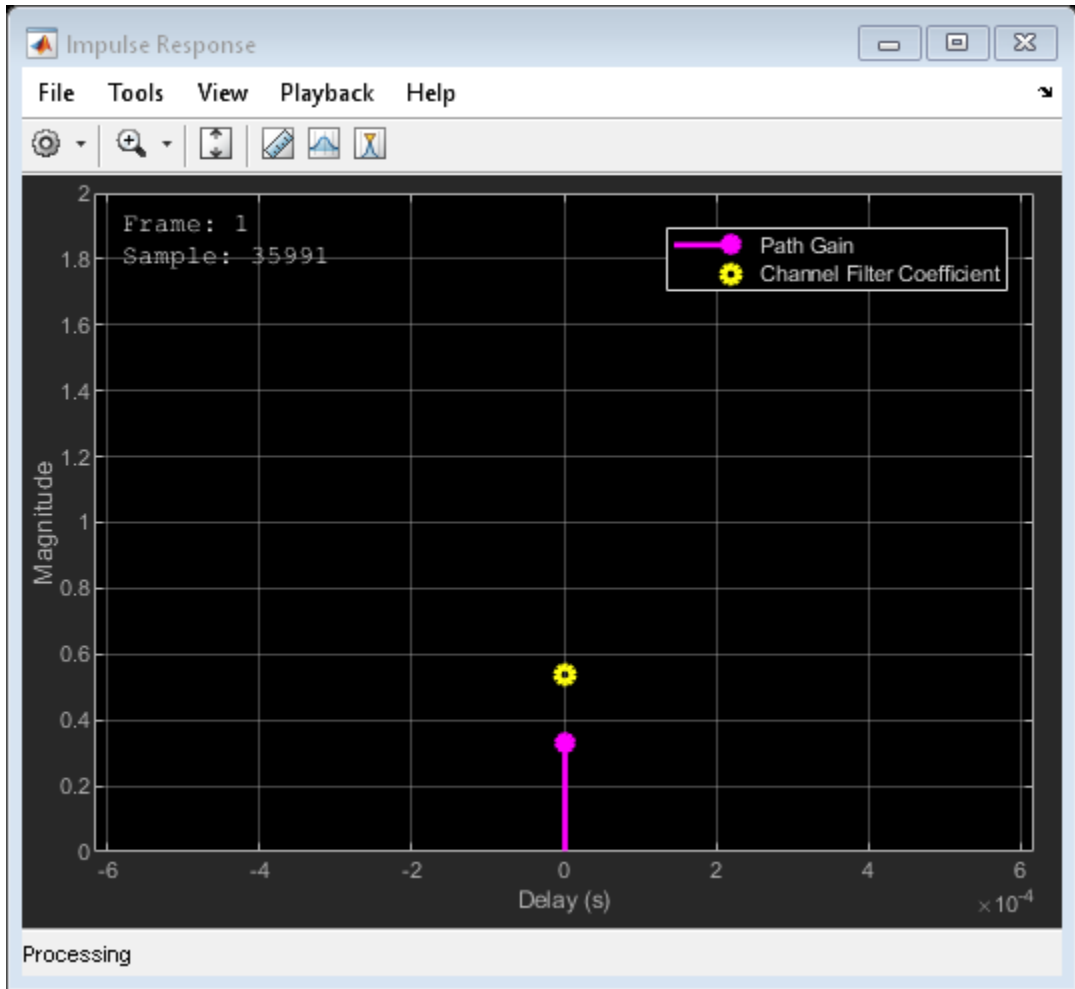
```
spectrum = dsp.SpectrumAnalyzer('SampleRate',fsamp);  
spectrum(modSig)
```

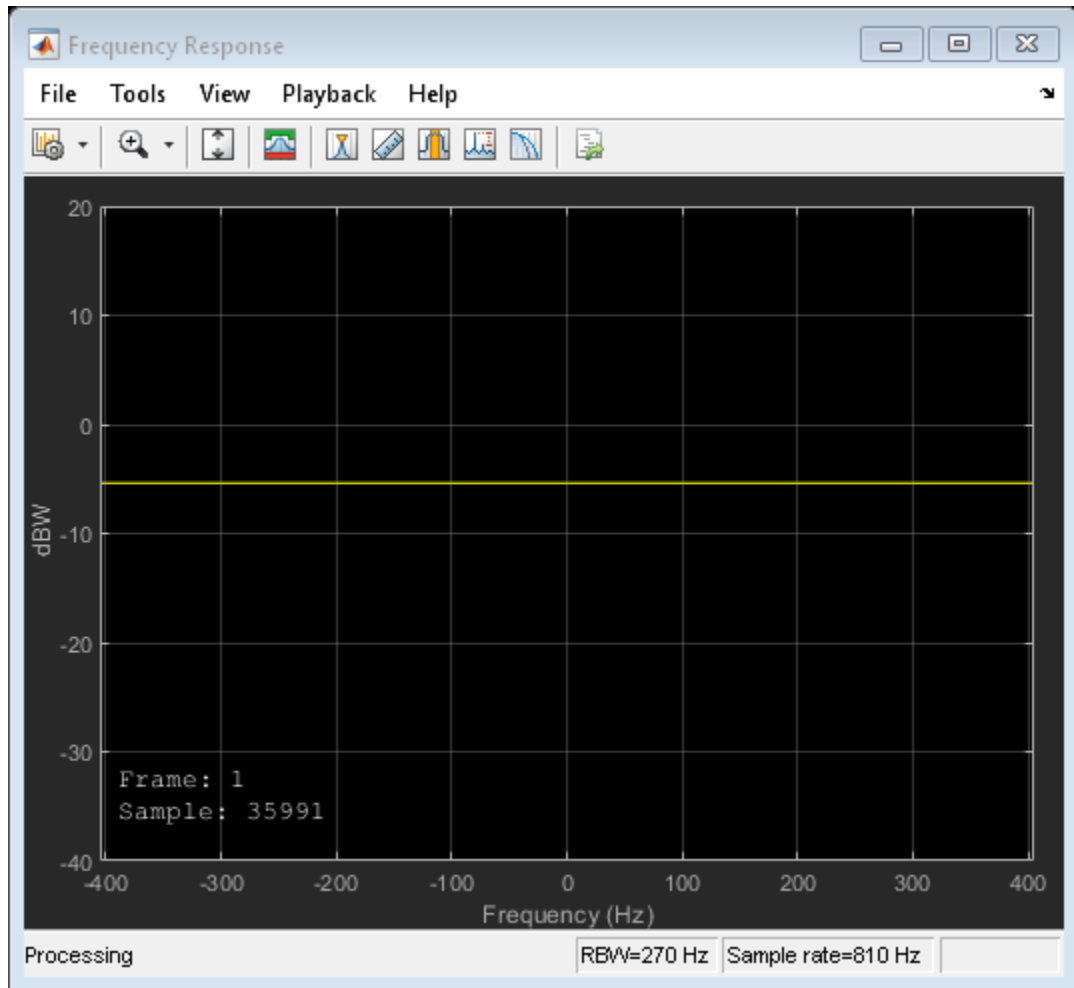



The modulated signal is composed of four tones each having approximately 20 dBm peak power separated by 200 Hz.

Pass the signal through the Rayleigh fading channel and apply AWGN having a 25 dB signal-to-noise ratio.

```
snrdB = 25;  
rxSig = awgn(channel(modSig),snrdB);
```

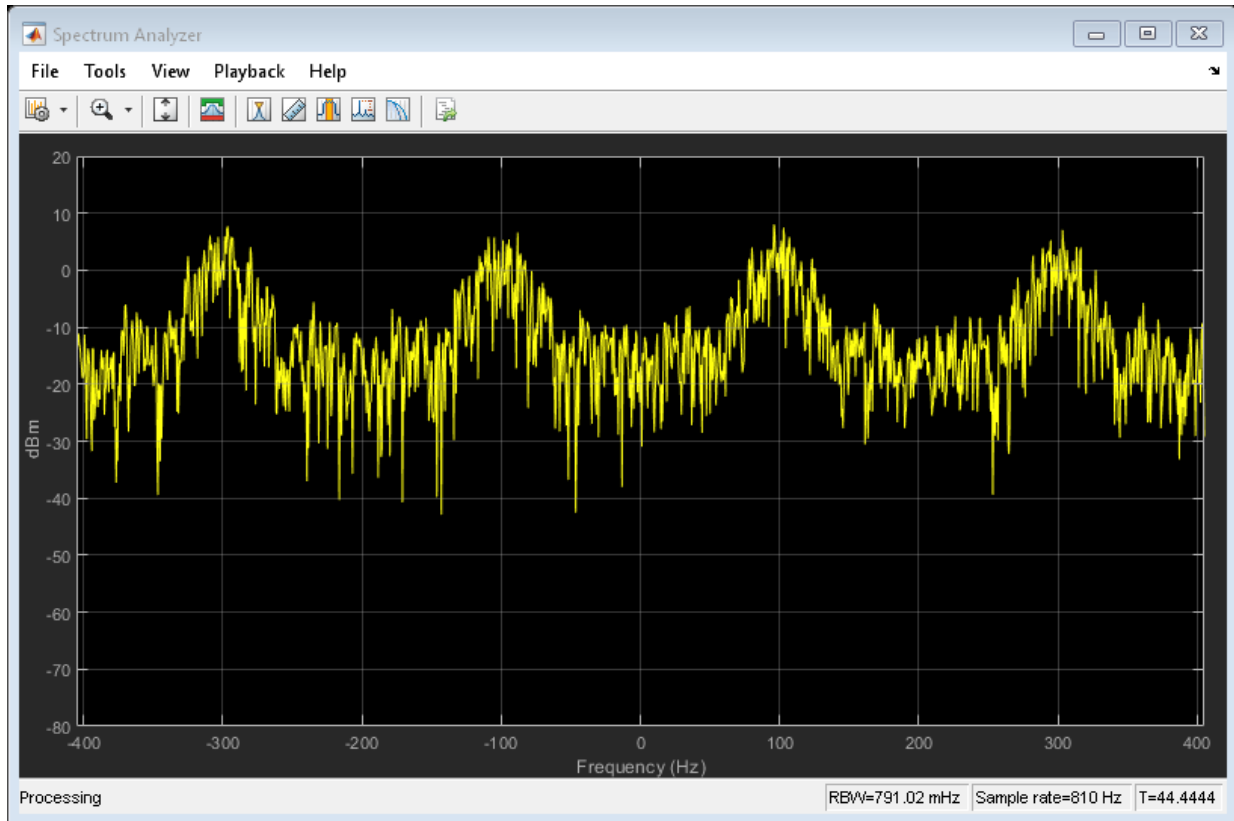




The impulse and frequency responses show that the channel behaves as though it were flat. This is because the signal bandwidth, 800 Hz, is much smaller than the coherence bandwidth, 50 kHz.

Plot the received signal spectrum.

```
spectrum(rxSig)
```



The four tones comprising the FSK signal maintain the same frequency separation and peak power levels relative to each other. The absolute peak power levels have decreased due to the fading channel.

FSK Modulation in Frequency-Selective Fading

Increase the symbol rate to 45 kbps and the frequency separation to 200 kHz. Calculate the new samples per symbol and sample rate parameters.

```
symbolRate = 45e3;
freqSep = 200e3;
sampPerSym = ceil(M*freqSep/symbolRate);
fsamp = sampPerSym*symbolRate;
```

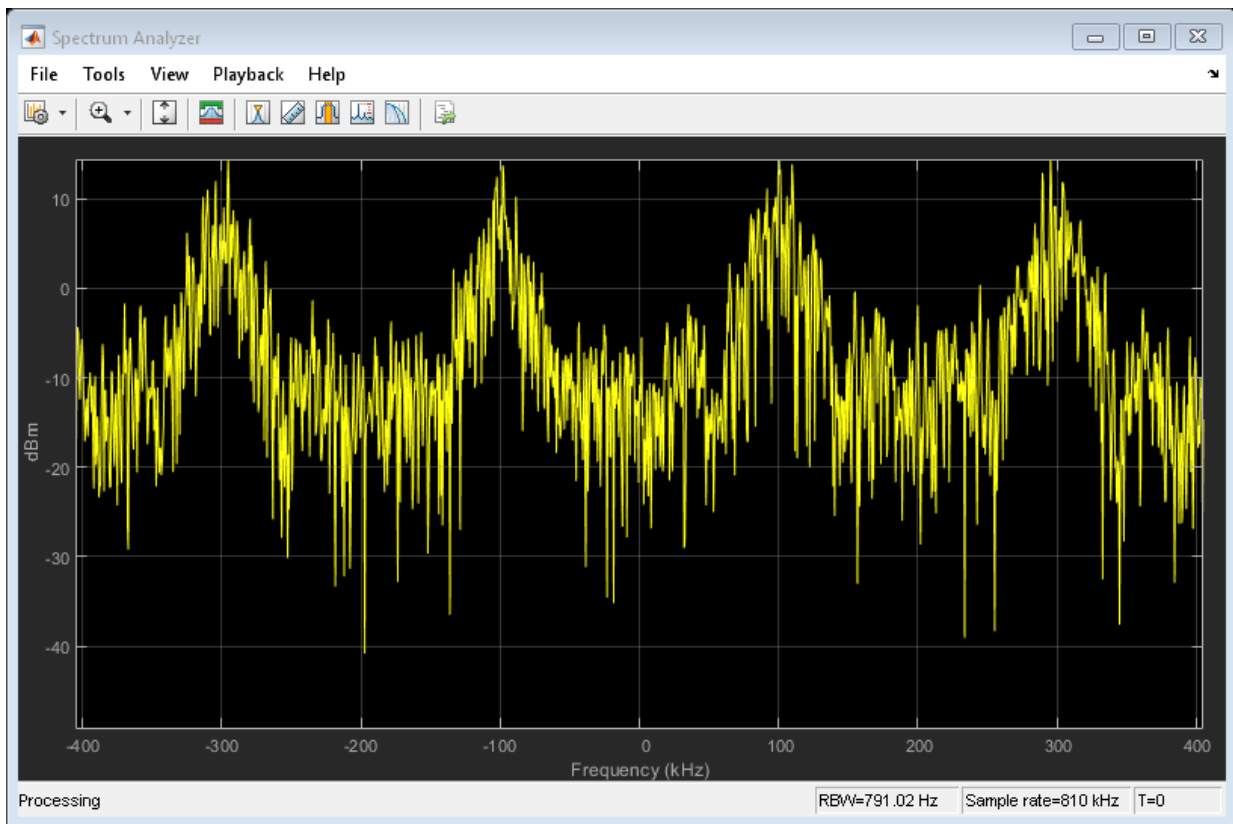
Update the FSK modulator properties.

```
release(fskMod)
fskMod.SymbolRate = symbolRate;
fskMod.FrequencySeparation = freqSep;
```

Update the spectrum analyzer sample rate property, `sa.SampleRate`. Apply FSK modulation and plot the resulting spectrum.

```
release(spectrum)
spectrum.SampleRate = sampPerSym*symbolRate;
```

```
modSig = fskMod(data);
spectrum(modSig)
```

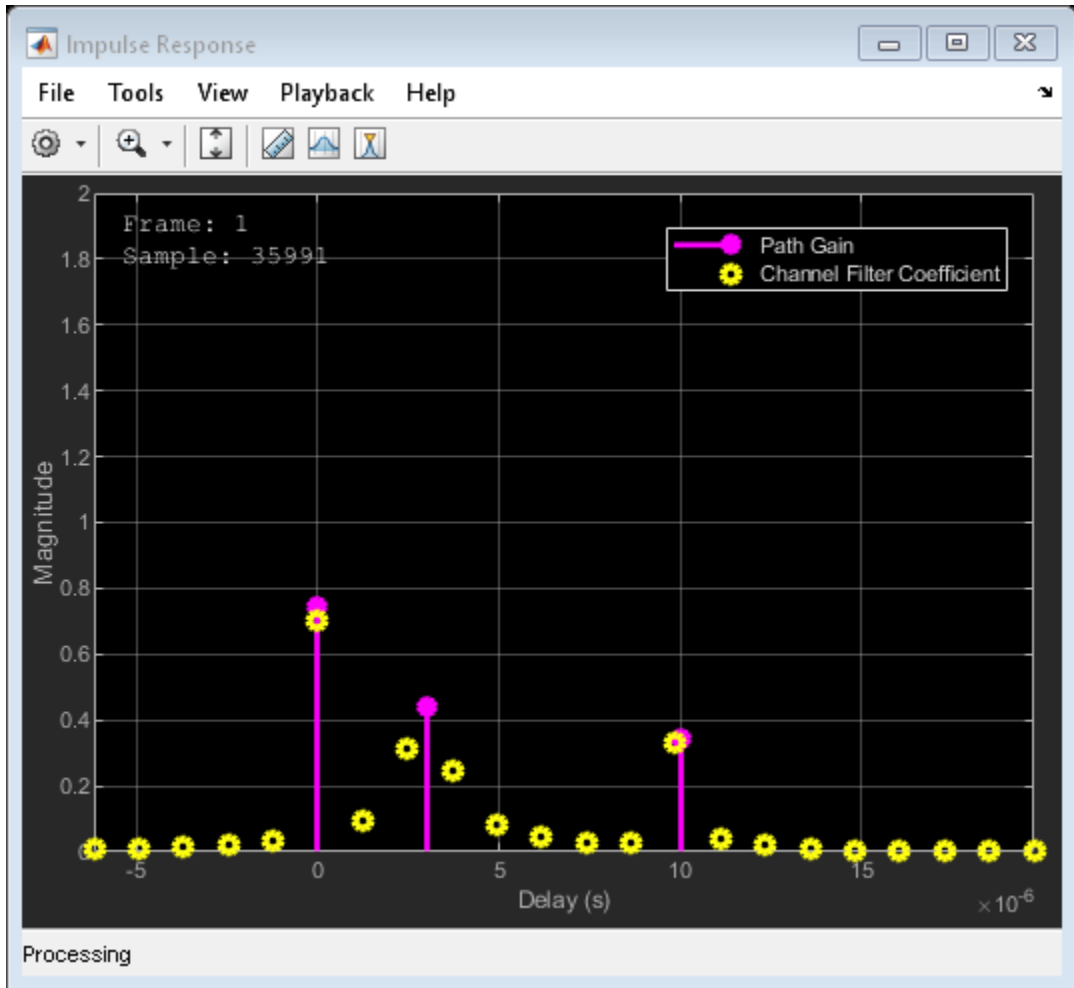


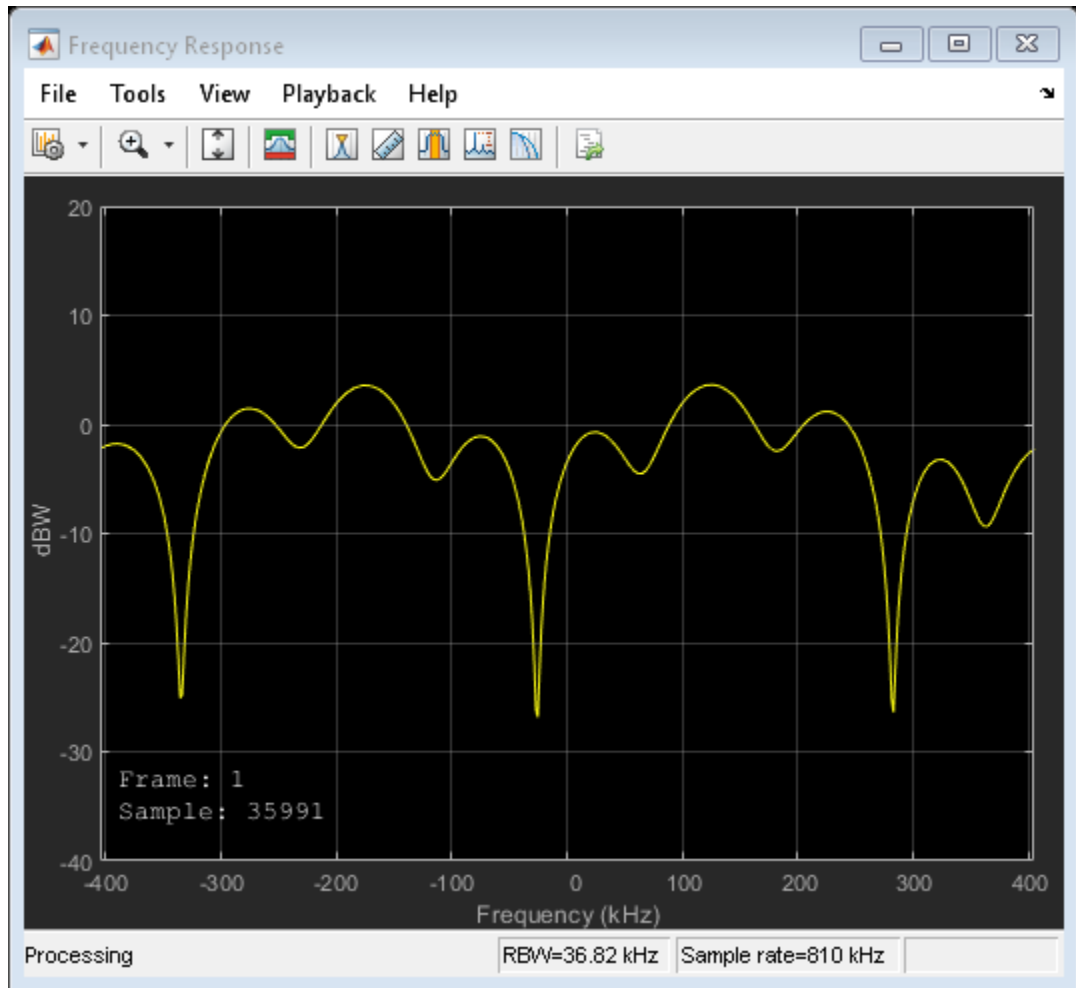
The spectrum has the same shape as in the flat-fading case but the four tones are now separated by 200 kHz.

Update the channel sample rate property. Pass the signal through the Rayleigh fading channel and apply AWGN.

```
release(channel)
channel.SampleRate = fsamp;

rxSig = awgn(channel(modSig),25);
```

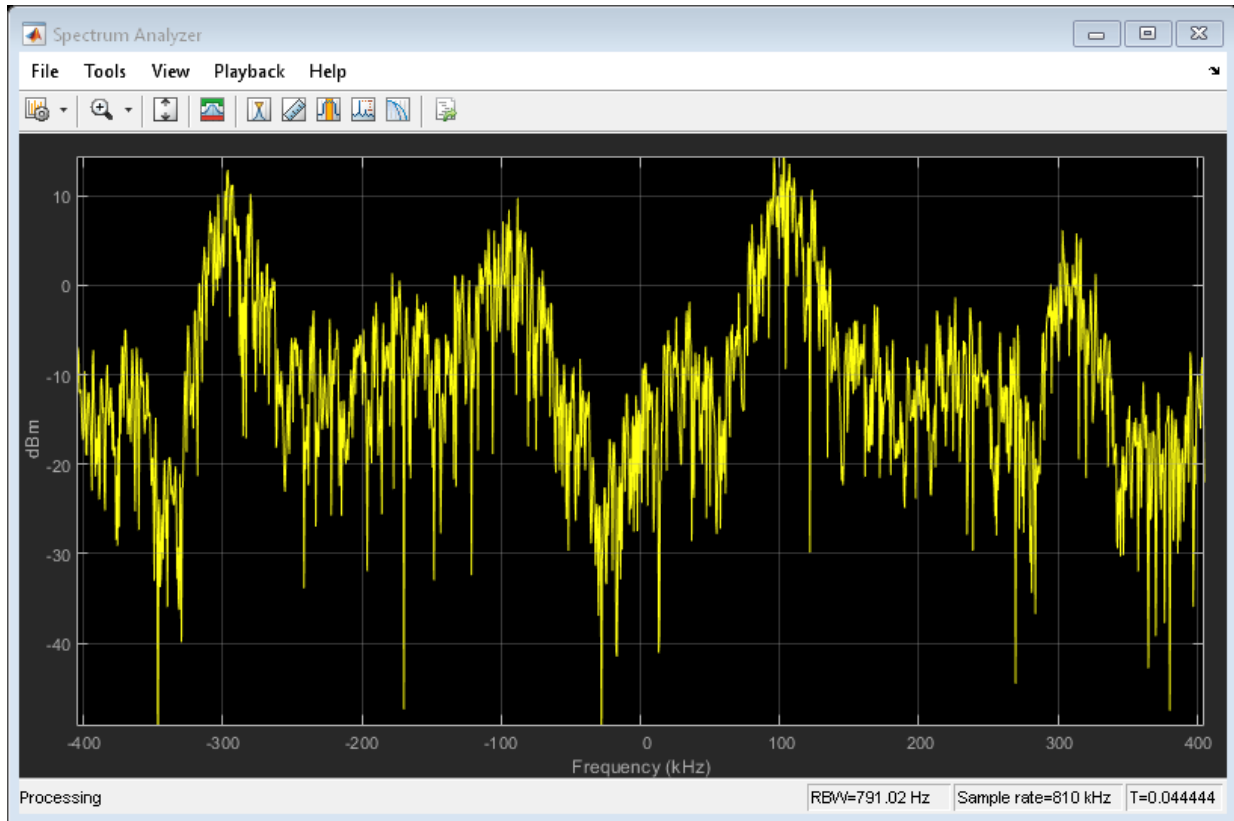




The impulse and frequency responses show that the multipath fading is frequency selective.

Plot the received signal spectrum.

```
spectrum(rxSig)
```



There are still four identifiable tones but their relative peak power levels differ due to the frequency-selective fading. The signal bandwidth, 800 kHz, is larger than the coherence bandwidth, 50 kHz.

QPSK Modulation in Fading Channel

Pass a QPSK signal through a Rayleigh multipath fading channel. Change the signal bandwidth to observe the impact of the fading channel on the QPSK constellation.

QPSK Modulation in Flat Fading

Set the symbol rate parameter to 500 bps.

```
symbolRate = 500;
```


Generate random data symbols and apply QPSK modulation.

```
data = randi([0 3],10000,1);  
modSig = pskmod(data,4,pi/4,'gray');
```

Set the path delays and average path gains for the fading channel.

```
pathDelays = [0 3 10]*1e-6;    % Discrete delays of three-path channel (s)  
avgPathGains = [0 -3 -6];    % Average path gains (dB)
```

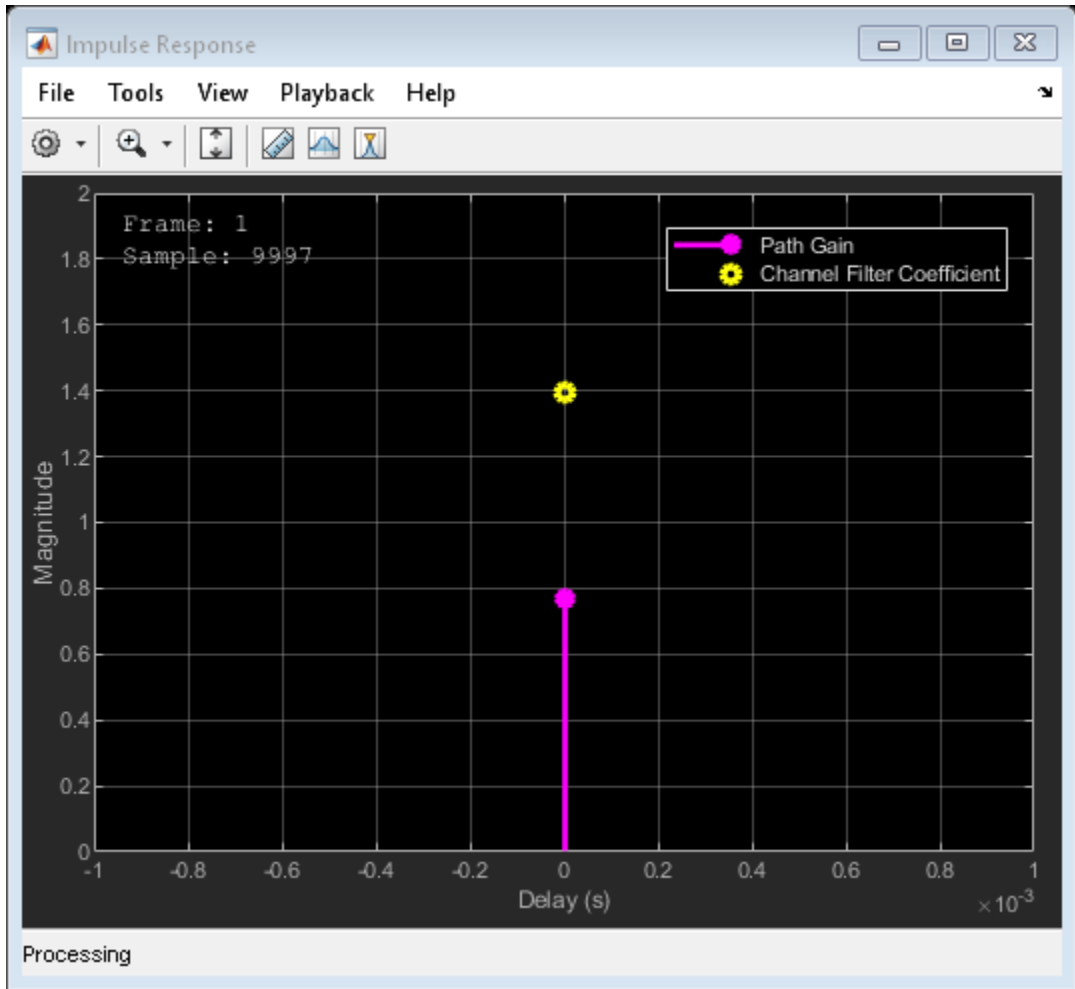
By convention, the delay of the first path is typically set to zero. For subsequent paths, a 1 microsecond delay corresponds to a 300 meter difference in path length. The path delays and path gains specify the average delay profile of the channel.

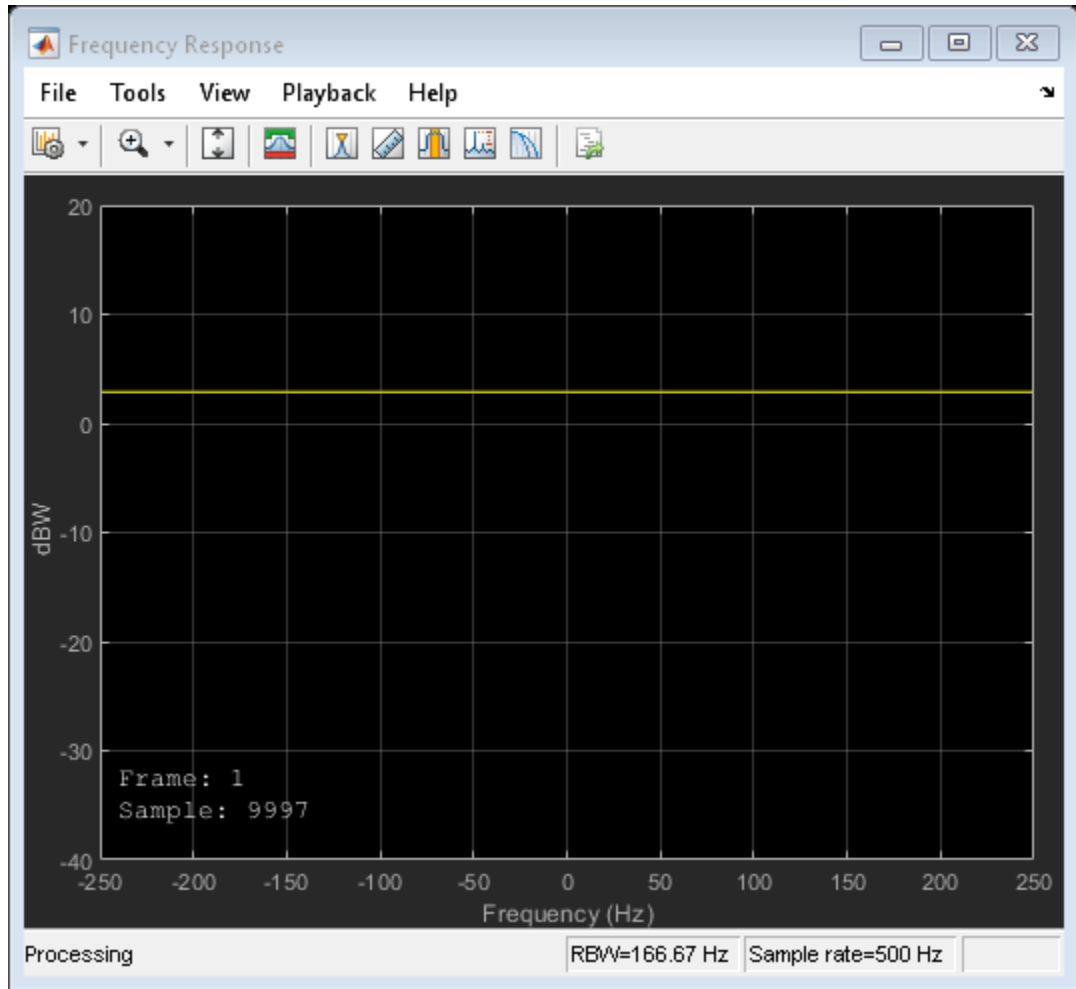
Create a Rayleigh channel using the defined parameters. Set the **Visualization** property to display the impulse and frequency responses.

```
fsamp = symbolRate;  
channel = comm.RayleighChannel(...  
    'SampleRate',fsamp, ...  
    'PathDelays',pathDelays, ...  
    'AveragePathGains',avgPathGains, ...  
    'MaximumDopplerShift',0.01, ...  
    'Visualization','Impulse and frequency responses');
```

Pass the signal through the Rayleigh channel and apply AWGN.

```
rxSig = awgn(channel(modSig),25);
```

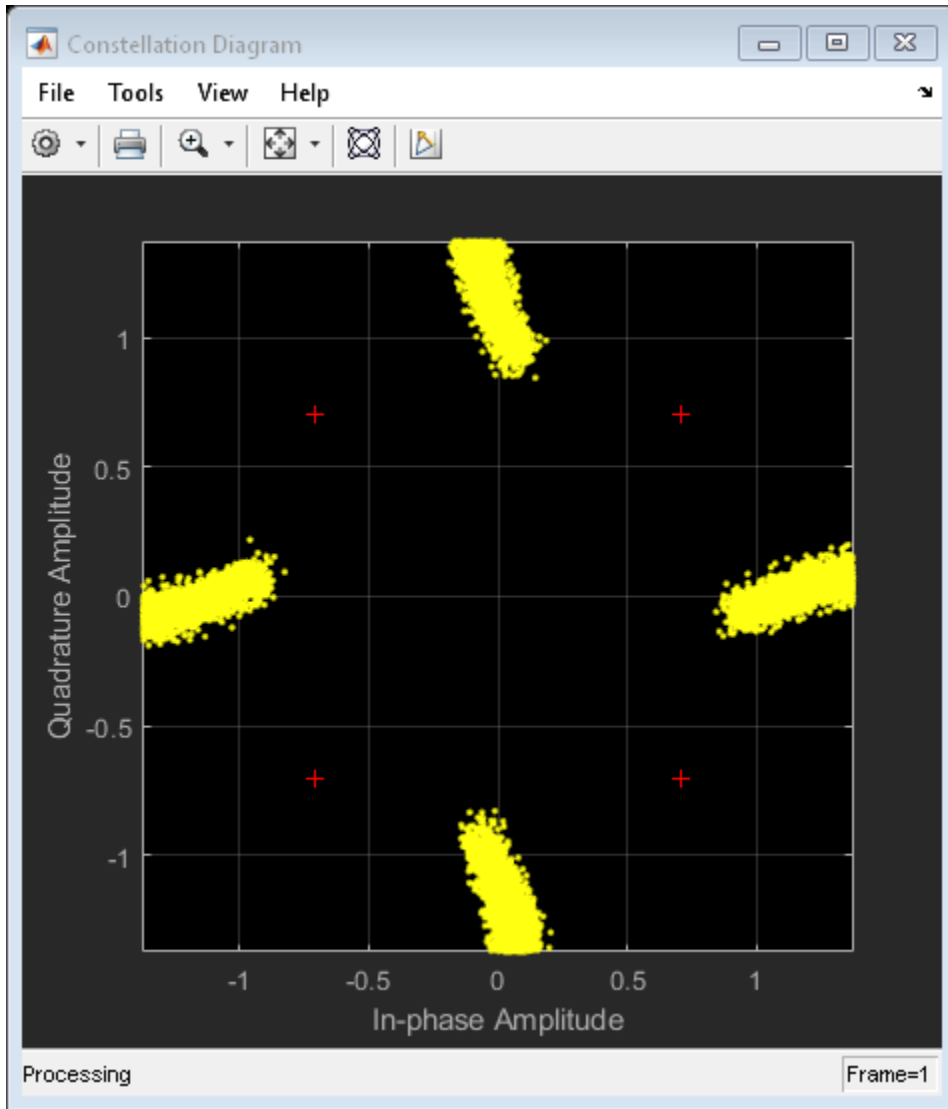




The impulse and frequency responses show that the channel behaves as though it were flat. This is because the signal bandwidth, 500 Hz, is much smaller than the coherence bandwidth, 50 kHz. Alternatively, the delay span of the channel (10 microseconds) is much smaller than the QPSK symbol period (2 milliseconds) so the resultant bandlimited impulse response is approximately flat.

Plot the constellation.

```
constDiagram = comm.ConstellationDiagram;  
constDiagram(rxSig)
```

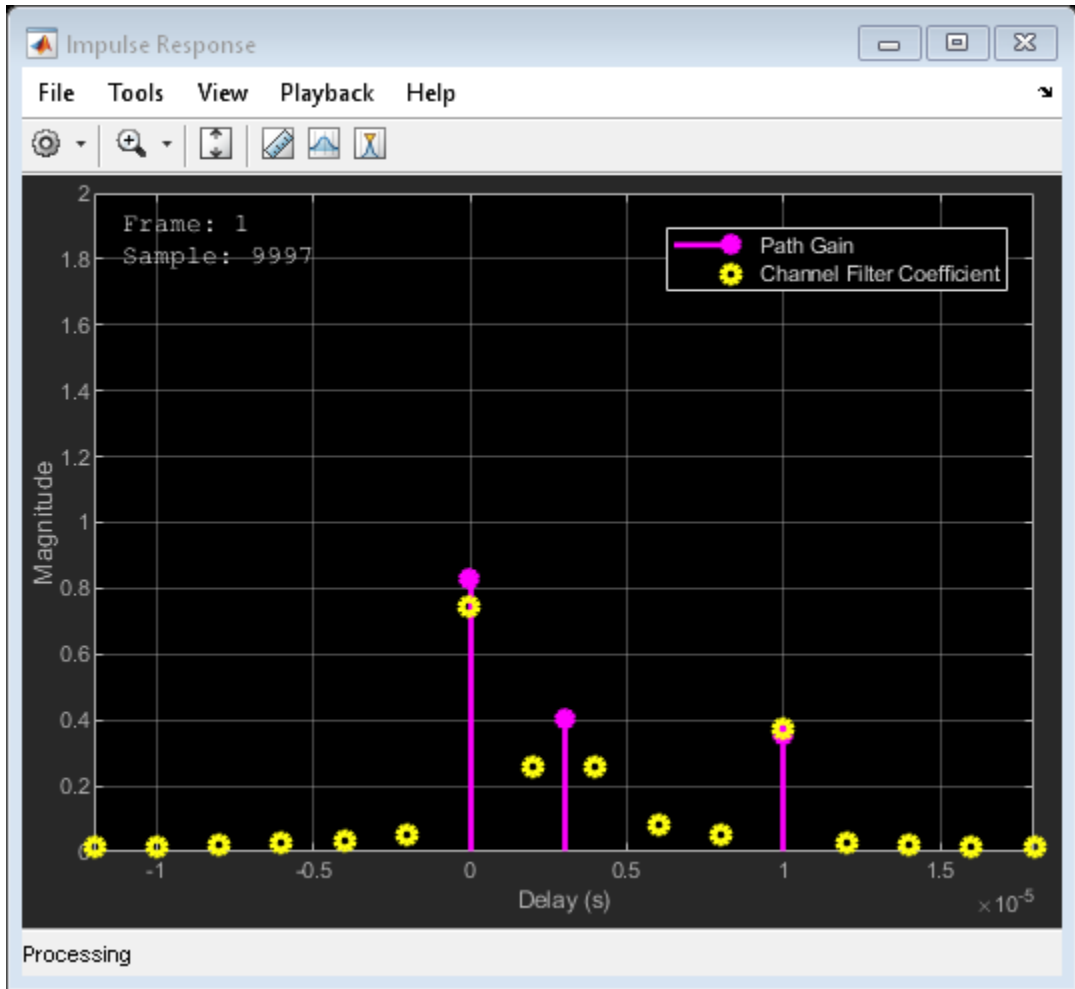


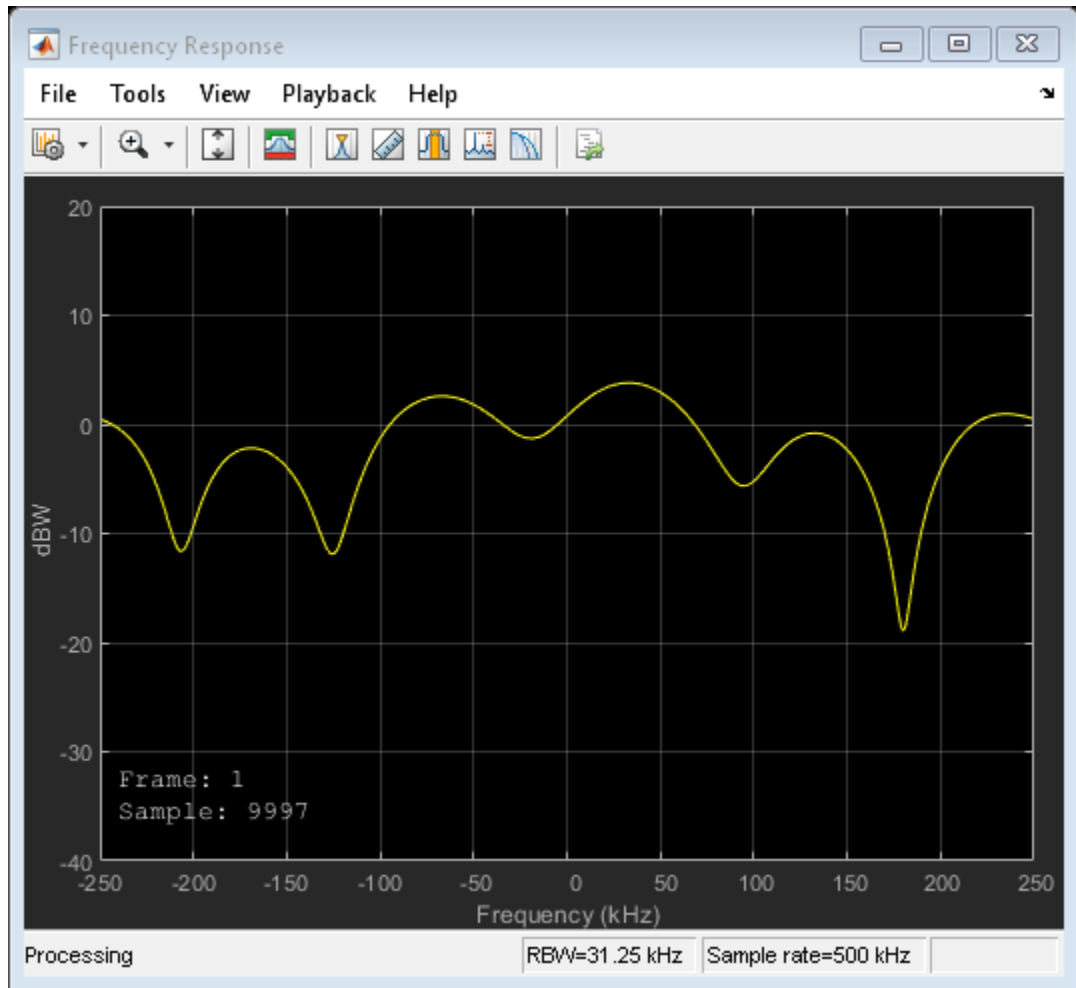
The QPSK constellation shows the effects of the fading channel; however, the signal still has four identifiable states.

QPSK Modulation in Frequency-Selective Fading

Increase the symbol rate to 500 kbps and update the related channel property. Pass the signal through the Rayleigh channel and apply AWGN.

```
symbolRate = 500e3;  
  
release(channel)  
channel.SampleRate = symbolRate;  
  
rxSig = awgn(channel(modSig),25);
```

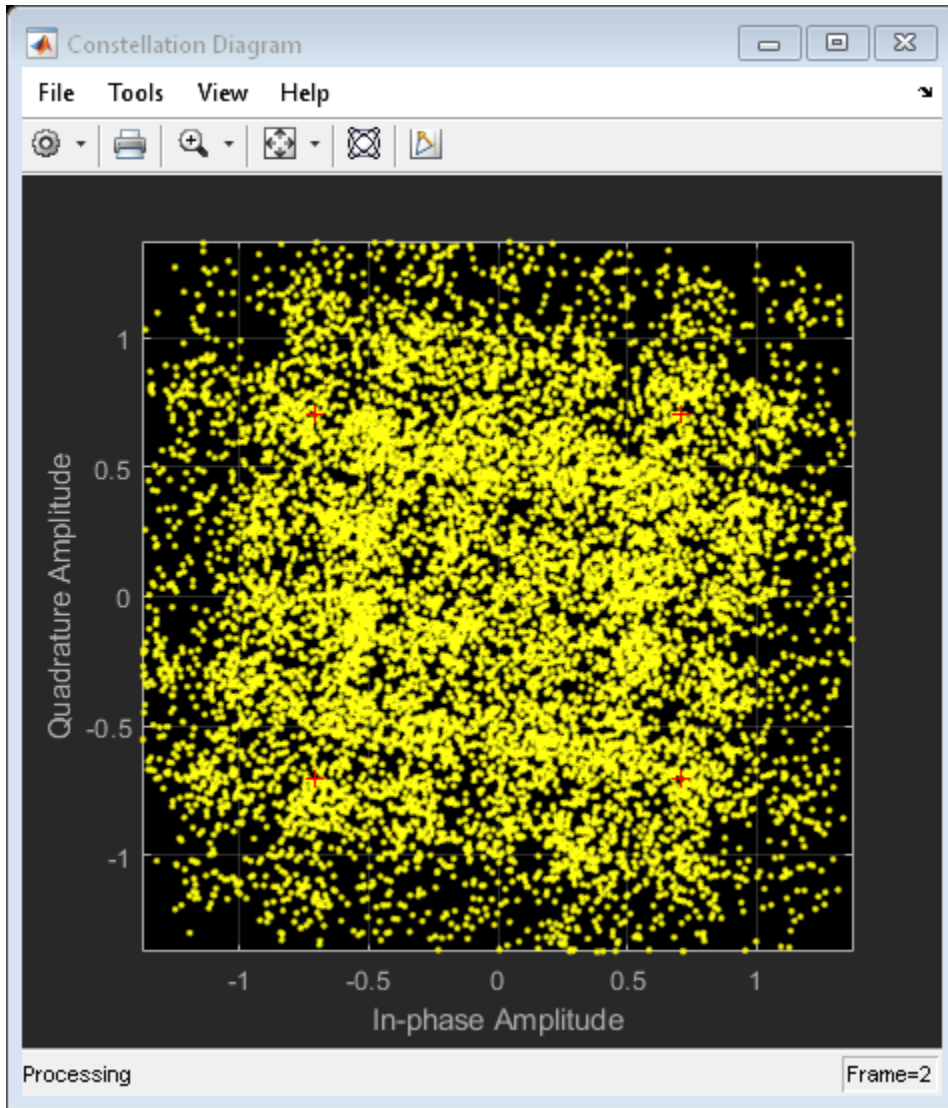




The impulse and frequency responses show that the multipath fading is frequency selective.

Plot the constellation.

```
constDiagram(rxSig)
```



As the signal bandwidth is increased from 500 Hz to 500 kHz, the signal becomes highly distorted. This distortion is due to the intersymbol interference (ISI) that comes from time dispersion of the wideband signal. The delay span of the channel (10 microseconds) is now larger than the QPSK symbol period (2 microseconds) so the resultant bandlimited

impulse response is no longer flat. Alternatively, the signal bandwidth is much larger than the coherence bandwidth, 50 kHz.

Correct Phase and Frequency Offset for 16-QAM Using Coarse and Fine Synchronization

Compensation of significant phase and frequency offsets for a 16-QAM signal in an AWGN channel is accomplished in two steps. First, correct the coarse frequency offset using the estimate provided by the coarse frequency compensator, and then fine-tune the correction using carrier synchronization. Because of the coarse frequency correction, the carrier synchronizer converges quickly even though the normalized bandwidth is set to a low value. Lower normalized bandwidth values enable better correction for small residual carrier offsets. After applying phase and frequency offset corrections to the received signal, resolve phase ambiguity using the preambles.

Define the simulation parameters.

```
fs = 10000;      % Sample rate (Hz)
sps = 4;        % Samples per symbol
M = 16;         % Modulation order
k = log2(M);    % Bits per symbol
rng(1996)       % Set seed for repeatable results
barker = comm.BarkerCode(...
    'Length',13,'SamplesPerFrame',13); % For preamble
msgLen = 1e4;
numFrames = 10;
frameLen = msgLen/numFrames;
```

Generate data payloads and add the preamble to each frame. The preamble is later used for phase ambiguity resolution.

```
preamble = (1+barker())/2; % Length 13, unipolar
data = zeros(msgLen, 1);
for idx = 1 : numFrames
    payload = randi([0 M-1],frameLen-barker.Length,1);
    data((idx-1)*frameLen + (1:frameLen)) = [preamble; payload];
end
```

Create a System object for the transmit pulse shape filtering, the receive pulse shape filtering, the QAM coarse frequency compensation, the carrier synchronization, and a constellation diagram.

```
txFilter = comm.RaisedCosineTransmitFilter( ...
    'OutputSamplesPerSymbol',sps);
rxFilter = comm.RaisedCosineReceiveFilter(...
```

```

    'InputSamplesPerSymbol',sps,'DecimationFactor',sps);
coarse = comm.CoarseFrequencyCompensator('SampleRate',fs, ...
    'FrequencyResolution',10);
fine = comm.CarrierSynchronizer( ...
    'DampingFactor',0.4,'NormalizedLoopBandwidth',0.001, ...
    'SamplesPerSymbol',1,'Modulation','QAM');
axislimits = [-6 6];
constDiagram = comm.ConstellationDiagram('ReferenceConstellation',qammod(0:M-1,M), ...
    'ChannelNames',{'Before convergence','After convergence'}, ...
    'ShowLegend',true,'XLimits',axislimits,'YLimits',axislimits);

```

Also create a System object for the AWGN channel, and the phase and frequency offset to add impairments to the signal. A phase offset greater than 90 degrees is added to induce a phase ambiguity that results in a constellation quadrant shift.

```

ebn0 = 8;
freqoffset = 110;
phaseoffset = 110;
awgnChannel = comm.AWGNChannel('EbNo',ebn0, ...
    'BitsPerSymbol',k,'SamplesPerSymbol',sps);
pfo = comm.PhaseFrequencyOffset('FrequencyOffset',freqoffset, ...
    'PhaseOffset',phaseoffset,'SampleRate',fs);

```

Generate random data symbols, apply 16-QAM modulation, and pass the modulated signal through the transmit pulse shaping filter.

```

txMod = qammod(data,M);
txSig = txFilter(txMod);

```

Apply phase and frequency offsets using the pfo System object, and then pass the signal through an AWGN channel to add white Gaussian noise.

```

txSigOffset = pfo(txSig);
rxSig = awgnChannel(txSigOffset);

```

The coarse frequency compensator System object provides a rough correction for the frequency offset. For the conditions in this example, correcting the frequency offset of the received signal correction to within 10 Hz of the transmitted signal is sufficient.

```

syncCoarse = coarse(rxSig);

```

Pass the signal through the receive pulse shaping filter, and apply fine frequency correction.

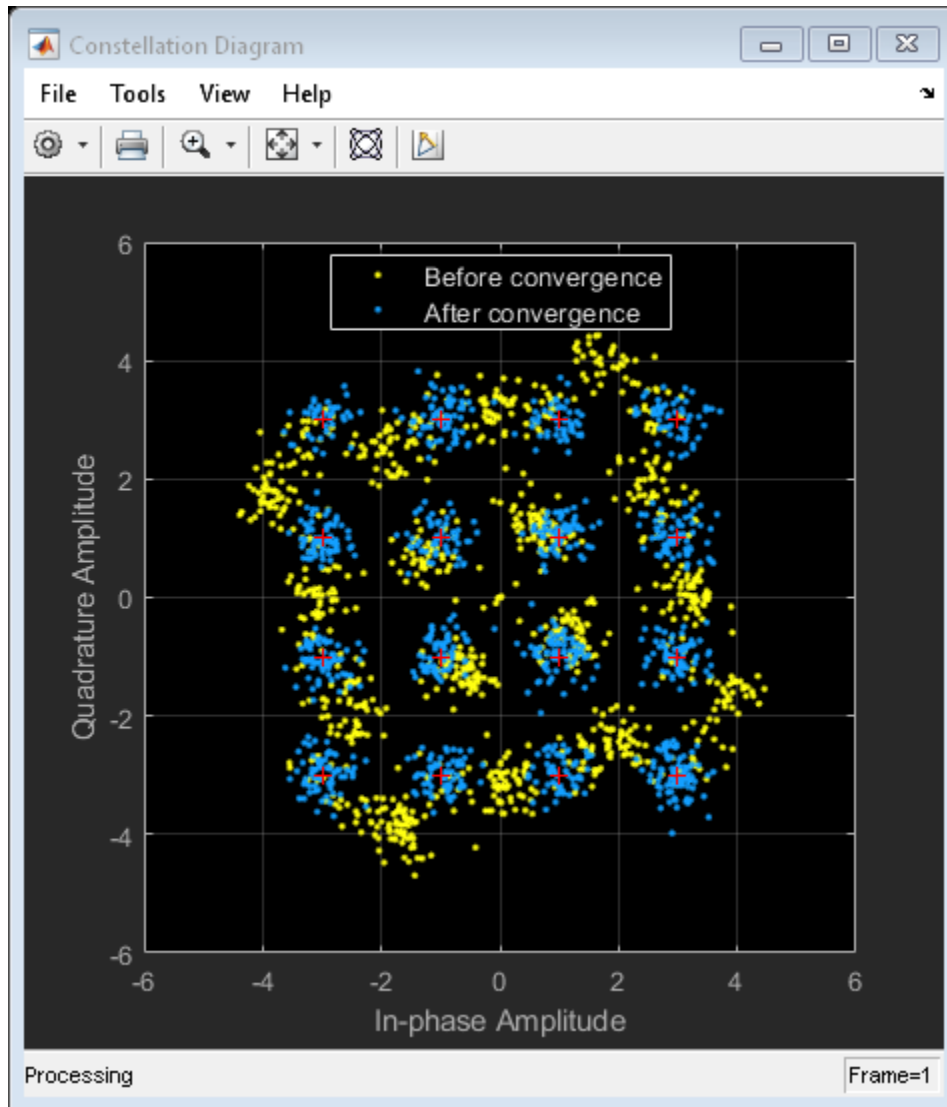
```

rxFiltSig = fine(rxFilter(syncCoarse));

```

Display the constellation diagram of the first and last 1000 symbols in the signal. Before convergence of the synchronization loop, the spiral nature of the diagram indicates that the frequency offset is not corrected. After the carrier synchronizer has converged to a solution, the symbols are aligned with the reference constellation.

```
constDiagram([rxFiltSig(1:1000) rxFiltSig(9001:end)])
```



Demodulate the signal. Account for the signal delay caused by the transmit and receive filters to align the received data with the transmitted data. Compute and display the total bit errors and BER. When checking the bit errors, use the later portion of the received signal to be sure the synchronization loop has converged.

```
rxData = qamdemod(rxFiltSig,M);
delay = (txFilter.FilterSpanInSymbols + rxFilter.FilterSpanInSymbols) / 2;
idxSync = 2000; % Check BER for the received signal after the synchronization loop has
[syncDataTtlErr,syncDataBER] = biterr(data(idxSync:end-delay),rxData(idxSync+delay:end-
syncDataTtlErr = 16116
syncDataBER = 0.5042
```

Depending on the random data used, there may be bit errors resulting from phase ambiguity in the received signal after the synchronization loop converges and locks. In this case, you can use the preamble to determine and then remove the phase ambiguity from the synchronized signal to reduce bit errors. If phase ambiguity is minimal, the number of bit errors may be unchanged.

```
idx = 9000 + (1:barker.Length);
phOffset = angle(txMod(idx) .* conj(rxFiltSig(idx+delay)));

phOffsetEst = mean(phOffset);
disp(['Phase offset = ',num2str(rad2deg(phOffsetEst)), ' degrees'])

Phase offset = -90.1401 degrees

resPhzSig = exp(1i*phOffsetEst) * rxFiltSig;
```

Demodulate the signal after resolving the phase ambiguity. Recompute the total bit errors and BER.

```
resPhzData = qamdemod(resPhzSig,M);
[resPhzTtlErr,resPhzBER] = biterr(data(idxSync:end-delay),resPhzData(idxSync+delay:end-
resPhzTtlErr = 5
resPhzBER = 1.5643e-04
```

Adjust Carrier Synchronizer Damping Factor to Correct Frequency Offset

Attempt to correct for a frequency offset using the carrier synchronizer object. Increase the damping factor of the synchronizer and determine if the offset was corrected.

Set the modulation order, sample rate, frequency offset, and signal-to-noise ratio parameters.

```
M = 8;
fs = 1e6;
foffset = 1000;
snrdb = 20;
```

Create a phase frequency offset object to introduce a frequency offset to a modulated signal. Create a constellation diagram object.

```
pfo = comm.PhaseFrequencyOffset('SampleRate',fs, ...
    'FrequencyOffset',foffset);
constDiagram = comm.ConstellationDiagram('ReferenceConstellation',pskmod(0:M-1,M,pi/M))
```

Create a carrier synchronizer object to correct for the frequency offset.

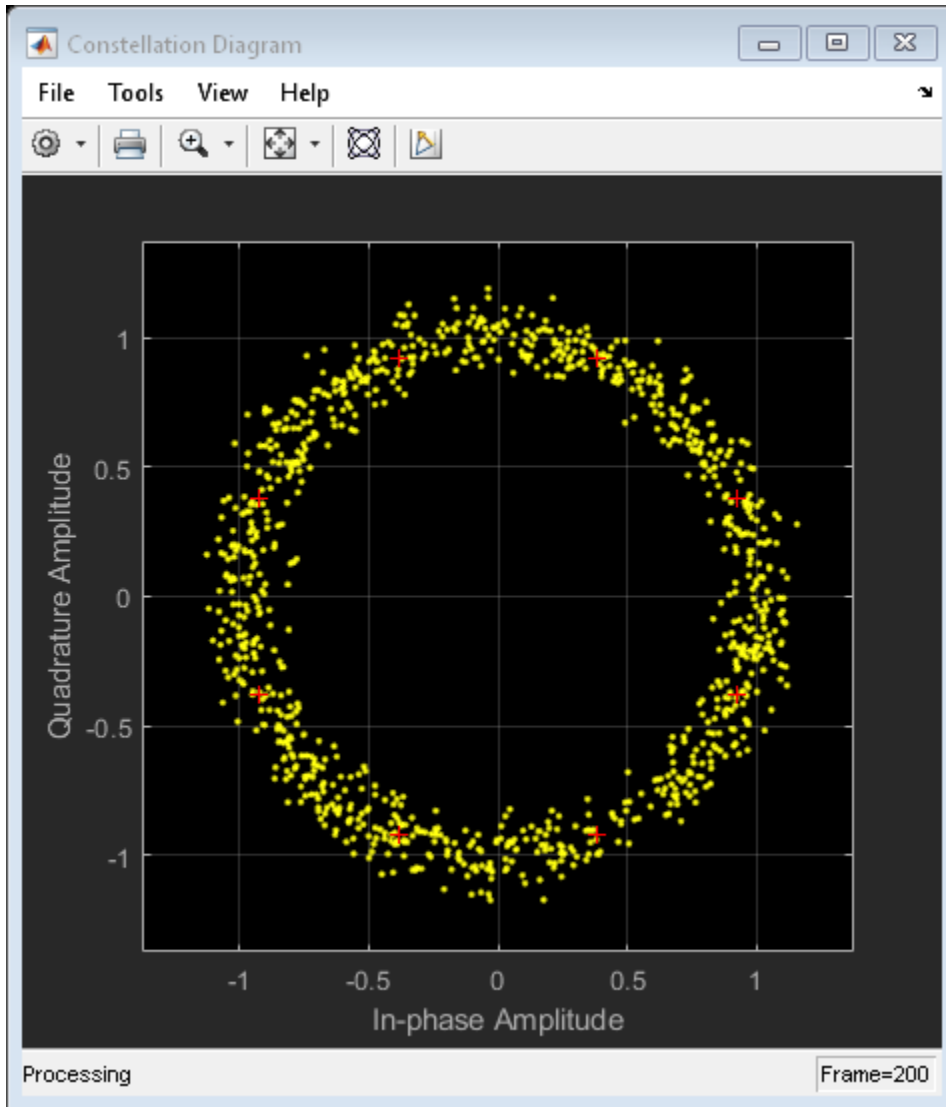
```
carriersync = comm.CarrierSynchronizer('Modulation','8PSK', ...
    'DampingFactor',0.05,'NormalizedLoopBandwidth',0.01);
```

The main processing loop includes these steps:

- Generate random data.
- Apply 8-PSK modulation.
- Introduce a frequency offset.
- Pass the signal through an AWGN channel.
- Correct for the frequency offset.
- Display the constellation diagram.

```
for k = 1:200
    data = randi([0 M-1],1000,1);
    modSig = pskmod(data,M);
    txSig = pfo(modSig);
    rxSig = awgn(txSig,snrdb);
    syncOut = carriersync(rxSig);
```

```
constDiagram(syncOut)  
end
```



The constellation points cannot be clearly identified indicating that the carrier synchronizer is unable to compensate for the frequency offset.

Determine the normalized pull-in range, the maximum frequency lock delay, and the maximum phase lock delay by using the `info` function.

```
syncInfo = info(carriersync)

syncInfo =

    struct with fields:

        NormalizedPullInRange: 0.0044
        MaxFrequencyLockDelay: 78.9568
        MaxPhaseLockDelay: 130
```

Convert the normalised pull-in range from radians to cycles. Compare the normalized frequency offset to the pull-in range.

```
[foffset/fs syncInfo.NormalizedPullInRange/(2*pi)]

ans =

    1.0e-03 *
    1.0000    0.7071
```

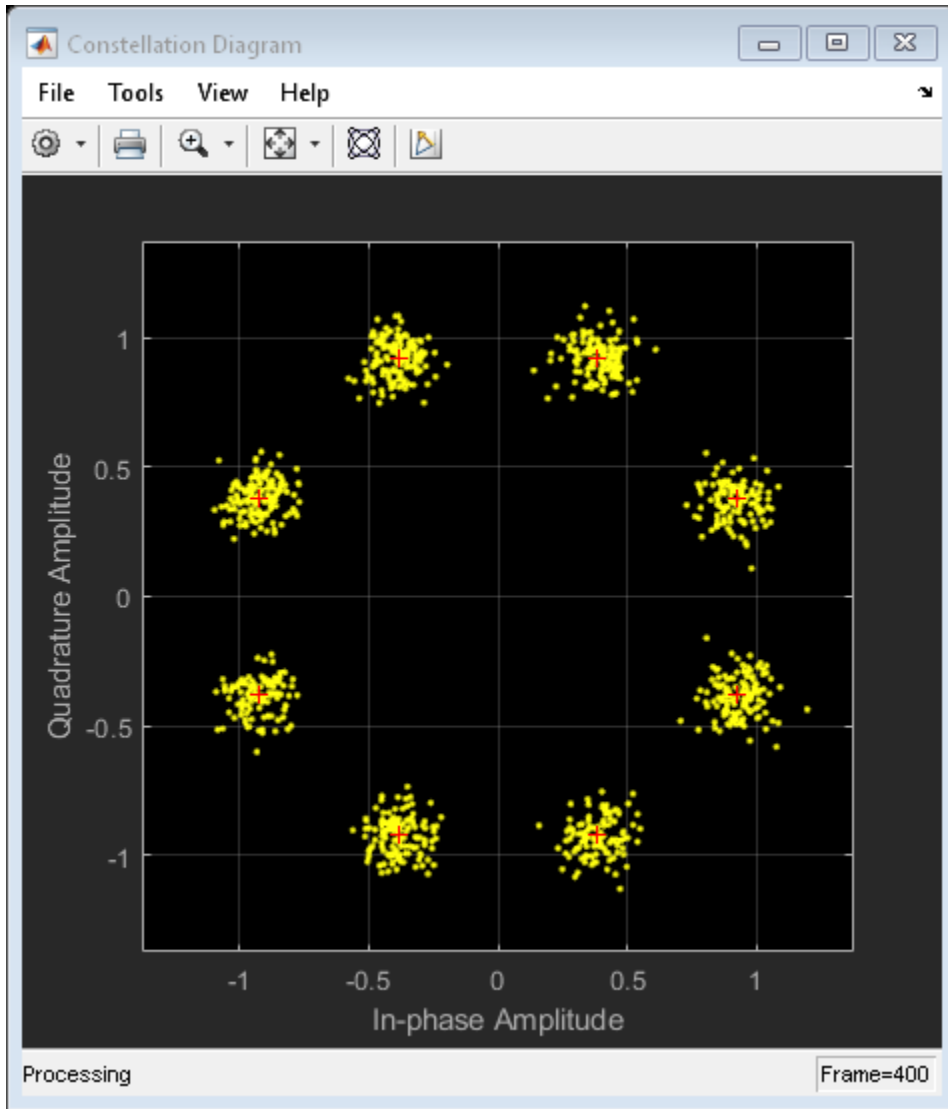
The offset is greater than the pull-in range. This is reason that the carrier synchronizer failed to correct the frequency offset.

Change the damping factor of the synchronizer to `0.707`.

```
carriersync.DampingFactor = 0.707;
```

Repeat the main processing loop.

```
for k = 1:200
    data = randi([0 M-1],1000,1);
    modSig = pskmod(data,M);
    txSig = pfo(modSig);
    rxSig = awgn(txSig,snrdb);
    syncOut = carriersync(rxSig);
    constDiagram(syncOut)
end
```



There are now eight observable clusters, which shows that the frequency offset was corrected.

Determine the new pull-in range. The normalized offset is less than the pull-in range. This explains why the carrier synchronizer was able to correct the offset.

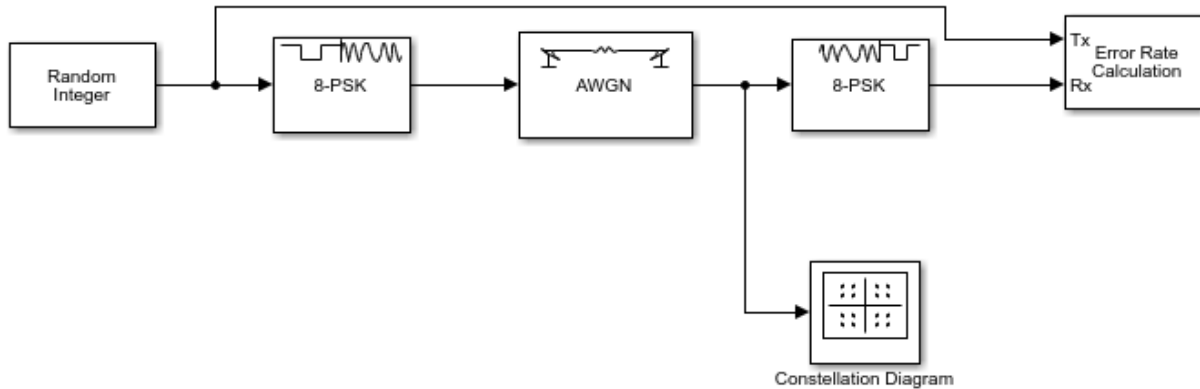
```
syncInfo = info(carriersync);  
[foffset/fs syncInfo.NormalizedPullInRange/(2*pi)]
```

```
ans =
```

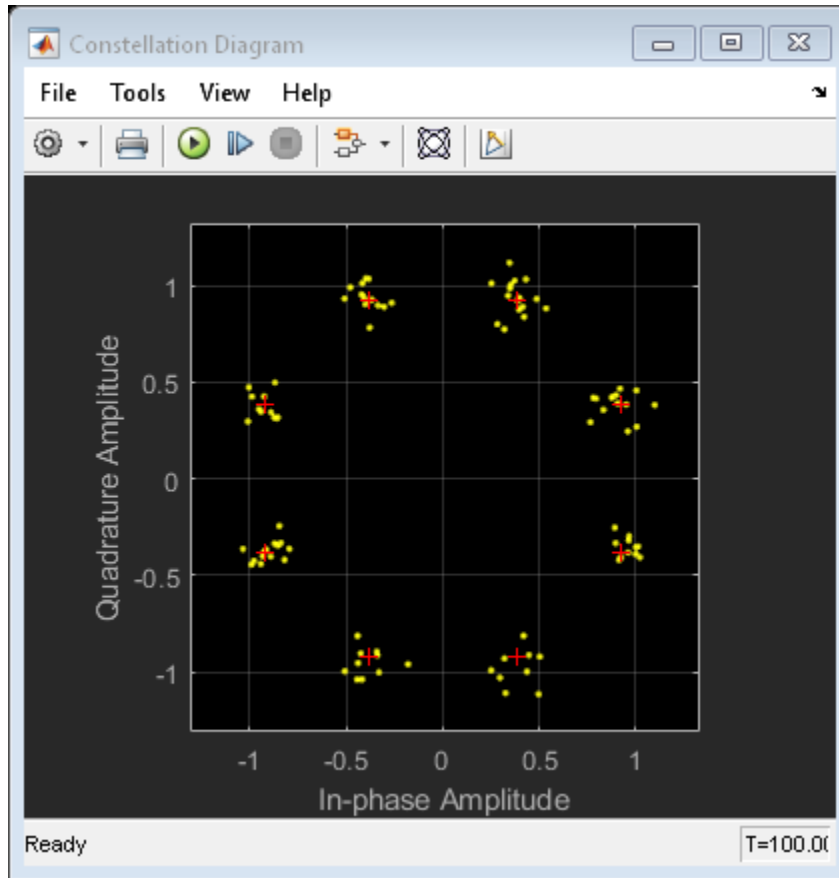
```
0.0010    0.0100
```

Modulate and Demodulate 8-PSK Signal

Open the 8-PSK model. The model generates an 8-PSK signal, applies white noise, displays the resulting constellation diagram, and computes the error statistics.



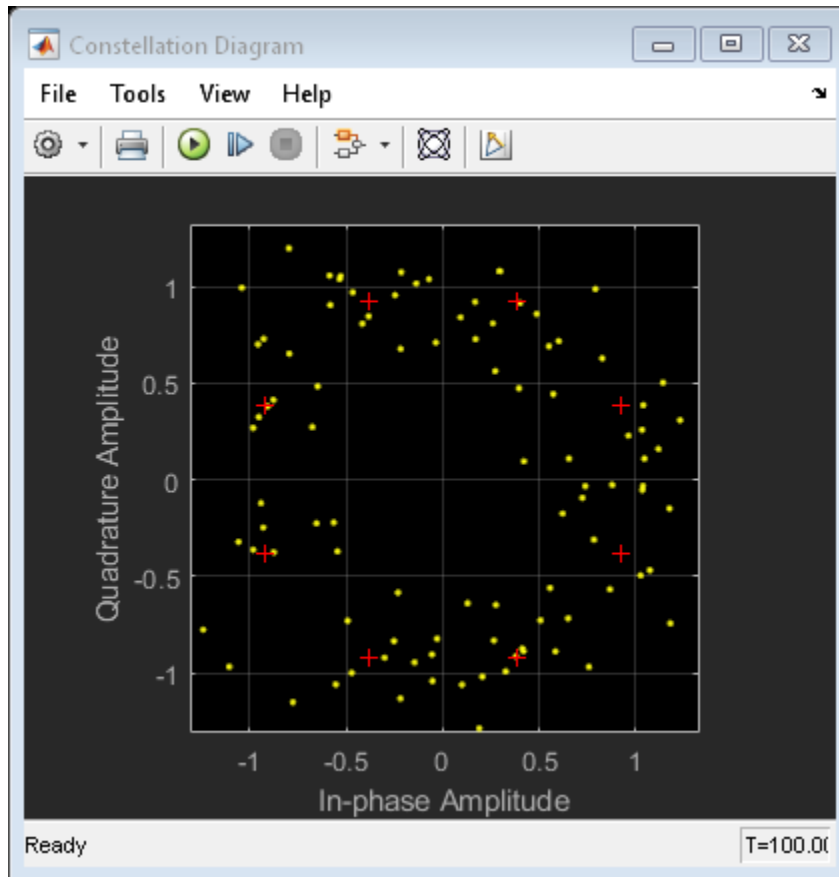
Run the model.



The error statistics are collected in vector `ErrorVec`. Because E_b/N_0 is 15 dB, there are no measured symbol errors.

Number of symbol errors = 0

Change the E_b/N_0 of the AWGN Channel block from 15 dB to 5 dB. The increase in the noise is shown in the constellation diagram.



Because of the increase in the noise level, the number of symbol errors is greater than zero.

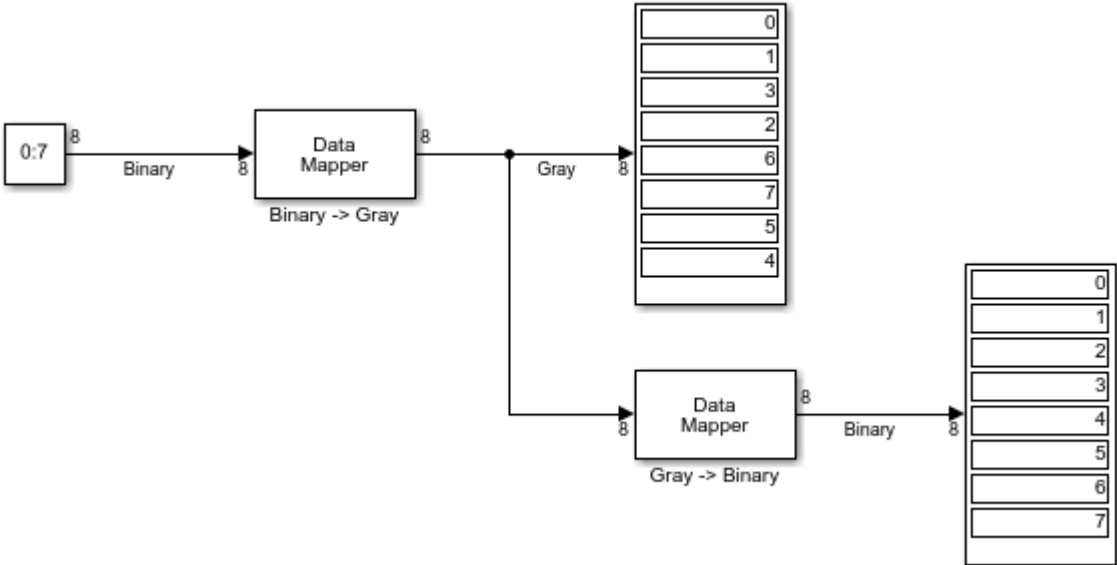
Number of symbol errors = 13

Binary to Gray Conversion in Simulink

Open the Binary-to-Gray model. The model converts a binary sequence to a Gray-coded sequence and vice versa by using Data Mapper blocks.

Run the model.

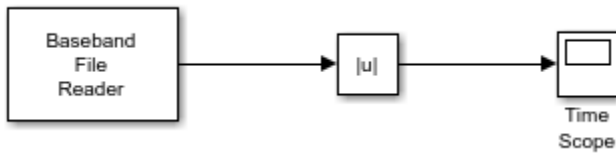
The Display blocks show the natural binary and Gray-coded sequence ordering.



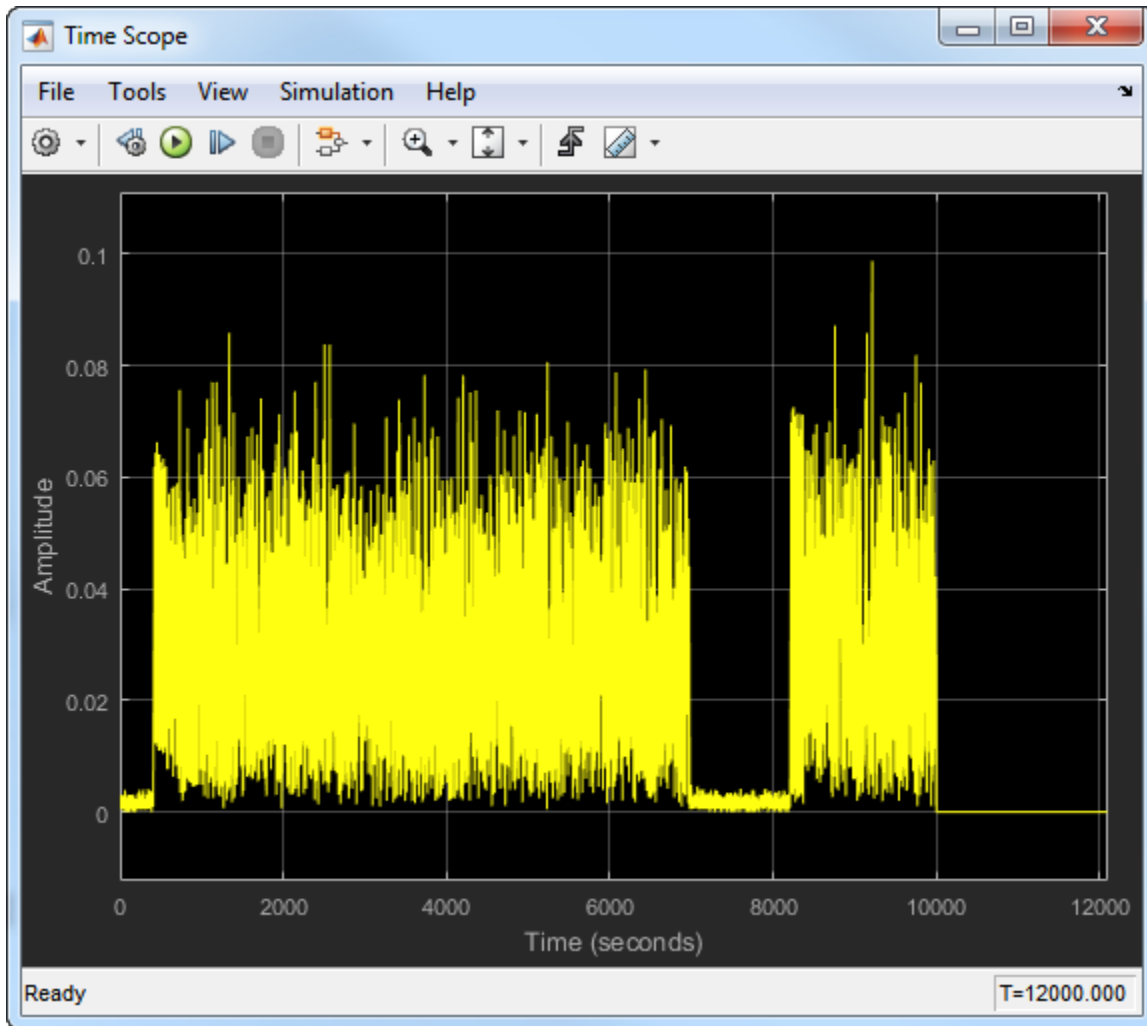
Read Baseband Signal from File

Reads a baseband signal from a saved file.

Load the model by typing `doc_baseband_reader_example` at the command line.

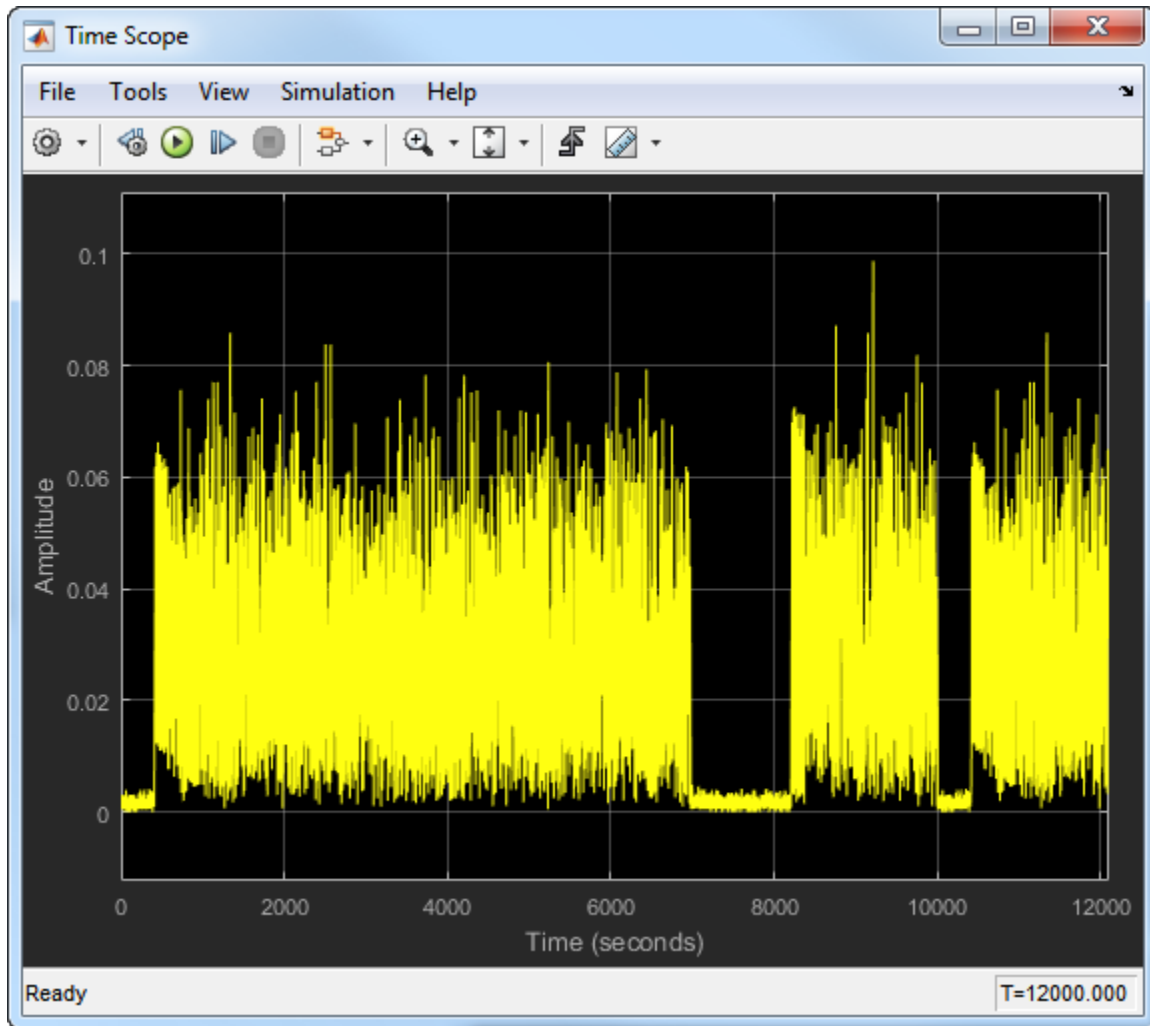


Run the model.



The file, `example.bb`, contains 10,000 samples having a sample rate of 1 Hz. The model runs for 12,000 s, which exceeds the amount of saved data. As a result, the last 2000 samples are zero padded.

Open the Baseband File Reader block, and click on the **Repeatedly read the file** check box. Rerun the model.

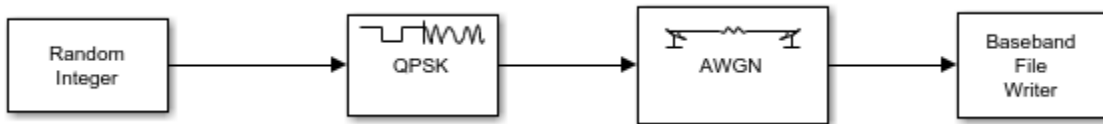


After 10,000 s, the file is read again so the samples from 10,000 to 12,000 s are a repeat of the first 2,000 samples.

Write Baseband Signal to File

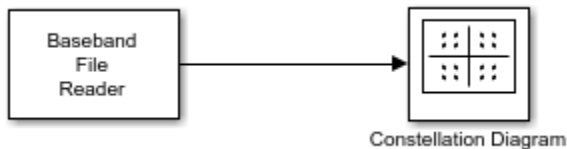
Generate and write a QPSK-modulated signal to a file. Then, read the file and plot the constellation diagram of the signal.

Type `doc_baseband_writer_ex1` at the command prompt to open the first model.

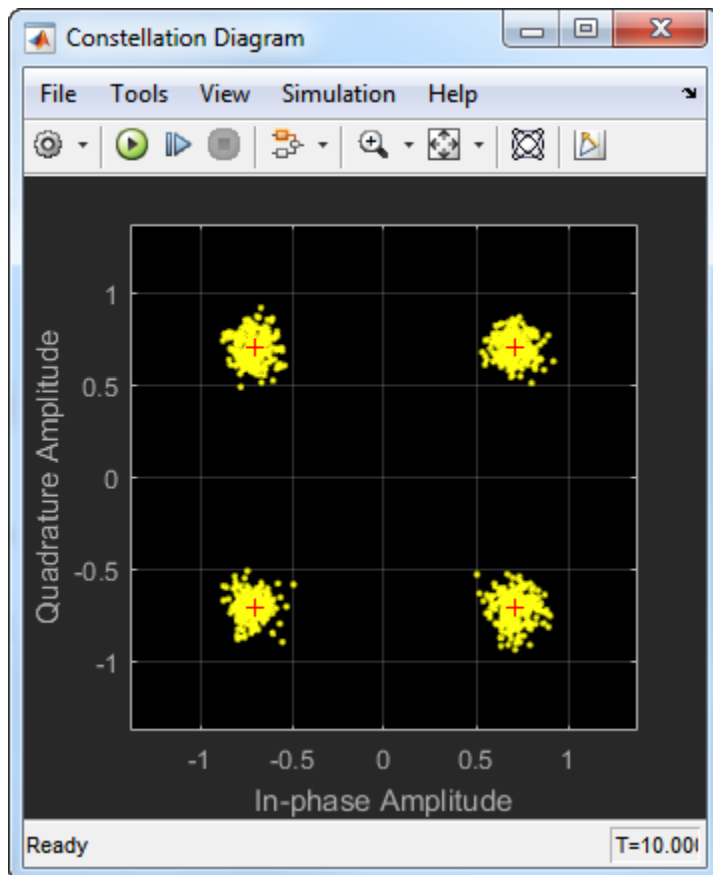


Run the model. The noisy QPSK data is saved to file `bbw_example.bb`.

Close the model. Type `doc_baseband_writer_ex2` at the command prompt to open the second model.



Run the model. The saved data is displayed by the constellation diagram.



Bluetooth Tutorials

- “What is Bluetooth?” on page 5-2
- “Bluetooth Protocol Stack” on page 5-9
- “Bluetooth Parameterization” on page 5-15
- “Create Configuration Objects” on page 5-17
- “Bluetooth Packet Structure” on page 5-24

What is Bluetooth?

Bluetooth wireless technology is the air interface intended to replace the cables connecting portable and fixed electronic equipment. Bluetooth device manufacturers have the flexibility to include optional core specification features to optimize and differentiate product offers.

Bluetooth is equated with the implementation specified by the Bluetooth Core Specification group of standards maintained by the Bluetooth Special Interest Group (SIG) industry consortium. Communications Toolbox Library for the Bluetooth Protocol functionality enables you to model Bluetooth Low Energy (BLE) communications system links, as specified in the Core System Package [Low Energy Controller volume], Specification Volume 6. It also enables you to explore variations on implementations for future evolution of the standard.

Network Architecture

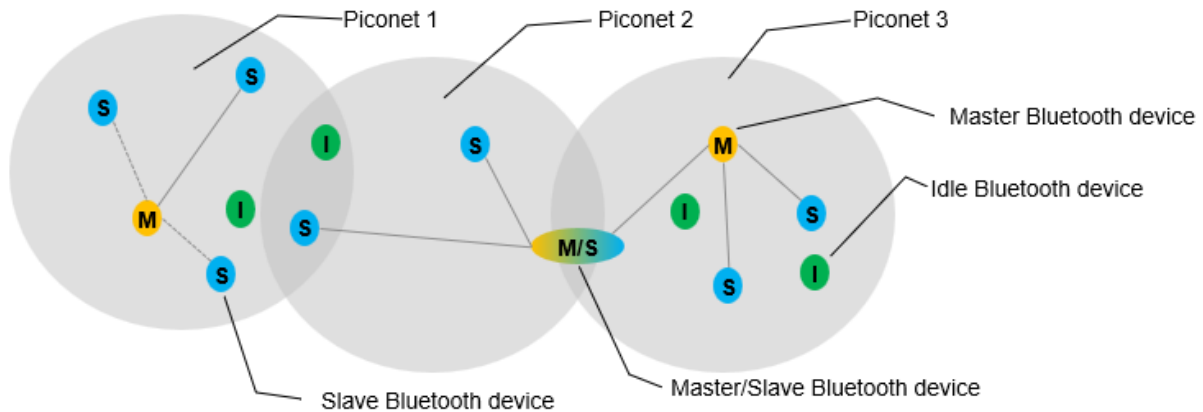
- “Point-to-Point and Point-to-Multipoint Connection Topology” on page 5-2
- “Broadcast Connection Topology” on page 5-3
- “Mesh Connection Topology” on page 5-3

Network topologies supported in Bluetooth include point-to-point, broadcast, and mesh connectivity. Point-to-point connectivity is available for devices that operate in classic Bluetooth or BLE. The broadcast and mesh connection topologies are only supported for BLE devices.

Point-to-Point and Point-to-Multipoint Connection Topology

Devices using point-to-point communication operate in a piconet. Point-to-point piconets define one-to-one device communication links. Examples of point-to-point links are links between PCs or mobile phones and peripherals such as headsets, printers, and fitness trackers. Multiple piconets connect to one another in a *scatternet* topology. Point-to-multipoint piconets define one to more than one device communication links.

This image shows a scatternet of three piconets. Each piconet shows one device in the role of *master* (M), with other devices in the *slave* (S) or *idle* (I) roles. The image also shows one device (M/S) assigned the master role in one piconet and slave in another piconet.



- A scatternet is an ad hoc network consisting of two or more piconets.
- A *piconet* is defined as a connection between two or more Bluetooth devices. Piconets are limited to a maximum of eight devices, with one master taking the master role at any given time and seven slaves.
- The individual Bluetooth devices assume the role of master, slave, or idle peer devices in a given piconet. An individual Bluetooth device can take the role of a slave in one piconet while taking the role of master in another piconet.
 - The master device provides the synchronization reference.
 - The slaves are other devices that synchronize to the clock and frequency hopping pattern of the master.
 - Other idle devices may be located in a piconet but are not active.

Broadcast Connection Topology

Broadcast piconets establish one-to-many communication links for BLE devices. Examples of broadcast links are retail point-of-interest information, indoor navigation, and asset tracking.

Mesh Connection Topology

Mesh networks establish the option of many-to-many communication links for BLE devices. Mesh topology enables the creation of large-scale device networks. Mesh is ideally suited for control, monitoring, and automation systems that require reliable and secure communication between thousands of devices.

The Bluetooth SIG specifies mesh networking requirements to enable an interoperable many-to-many (m:m) mesh networking solution for Bluetooth Low Energy (LE) wireless technology. Mesh networks are ideally suited for large-scale device networks supporting building automation, sensor networks, asset tracking, and other solutions requiring reliable and secure communication between multiple devices. Bluetooth SIG adopted these specifications.

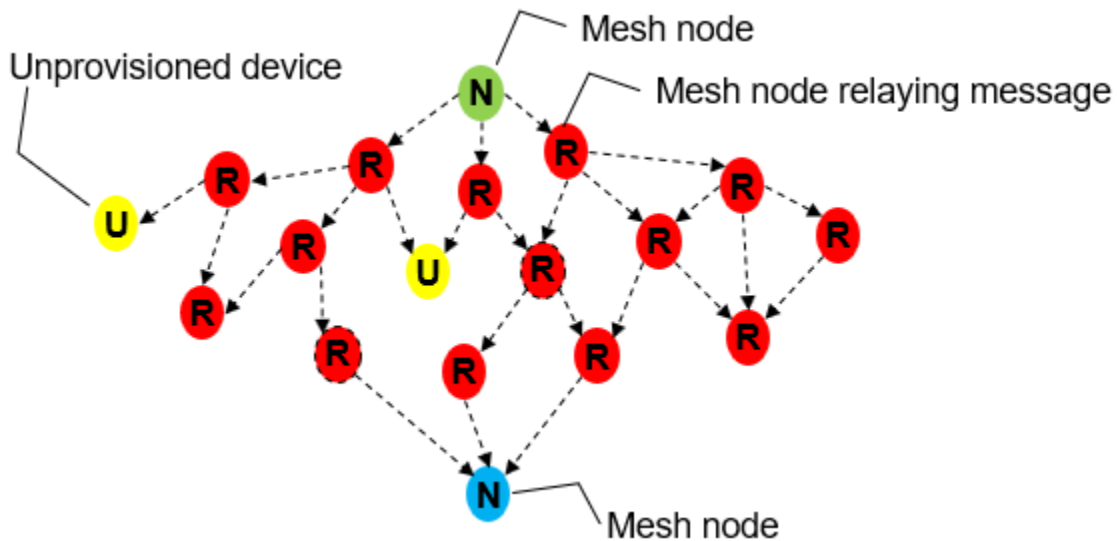
- Mesh Profile Specification - Defines fundamental requirements to enable an interoperable mesh networking solution for Bluetooth LE wireless technology.
- Mesh Model Specification - Introduces models, used to define basic functionality of nodes on a mesh network.
- Mesh Device Properties - Defines device properties required for the Mesh Model specification.

The Bluetooth Mesh Model Specification defines categories of BLE mesh models. Model categories include

- Foundation models
- Generic models
- Sensors
- Time and scenes
- Lighting

All devices must implement the foundation models for configuration server and health server. All other models in the model categories are optional and implemented based on the service the BLE device performs.

As described in the Mesh Profile Bluetooth specification, mesh networks operate as managed-flood-based networks. Devices use broadcast channels to transmit messages to other devices, and the messages are relayed forward to other devices extending the range of the original message.



A device that is not a member of a mesh network is referred to as an *unprovisioned device*. A device that is a member of a mesh network is known as a *node*. Devices are added to a mesh network by a *Provisioner*. Nodes in a mesh network share network keys that enable them to receive and to relay messages from other nodes in their network or subnet. Network keys are used to secure and authenticate messages at the network layer. Unprovisioned devices cannot receive messages because they do not have the network key to recover the message.

Bluetooth Products

The Bluetooth SIG defines a Bluetooth product as any product containing an implementation of Bluetooth wireless technology. Bluetooth products are classified as:

- Bluetooth End Product
- Bluetooth Host Subsystem Product
- Bluetooth Controller Subsystem Product
- Bluetooth Profile Subsystem Product
- Bluetooth Component Product
- Bluetooth Development Tool

- Bluetooth Test Equipment

The Communications Toolbox Library for the Bluetooth Protocol provides features enabling you to model Bluetooth Host and Controller Subsystem Products fully compliant Bluetooth links with the low energy (LE) core configuration.

Bluetooth Low Energy Core Configuration

The Bluetooth Core Specification, Volume 0, Part B, Section 4.4 specifies a set of required features that must be implemented to model fully compliant Bluetooth links with the low energy (LE) core configuration.

The LE core configuration defines three main layers - Application, Host, and Controller. The Communications Toolbox Library for the Bluetooth Protocol provides features to model the host and controller layers. Requirements defined in the Bluetooth Core Specification for the host and controller include

Layer	Sublayer	Bluetooth Specification Volume	Required Features
Host	Logical link control and adaptation protocol (L2CAP)	Volume 3, part A	If the GAP Peripheral or Central role is supported, L2CAP LE Signaling Channel (CID 0x0005) and all mandatory features associated with it.
	Generic access profile (GAP)	Volume 3, part C	All mandatory features for at least one of the LE GAP roles (Broadcaster, Observer, Peripheral, or Central) in sections 9–12 and section 15.

Layer	Sublayer	Bluetooth Specification Volume	Required Features
	Attribute profile (ATT)	Volume 3, part F	If the GAP Peripheral or Central role is supported, all mandatory features.
	Generic attribute profile (GATT)	Volume 3, part G	GATT is mandatory when ATT is supported. When supported, all mandatory features.
	Security manager (SM)	Volume 3, part H	If the GAP Peripheral or Central role is supported, all mandatory features.
Controller	Physical (PHY)	Volume 6, part A	All mandatory features.
	Link layer (LL)	Volume 6, part B	All mandatory features.

For a description of the mapping between Bluetooth protocol stack functionality and the OSI reference model, see "Bluetooth Protocol Stack" on page 5-9.

References

[1] <https://www.bluetooth.com/>

[2] "Bluetooth Core Specification." *Bluetooth Special Interest Group (SIG)*.

[3] "Supplement to the Bluetooth Core Specification, CSS Version 7." *Bluetooth Special Interest Group (SIG)*.

[4] "Bluetooth Core Specification Addendum 6." *Bluetooth Special Interest Group (SIG)*.

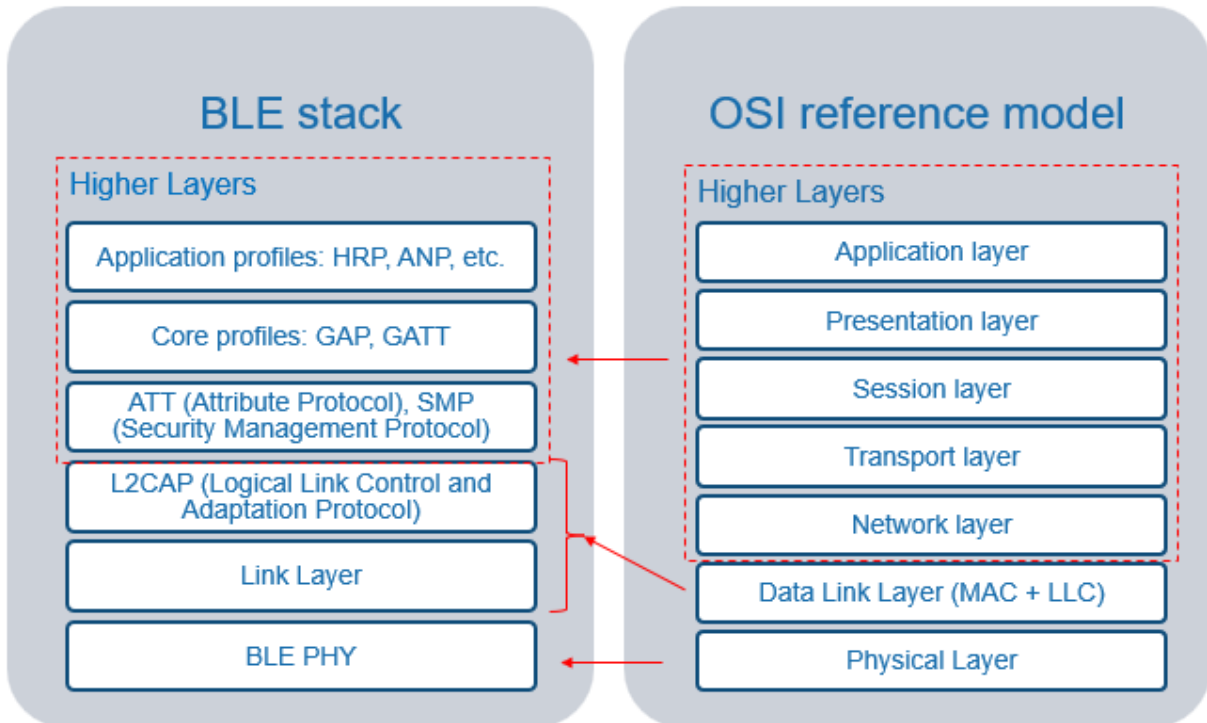
[5] "Mesh Profile Bluetooth Specification." *Bluetooth Special Interest Group (SIG)*.

[6] "Mesh Model Bluetooth Specification." *Bluetooth Special Interest Group (SIG)*.

[7] "Mesh Device Properties Bluetooth Specification." *Bluetooth Special Interest Group (SIG)*.

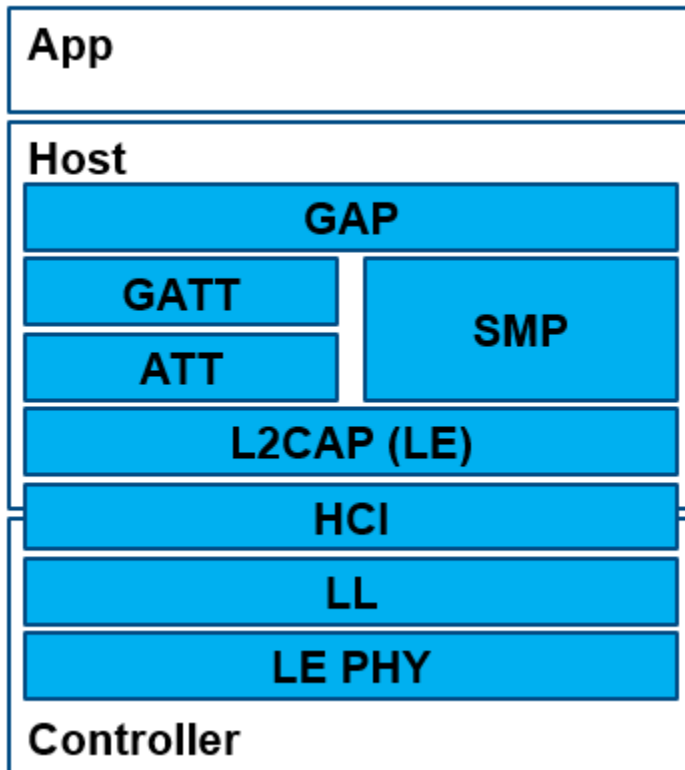
Bluetooth Protocol Stack

This figure compares the BLE protocol stack to the OSI reference model.



- There is one-to-one mapping at the physical layer (PHY)
- The OSI data link layer (DLL) maps to the BLE logical link control and adaptation protocol (L2CAP) and link layer (LL).
- In the BLE stack, the higher layers provide application layer services, device roles and modes, connection management, and security protocol.

Focusing on the BLE protocol stack, the functionality is divided between three main layers — Application, Host, and Controller.



Application (App) Layer

The Application (App) is the direct user interface and Bluetooth SIG defines profiles that afford interoperability between various applications. The Bluetooth specification also allows vendors to define proprietary profiles for use cases not defined by SIG profiles.

Host Layer

As shown in the image, the host layer includes the GAP, GATT, ATT, SMP, L2CAP, and host-side HCI sublayers

Generic Access Profile

The generic access profile (GAP) specifies roles, modes, and procedures of a device. The GAP also manages the connection establishment and the security. It interfaces directly with the Application layer.

Generic Attribute Profile

Bluetooth LE specifies how data is stored and exchanged between LE devices. The generic attribute profile (GATT) provides a reference framework for all GATT-based profiles. The GATT encapsulates the ATT and is responsible for coordinating the exchange of profiles in a BLE link. Profiles include information and data such as handle assignment, a Universally Unique Identifier (UUID), and a set of permissions.

For devices that implement the GATT profile,

- The client is the device that initiates commands and requests toward the server, and can receive responses, indications, and notifications.
- The server is the device that accepts incoming commands and requests from the client, and sends responses, indications, and notifications to the client.

GATT uses client-server architecture. The roles are not fixed and are determined when a device initiates a defined procedure. Roles are released when the procedure ends.

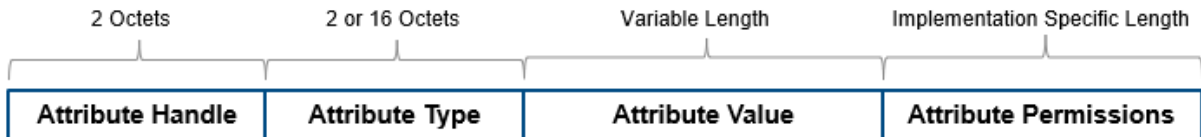
GATT terminology used in GATT includes

- Service - A collection of data and associated behaviors used to accomplish a particular function or feature.
- Characteristic - A value used in a service along with its permissions.
- Characteristic descriptor - Description of the associated characteristic behaviour.
- GATT-Client - Initiates commands and requests towards the server and can receive responses, indications, and notifications sent by the server.
- GATT-Server - Accepts incoming commands and requests from a client and sends responses, indications, and notifications to the client.

Attribute Protocol

Bluetooth LE specifies the attribute protocol (ATT) to transfer attribute data between clients and servers in GATT-based profiles. The ATT defines the roles of the client-server architecture. The roles typically correspond to the master and the slave as defined in the

link layer. In general, a device could be a client, a server, or both, irrespective of whether it is a master or a slave. The ATT also performs data organization into attributes.



Device attributes are represented as

- Attribute handle is a 16-bit identifier value assigned by the server to allow a client to reference those attributes.
- Attribute type is a universally unique identifier (UUID) defined by Bluetooth SIG. For example, UUID 0x2A37 represents a heart rate measurement.
- Attribute value is a variable length field. The UUID associated with it and the service class of the service record containing it determine the length of the attribute value field.
- Attribute permissions are sets of permission values associated with each attribute. These permissions specify read and write privileges for an attribute, and the security level required for read and write permission.

Security Manager Protocol

The security manager protocol (SMP) applies security algorithms to encrypt and decrypt data packets. It defines the initiator and the responder, corresponding to the master and the slave, once the connection is established.

Logical Link Control and Adaptation Protocol

The logical link control and adaptation protocol (L2CAP) encapsulates data from the BLE higher layers into the standard BLE packet format for transmission (or extracts data from the standard BLE LL packet on reception) according to link configuration specified at the ATT and SMP.

Host-Side Host Controller Interface (HCI)

The host controller interface (HCI) on the host side handles the interface between the host and the controller. The HCI defines a set of commands and events for transmission

and reception of packet data. When transmitting data, the HCI translates raw data into packets to send them from the host layer to the controller.

Controller Layer

As shown in the image, the controller layer includes the controller-side HCI, LL, and PHY.

Controller-Side Host Controller Interface (HCI)

The host controller interface (HCI) on the controller side handles the interface between the host and the controller. The HCI defines a set of commands and events for transmission and reception of packet data. When receiving packets from the controller, the HCI extracts raw data at the controller to send to the host.

Link Layer

The link layer (LL) performs task similar to the MAC layer of the OSI model. In Bluetooth, the LL interfaces directly with the PHY and manages the link state of the radio to define the role a device operates in, such as master, slave, advertiser, and scanner.

Physical Layer

The physical layer (PHY) air interface operates in the same unlicensed 2.4 GHz Industrial, Scientific, and Medical (ISM) frequency band as Wi-Fi®. Other characteristics of the Bluetooth PHY air interface include

- Operating RF frequency band from 2.4000 GHz to 2.4835 GHz.
- Channel bandwidth is 2 MHz. The operating band is divided into 40 channels, $k = 0, \dots, 39$. Center frequencies of the k channels are $2402 + k \times 2$ MHz.
 - User data packets are transmitted using channels in the range [0,36].
 - Advertising data packets are transmitted in channels 37, 38, and 39.
- GFSK modulation scheme.
- Uses frequency-hopping spread spectrum to reduce interference and to counter the impact of fading channels. The time between frequency hops can vary from 7.5 ms to 4 seconds and is set at connection time for each slave.
- Support for throughput at 1 Mbps is mandatory (for specification version 4.x compliant devices). At a data rate of 1 Mbps, the transmission is uncoded.
- Optionally, specification version 5.0 compliant devices support these additional data rates.

- Coded transmission at bit rates of 500 kbps or 125 kbps.
- Uncoded transmission at a bit rate of 2 Mbps.

Bluetooth Parameterization

Communications Toolbox Library for the Bluetooth Protocol configuration objects initialize, store, and validate configuration properties.

Configuration Objects in Communications Toolbox Library for the Bluetooth Protocol

The configuration objects are designed specifically as containers to store properties. They also provide some level of data validation for the function inputs that they maintain. Functions perform further data validation across input settings based on run-time conditions.

The configuration objects are optimized for iterative computations that process large streams of data, such as communications systems.

Communications Toolbox Library for the Bluetooth Protocol configuration objects define and configure format-specific and function-specific properties. The property page of each object contains descriptions, valid settings, ranges, and other information about the object properties.

- `bleLLDataChannelPDUConfig` — The Link Layer Data Channel Protocol Data Unit configuration object defines and configures LL data channel PDUs.
- `bleLLAdvertisingChannelPDUConfig` — The Link Layer Advertising Channel Protocol Data Unit configuration object defines and configures LL advertising channel PDUs.
- `bleL2CAPFrameConfig` — The Layer 2 CAP Frame configuration object defines and configures Layer 2 CAP frame.
- `bleGAPDataBlockConfig` — The GAP Data Block configuration object defines and configures GAP data block.
- `bleATTPDUConfig` — The ATT Protocol Data Unit configuration object defines and configures ATT PDUs.

See Also

More About

- “Create Configuration Objects” on page 5-17

Create Configuration Objects

Communications Toolbox Library for the Bluetooth Protocol uses value objects to organize properties required for generation of higher layer Bluetooth PDUs. After you create the various configuration objects described here, you can use them to generate waveforms.

Create Link Layer Data Channel PDU Configuration Object

This example shows how to create a BLE link layer data channel PDU configuration object. It also shows how to change the default property settings by using dot notation or by overriding the default settings by using `Name, Value` pairs when creating the object.

Create Object and Then Modify Properties

Create a BLE link layer data channel PDU configuration object with default settings.

```
lldatapdu = bleLLDataChannelPDUConfig

lldatapdu =
    bleLLDataChannelPDUConfig with properties:
        LLID: 'Data (continuation fragment/empty)'
        NESN: 0
        SequenceNumber: 0
        MoreData: 0
```

Modify the defaults to specify a 'Data (start fragment/complete)' PDU.

```
lldatapdu.LLID = 'Data (start fragment/complete)'

lldatapdu =
    bleLLDataChannelPDUConfig with properties:
        LLID: 'Data (start fragment/complete)'
        NESN: 0
        SequenceNumber: 0
        MoreData: 0
```

Override Default Property Values During Object Creation

Create a BLE link layer data channel PDU configuration object, using Name, Value pairs to specify a 'Control' PDU.

```
lldatapdu2 = bleLLDataChannelPDUConfig('LLID', 'Control')

lldatapdu2 =
  bleLLDataChannelPDUConfig with properties:

          LLID: 'Control'
          NESN: 0
  SequenceNumber: 0
          MoreData: 0
  ControlConfig: [1x1 bleLLControlPDUConfig]
```

Create Link Layer Advertising Channel PDU Configuration Object

This example shows how to create a BLE link layer advertising channel PDU configuration object. It also shows how to change the default property settings by using dot notation or by overriding the default settings by using Name, Value pairs when creating the object.

Create Object and Then Modify Properties

Create a BLE link layer advertising channel PDU configuration object with default settings.

```
lladvert pdu = bleLLAdvertisingChannelPDUConfig

lladvert pdu =
  bleLLAdvertisingChannelPDUConfig with properties:

          PDUType: 'Advertising indication'
          ChannelSelection: 'Algorithm1'
  AdvertiserAddressType: 'Random'
          AdvertiserAddress: '0123456789AB'
          AdvertisingData: [3x2 char]
```

Modify the defaults to specify a 'Advertising direct indication' PDU.

```
lladvert pdu.PDUType = 'Advertising direct indication'
```

```

l1advert pdu =
  bleLLAdvertisingChannelPDUConfig with properties:
      PDUType: 'Advertising direct indication'
      ChannelSelection: 'Algorithm1'
      AdvertiserAddressType: 'Random'
      AdvertiserAddress: '0123456789AB'
      TargetAddressType: 'Random'
      TargetAddress: '0123456789CD'

```

Override Default Property Values During Object Creation

Create a BLE link layer advertising channel PDU configuration object, using Name, Value pairs to specify a 'Scan response' PDU using channel section algorithm 2.

```

l1datapdu2 = bleLLAdvertisingChannelPDUConfig('PDUType', 'Scan response', 'ChannelSelectio
l1datapdu2 =
  bleLLAdvertisingChannelPDUConfig with properties:
      PDUType: 'Scan response'
      ChannelSelection: 'Algorithm2'
      AdvertiserAddressType: 'Random'
      AdvertiserAddress: '0123456789AB'
      ScanResponseData: [3x2 char]

```

Create L2CAP Frame Configuration Object

This example shows how to create a BLE logical link control and adaptation protocol (L2CAP) frame configuration object. It also shows how to change the default property settings by using dot notation or by overriding the default settings by using Name, Value pairs when creating the object.

Create Object and Then Modify Properties

Create a BLE L2CAP frame configuration object with default settings.

```

l2capframe = bleL2CAPFrameConfig
l2capframe =
  bleL2CAPFrameConfig with properties:

```

```
ChannelIdentifier: '0005'  
  CommandType: 'Credit based connection request'  
  SignalIdentifier: '01'  
SourceChannelIdentifier: '0040'  
  LEPSM: '001F'  
MaxTransmissionUnit: 23  
MaxPDUPayloadSize: 23  
Credits: 1
```

Modify the defaults setting the channel identifier to '0004' to specify an ATT channel.

```
l2capframe.ChannelIdentifier = '0004'  
  
l2capframe =  
  bleL2CAPFrameConfig with properties:  
  
    ChannelIdentifier: '0004'
```

Override Default Property Values During Object Creation

Create a BLE L2CAP frame configuration object, using `Name, Value` pairs to specify a 'Command reject' signaling channel command with the reject reason 'Invalid CID in request'.

```
l2capframe = bleL2CAPFrameConfig ('CommandType', 'Command reject', 'CommandRejectReason'  
  
l2capframe =  
  bleL2CAPFrameConfig with properties:  
  
    ChannelIdentifier: '0005'  
    CommandType: 'Command reject'  
    SignalIdentifier: '01'  
    CommandRejectReason: 'Invalid CID in request'  
    SourceChannelIdentifier: '0040'  
    DestinationChannelIdentifier: '0040'
```


Create GAP Data Block Configuration Object

This example shows how to create a BLE GAP data block configuration object. It also shows how to change the default property settings by using dot notation or by overriding the default settings by using Name, Value pairs when creating the object.

Create Object and Then Modify Properties

Create a BLE GAP data block configuration object with default settings.

```
gapDataBlk = bleGAPDataBlockConfig

gapDataBlk =
  bleGAPDataBlockConfig with properties:

    AdvertisingDataTypes: {'Flags'}
    LEDiscoverability: 'General'
    BREDR: 0
```

Modify the defaults to specify for an advertising data block for 'Flags' and 'Tx power level' advertising data types.

```
gapDataBlk.AdvertisingDataTypes = {'Flags'; 'Tx power level'}

gapDataBlk =
  bleGAPDataBlockConfig with properties:

    AdvertisingDataTypes: {2x1 cell}
    LEDiscoverability: 'General'
    BREDR: 0
    TxPowerLevel: 0
```

```
gapDataBlk.AdvertisingDataTypes
```

```
ans = 2x1 cell array
    {'Flags' }
    {'Tx power level'}
```

Override Default Property Values During Object Creation

Create a BLE GAP data block configuration object, using Name, Value pairs to specify 'Advertising interval' and 'Local name' advertising data types.

```
gapDataBlk = bleGAPDataBlockConfig ('AdvertisingDataTypes', {'Advertising interval',
gapDataBlk =
  bleGAPDataBlockConfig with properties:

    AdvertisingDataTypes: {2x1 cell}
                        LocalName: 'Bluetooth'
    LocalNameShortening: 0
    AdvertisingInterval: 32

gapDataBlk.AdvertisingDataTypes

ans = 2x1 cell array
    {'Advertising interval'}
    {'Local name'          }
```

Create Attribute PDU Configuration Object

This example shows how to create a BLE attribute (ATT) PDU configuration object. It also shows how to change the default property settings by using dot notation or by overriding the default settings by using Name, Value pairs when creating the object.

Create Object and Then Modify Properties

Create a BLE ATT PDU configuration object with default settings.

```
attpdu = bleATTPDUConfig

attpdu =
  bleATTPDUConfig with properties:

    Opcode: 'Read request'
    AttributeHandle: '0001'
```

Modify the defaults to specify a 'Read blob request' operation code.

```
attpdu.Opcode = 'Read blob request'

attpdu =
  bleATTPDUConfig with properties:
```

```
        Opcode: 'Read blob request'  
AttributeHandle: '0001'  
        Offset: 0
```

Override Default Property Values During Object Creation

Create a BLE ATT PDU configuration object, using Name, Value pairs to specify 'Information request' for the operation code of a request PDU that caused an error.

```
l1datapdu2 = bleATTPDUConfig('RequestedOpcode', 'Information request')
```

```
l1datapdu2 =  
    bleATTPDUConfig with properties:  
  
        Opcode: 'Read request'  
AttributeHandle: '0001'
```

See Also

More About

- “Bluetooth Parameterization” on page 5-15

Bluetooth Packet Structure

Equalization

- “Equalization” on page 6-2
- “Adaptive Equalizers” on page 6-6
- “MLSE Equalizers” on page 6-46

Equalization

In this section...
“Equalizer Structure Options” on page 6-2
“Selected References for Equalizers” on page 6-4

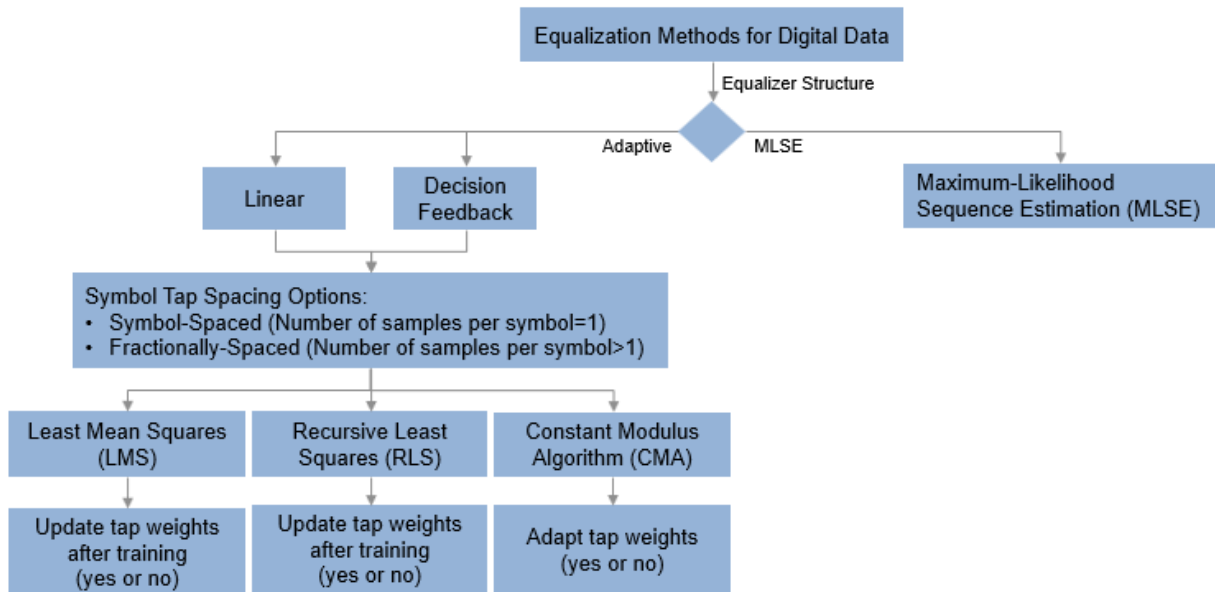
In a multipath fading scattering environment, the receiver typically detects several constantly changing, delayed versions of the transmitted signal. These time-dispersive channels cause intersymbol interference (ISI) that occurs when symbols received from multiple paths are delayed and overlap in time. ISI causes high error rates because the symbols from multiple received paths interfere with each other and become indistinguishable by the receiver.

Equalizers attempt to mitigate ISI and improve the receiver performance. Equalizer structures are filters that attempt to match the propagation channel response. For time-varying propagation channels, adapting the equalization filter tap weights so that they maintain a match to the channel over time improves the error rate performance.

Equalizer Structure Options

The Communications Toolbox includes System objects and blocks to recover transmitted data using by linear, decision-feedback, or maximum-likelihood sequence estimation (MLSE) equalization structures. For more information, see “Selected References for Equalizers” on page 6-4.

This figure shows the high-level configuration options for each equalization structure.



For each equalizer structure, you can configure structural settings (such as the number of taps and initial set of tap weights), algorithmic settings (such as the step size), and the signal constellation used by the modulator in your design. You also specify adaptability of the equalizer tap weights throughout the simulation.

- Linear and decision-feedback filter equalizer structures adapt tap weights by using the LMS, RLS, or CMA adaptive algorithm. When using these equalizer structures, the number of samples per symbol determines whether symbols are processed using whole or fractional symbol spacing.
 - When using LMS and RLS adaptive algorithms, the equalizer begins operating in tap weights training mode. Configure the equalizer to operate adaptively in decision-directed mode or without further adjustment of taps after training is completed.
 - When using the CMA adaptive algorithm, the equalizer has no training mode. You can configure the equalizer to operate adaptively in decision-directed mode or in nonadaptive mode.

To explore the linear and decision-feedback filter equalizer capabilities, see “Adaptive Equalizers” on page 6-6.

- Maximum-Likelihood Sequence Estimation (MLSE) equalizers use the Viterbi algorithm. The MLSE equalization structure provides the optimal match to the received symbols but it requires an accurate channel estimate and is the most computationally complex structure. To explore MLSE equalizer capabilities, see “MLSE Equalizers” on page 6-46.

The computational complexity of each equalization structure grows with the length of the channel time dispersion. Considering the Doppler and frequency selectivity characteristics of the channel, use the information in this table when selecting which equalization structure to use in your simulation.

Equalizer Structure	Doppler Speed	Is Channel Frequency Selective?	Computational Complexity
Linear RLS	High	No	Medium
Linear LMS	Low	No	Lowest
Linear CMA	Low	No	Lowest
DFE RLS	High	Yes	Medium
DFE LMS	Low	Yes	Lowest
DFE CMA	Low	Yes	Lowest
MLSE	Low	Yes	Highest

Selected References for Equalizers

- [1] Farhang-Boroujeny, B., *Adaptive Filters: Theory and Applications*, Chichester, England, John Wiley & Sons, 1998.
- [2] Haykin, Simon, *Adaptive Filter Theory*, Third Ed., Upper Saddle River, NJ, Prentice-Hall, 1996.
- [3] Kurzweil, Jack, *An Introduction to Digital Communications*, New York, John Wiley & Sons, 2000.
- [4] Proakis, John G., *Digital Communications*, Fourth Ed., New York, McGraw-Hill, 2001.
- [5] Steele, Raymond, Ed., *Mobile Radio Communications*, Chichester, England, John Wiley & Sons, 1996.

See Also

Objects

`comm.DecisionFeedbackEqualizer` | `comm.LinearEqualizer` |
`comm.MLSEEqualizer`

Blocks

Decision Feedback Equalizer | Linear Equalizer | MLSE Equalizer

More About

- “Adaptive Equalizers” on page 6-6
- “MLSE Equalizers” on page 6-46

Adaptive Equalizers

In this section...
“Symbol Tap Spacing” on page 6-7
“Linear Equalizers” on page 6-7
“Decision-Feedback Equalizers” on page 6-8
“Reference Signal and Operating Modes” on page 6-10
“Error Calculation” on page 6-10
“Updating Tap Weights” on page 6-10
“Configuring Adaptive Equalizers” on page 6-12
“Using Adaptive Equalizers in Simulink” on page 6-37
“Adaptive Equalization with Filtering and Fading Channel” on page 6-37

Adaptive equalizer structures provide suboptimal equalization of time variations in the propagation channel characteristics. However, these equalizers are appealing because their computational complexity is lower than “MLSE Equalizers” on page 6-46.

In Communications Toolbox, the `comm.LinearEqualizer` and `comm.DecisionFeedbackEqualizer` System objects and the Linear Equalizer and Decision Feedback Equalizer blocks use tap delay line filters to equalize a linearly modulated signal through a dispersive channel. These features output the estimate of the signal by using an estimate of the channel modeled as a finite input response (FIR) filter.

To decode a received signal, the adaptive equalizer:

- 1 Applies the FIR filter to the symbols in the input signal. The FIR filter tap weights correspond to the channel estimate.
- 2 Outputs the signal estimate and uses the signal estimate to update the tap weights for the next symbol. The signal estimate and updating of weights depends on the adaptive equalizer structure and algorithm.

Adaptive equalizer structure options are linear or decision-feedback. Adaptive algorithm options are least mean square (LMS), recursive mean square (RMS), or constant modulus algorithm (CMA). For background material on adaptive equalizers, see “Selected References for Equalizers” on page 6-4.

Symbol Tap Spacing

You can configure the equalizer to operate as a symbol-spaced equalizer or as a fractional symbol-spaced equalizer.

- To operate the equalizer at a symbol-spaced rate, specify the number of samples per symbol as 1. Symbol-rate equalizers have taps spaced at the symbol duration. Symbol-rate equalizers are sensitive to timing phase.
- To operate the equalizer at a fractional symbol-spaced rate, specify the number of input samples per symbol as an integer greater than 1 and provide an input signal oversampled at that sampling rate. Fractional symbol-spaced equalizers have taps spaced at an integer fraction of the input symbol duration. Fractional symbol-spaced equalizers are not sensitive to timing phase.

Note The MLSE equalizer supports fractional symbol spacing but using it is not recommended. The MLSE computational complexity and burden grows exponentially with the length of the channel time dispersion. Oversampling the input means multiplying the exponential term by the number of samples per symbol.

Linear Equalizers

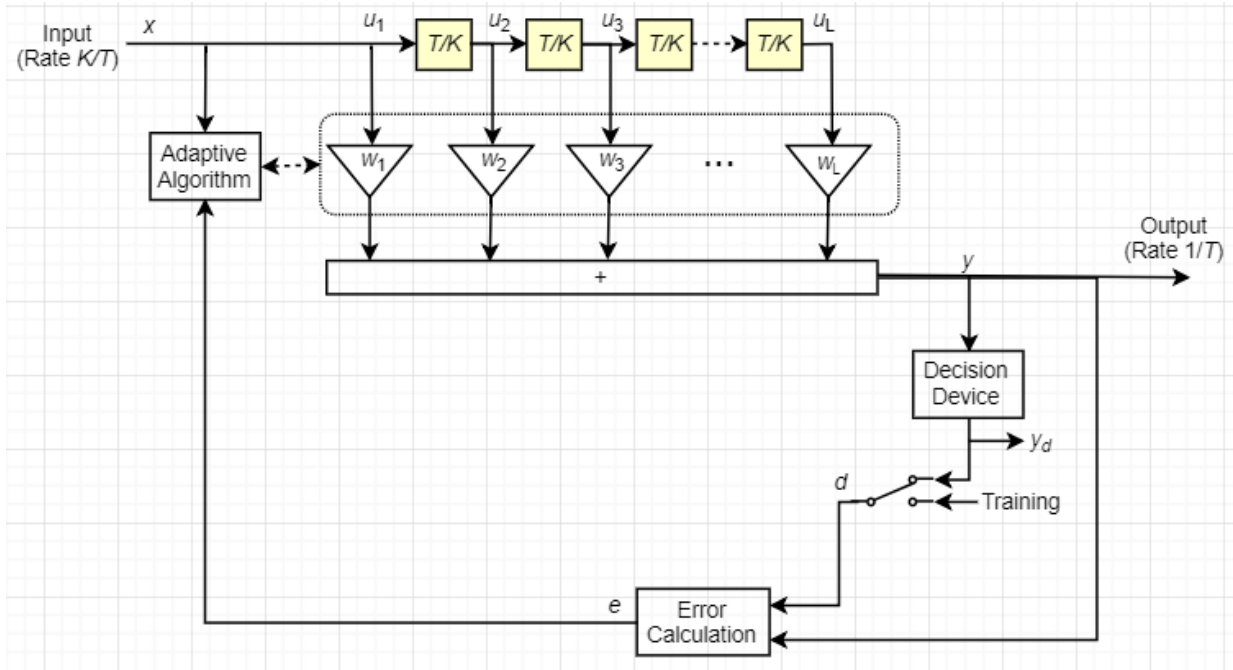
Linear equalizers can remove intersymbol interference (ISI) when the frequency response of a channel has no null. If a null exists in the frequency response of a channel, linear equalizers tend to enhance the noise. In this case, use decision feedback equalizers to avoid enhancing the noise.

A linear equalizer consists of a tapped delay line that stores samples from the input signal. Once per symbol period, the equalizer outputs a weighted sum of the values in the delay line and updates the weights to prepare for the next symbol period.

Linear equalizers can be symbol-spaced or fractional symbol-spaced.

- For a symbol-spaced equalizer, the number of samples per symbol, K , is 1. The output sample rate equals the input sample rate.
- For a fractional symbol-spaced equalizer, the number of samples per symbol, K , is an integer greater than 1. Typically, K is 4 for fractionally spaced equalizers. The output sample rate is $1/T$ and the input sample rate is K/T , where T is the symbol period. Tap-weight updating occurs at the output rate.

This schematic shows a linear equalizer with L weights, a symbol period of T , and K samples per symbol. If K is 1, the result is a symbol-spaced linear equalizer instead of a fractional symbol-spaced linear equalizer.



In each symbol period, the equalizer receives K input samples at the tapped delay line. The equalizer then outputs a weighted sum of the values in the tapped delay line and updates the weights to prepare for the next symbol period.

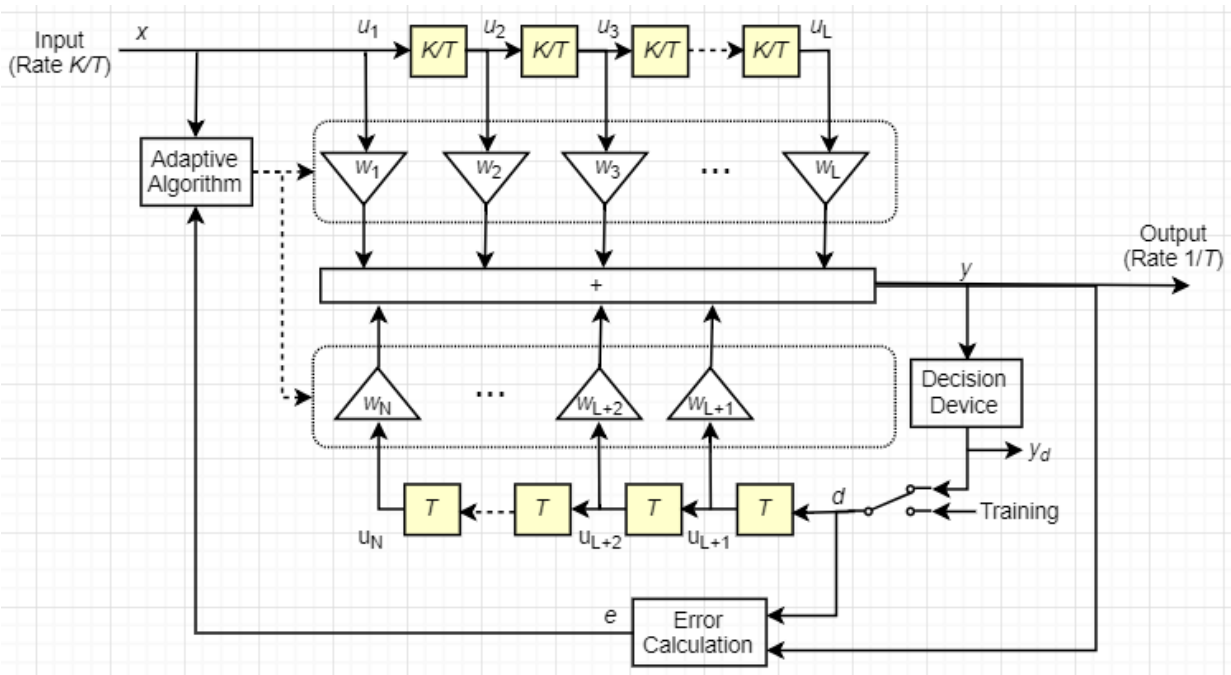
Decision-Feedback Equalizers

A decision feedback equalizer (DFE) is a nonlinear equalizer that reduces intersymbol interference (ISI) in frequency-selective channels. If a null exists in the frequency response of a channel, DFEs do not enhance the noise. A DFE consists of a tapped delay line that stores samples from the input signal and contains a forward filter and a feedback filter. The forward filter is similar to a linear equalizer. The feedback filter contains a tapped delay line whose inputs are the decisions made on the equalized signal. Once per symbol period, the equalizer outputs a weighted sum of the values in the delay line and updates the weights to prepare for the next symbol period.

DFEs can be symbol-spaced or fractional symbol-spaced.

- For a symbol-spaced equalizer, the number of samples per symbol, K , is 1. The output sample rate equals the input sample rate.
- For a fractional symbol-spaced equalizer, the number of samples per symbol, K , is an integer greater than 1. Typically, K is 4 for fractional symbol-spaced equalizers. The output sample rate is $1/T$ and the input sample rate is K/T . Tap weight updating occurs at the output rate.

This schematic shows a fractional symbol-spaced DFE with a total of N weights, a symbol period of T , and K samples per symbol. The filter has L forward weights and $N-L$ feedback weights. The forward filter is at the top, and the feedback filter is at the bottom. If K is 1, the result is a symbol-spaced DFE instead of a fractional symbol-spaced DFE.



In each symbol period, the equalizer receives K input samples at the forward filter and one decision or training sample at the feedback filter. The equalizer then outputs a weighted sum of the values in the forward and feedback delay lines and updates the weights to prepare for the next symbol period.

Note The algorithm for the Adaptive Algorithm block in the schematic jointly optimizes the forward and feedback weights. Joint optimization is especially important for convergence in the recursive least square (RLS) algorithm.

Reference Signal and Operating Modes

In default applications, the equalizer first operates in training mode to gather information about the channel. The equalizer later switches to decision-directed mode.

- When the equalizer is operating in training mode, the reference signal is a preset, known transmitted sequence.
- When the equalizer is operating in decision-directed mode, the reference signal is a detected version of the output signal, denoted by y_d in the schematic.

The CMA algorithm has no training mode. Training mode applies only when the equalizer is configured to use the LMS or RLS algorithm.

Error Calculation

The error calculation operation produces a signal given by this expression, where R is a constant related to the signal constellation.

$$e = \begin{cases} d - y & \text{LMS or RLS} \\ y(R - |y|^2) & \text{CMA} \end{cases}$$

Updating Tap Weights

- “Least Mean Square Algorithm” on page 6-11
- “Recursive Least Square Algorithm” on page 6-11
- “Constant Modulus Algorithm” on page 6-12

For linear and decision-feedback equalizer structures, the choice of LMS, RLS, or CMA determines the algorithms that are used to set the tap weights and perform the error calculation. The new set of tap weights depends on:

- The current set of tap weights
- The input signal

- The output signal
- The reference signal, d , for LMS and RLS adaptive algorithms only. The reference signal characteristics depend on the operating mode of the equalizer.

Least Mean Square Algorithm

For the LMS algorithm, in the previous schematic, w is a vector of all weights w_i , and u is a vector of all inputs u_i . Based on the current set of weights, the LMS algorithm creates the new set of weights as

$$w_{\text{new}} = w_{\text{current}} + (\text{StepSize}) u^*e.$$

The step size used by the adaptive algorithm is specified as a positive scalar. Increasing the step size reduces the equalizer convergence time but causes the equalized output signal to be less stable. To determine the maximum step size allowed when using the LMS adaptive algorithm, use the `maxstep` object function. The $*$ operator denotes the complex conjugate and the error calculation $e = d - y$.

Recursive Least Square Algorithm

For the RLS algorithm, in the previous schematic, w is the vector of all weights w_i , and u is the vector of all inputs u_i . Based on the current set of inputs, u , and the inverse correlation matrix, P , the RLS algorithm first computes the Kalman gain vector, K , as

$$K = \frac{Pu}{(\text{ForgettingFactor}) + u^H P u}.$$

The forgetting factor used by the adaptive algorithm is specified as a scalar in the range (0, 1]. Decreasing the forgetting factor reduces the equalizer convergence time but causes the equalized output signal to be less stable. H denotes the Hermitian transpose. Based on the current inverse correlation matrix, the new inverse correlation matrix is

$$P_{\text{new}} = \frac{P_{\text{current}}(1 - Ku^H)}{\text{ForgettingFactor}}.$$

Based on the current set of weights, the RLS algorithm creates the new set of weights as

$$w_{\text{new}} = w_{\text{current}} + K^*e.$$

The $*$ operator denotes the complex conjugate and the error calculation $e = d - y$.

Constant Modulus Algorithm

For the CMA adaptive algorithm, in the previous schematic, w is the vector of all weights w_i , and u is the vector of all inputs u_i . Based on the current set of weights, the CMA adaptive algorithm creates the new set of weights as

$$w_{\text{new}} = w_{\text{current}} + (\text{StepSize}) u^*e.$$

The step size used by the adaptive algorithm is specified as a positive scalar. Increasing the step size reduces the equalizer convergence time but causes the equalized output signal to be less stable. To determine the maximum step size allowed by the CMA adaptive algorithm, use the `maxstep` object function. The `*` operator denotes the complex conjugate and the error calculation $e = y(R - |y|^2)$, where R is a constant related to the signal constellation.

Configuring Adaptive Equalizers

Choose the linear or decision-feedback equalizer structure. Decide which adaptive algorithm to use — LMS, RLS, or CMA. Specify settings for structure and algorithm-specific operation modes.

Configuring an equalizer involves selecting a linear or decision-feedback structure, selecting an adaptive algorithm, and specifying the structure and algorithm specific operation modes.

- “Specify an Adaptive Equalizer” on page 6-13
- “Equalizer Training” on page 6-17
- “Managing Delays When Using Equalizers” on page 6-31

When deciding which adaptive algorithm best fits your needs, consider:

- The LMS algorithm executes quickly but converges slowly. Its complexity grows linearly with the number of weights.
- The RLS algorithm converges quickly. Its complexity grows approximately with the square of the number of weights. This algorithm can also be unstable when the number of weights is large.
- The constant modulus algorithm (CMA) is useful when no training signal is available. It works best for constant modulus modulations such as PSK.
 - If CMA has no additional side information, it can introduce phase ambiguity. For example, the weights found by the CMA might produce a perfect QPSK

constellation but introduce a phase rotation of 90, 180, or 270 degrees. In this case, employ a phase ambiguity correction algorithm or choose a differential modulation scheme. Differential modulation schemes are insensitive to phase ambiguity.

To view or change any properties of an adaptive equalizer, use the syntax described for channel objects in “Displaying and Changing Object Properties” on page 14-19.

For more information about adaptive algorithms, see the references listed in “Selected References for Equalizers” on page 6-4.

Specify an Adaptive Equalizer

- “Defining an Equalizer Object” on page 6-13
- “Adaptive Algorithm Assignment” on page 6-15

To create an adaptive equalizer object for use in MATLAB, select the `comm.LinearEqualizer` or `comm.DecisionFeedbackEqualizer` System object. For Simulink, use the Linear Equalizer or Decision Feedback Equalizer block. Based on the propagation channel characteristics in your simulation, use the criteria in “Equalization” on page 6-2 to select the equalizer structure.

The equalizer object has many properties that record information about the equalizer. Properties can be related to:

- The structure of the equalizer, such as the number of taps.
- The adaptive algorithm that the equalizer uses, such as the step size in the LMS or CMA algorithm.
- Information about the current state of the equalizer. The equalizer object can output the values of the weights.

To view or change any properties of an equalizer object, use the syntax described for channel objects in “Displaying and Changing Object Properties” on page 14-19.

Defining an Equalizer Object

The code creates equalizer objects for these configurations:

- A symbol-spaced linear RLS equalizer with 10 weights.
- A fractionally spaced linear RLS equalizer with 10 weights, a BPSK constellation, and two samples per symbol.

- A decision-feedback RLS equalizer with three weights in the feedforward filter and two weights in the feedback filter.

All three equalizer objects specify the RLS adaptive algorithm with a forgetting factor of 0.3.

Create equalizer objects of different types. The default settings are used for properties not set using 'Name, Value' pairs.

```
eqlin = comm.LinearEqualizer('Algorithm','RLS','NumTaps',10,'ForgettingFactor',0.3)
```

```
eqlin =  
comm.LinearEqualizer with properties:  
  
           Algorithm: 'RLS'  
           NumTaps: 10  
           ForgettingFactor: 0.3000  
InitialInverseCorrelationMatrix: 0.1000  
           Constellation: [1x4 double]  
           ReferenceTap: 3  
           InputDelay: 0  
InputSamplesPerSymbol: 1  
TrainingFlagInputPort: false  
           AdaptAfterTraining: true  
           InitialWeightsSource: 'Auto'  
           WeightUpdatePeriod: 1
```

```
eqfrac = comm.LinearEqualizer('Algorithm','RLS','NumTaps',10,'ForgettingFactor',0.3, .  
           'Constellation',[-1 1],'InputSamplesPerSymbol',2)
```

```
eqfrac =  
comm.LinearEqualizer with properties:  
  
           Algorithm: 'RLS'  
           NumTaps: 10  
           ForgettingFactor: 0.3000  
InitialInverseCorrelationMatrix: 0.1000  
           Constellation: [-1 1]  
           ReferenceTap: 3  
           InputDelay: 0  
InputSamplesPerSymbol: 2  
TrainingFlagInputPort: false  
           AdaptAfterTraining: true  
           InitialWeightsSource: 'Auto'
```

```
WeightUpdatePeriod: 1
```

```
eqdfe = comm.DecisionFeedbackEqualizer('Algorithm','RLS','NumForwardTaps',3, ...
    'NumFeedbackTaps',2,'ForgettingFactor',0.3)
```

```
eqdfe =
    comm.DecisionFeedbackEqualizer with properties:
```

```

        Algorithm: 'RLS'
        NumForwardTaps: 3
        NumFeedbackTaps: 2
        ForgettingFactor: 0.3000
InitialInverseCorrelationMatrix: 0.1000
        Constellation: [1x4 double]
        ReferenceTap: 3
        InputDelay: 0
InputSamplesPerSymbol: 1
TrainingFlagInputPort: false
AdaptAfterTraining: true
InitialWeightsSource: 'Auto'
        WeightUpdatePeriod: 1
```

Adaptive Algorithm Assignment

Use the `Algorithm` property to assign the adaptive algorithm used by the equalizer.

Algorithm Assignment

When creating the equalizer object, assign the adaptive algorithm.

```
eqlms = comm.LinearEqualizer('Algorithm','LMS');
```

Create the equalizer object with default property settings. LMS is the default adaptive algorithm.

```
eqrls = comm.LinearEqualizer;
eqrls.Algorithm
```

```
ans =
'LMS'
```

Update `eqrls` to use the RLS adaptive algorithm.

```
eqrls.Algorithm = 'RLS';  
eqrls.Algorithm
```

```
ans =  
'RLS'
```

Cloning and Duplicating Objects

Configure a new equalizer object by cloning an existing equalizer object, and then changing its properties. Clone `eqlms` to create an independent equalizer, `eqcma`, then update the algorithm to 'CMA'.

```
eqcma = clone(eqlms);  
eqcma.Algorithm
```

```
ans =  
'LMS'
```

```
eqcma.Algorithm = 'CMA';  
eqcma.Algorithm
```

```
ans =  
'CMA'
```

If you want an independent duplicate, use the `clone` command.

```
eqlms.NumTaps
```

```
ans = 5
```

```
eq2 = eqlms;  
eq2.NumTaps = 6;  
eq2.NumTaps
```

```
ans = 6
```

```
eqlms.NumTaps
```

```
ans = 6
```

The `clone` command creates a copy of `eqLms` that is independent of `eqLms`. By contrast, the command `eqB = eqA` creates `eqB` as a reference to `eqA`, so that `eqB` and `eqA` always have identical property settings.

Equalizer Training

- “Linearly Equalize System By Using Different Training Schemes” on page 6-17
- “Linearly Equalize Symbols By Using EVM-Based Training” on page 6-26

Linearly Equalize System By Using Different Training Schemes

Demonstrate linear equalization by using the least mean squares (LMS) algorithm to recover QPSK symbols passed through an AWGN channel. Apply different equalizer training schemes and show the symbol error magnitude.

System Setup

Simulate a QPSK-modulated system subject to AWGN. Transmit packets composed of 200 training symbols and 1800 random data symbols. Configure a linear LMS equalizer to recover the packet data.

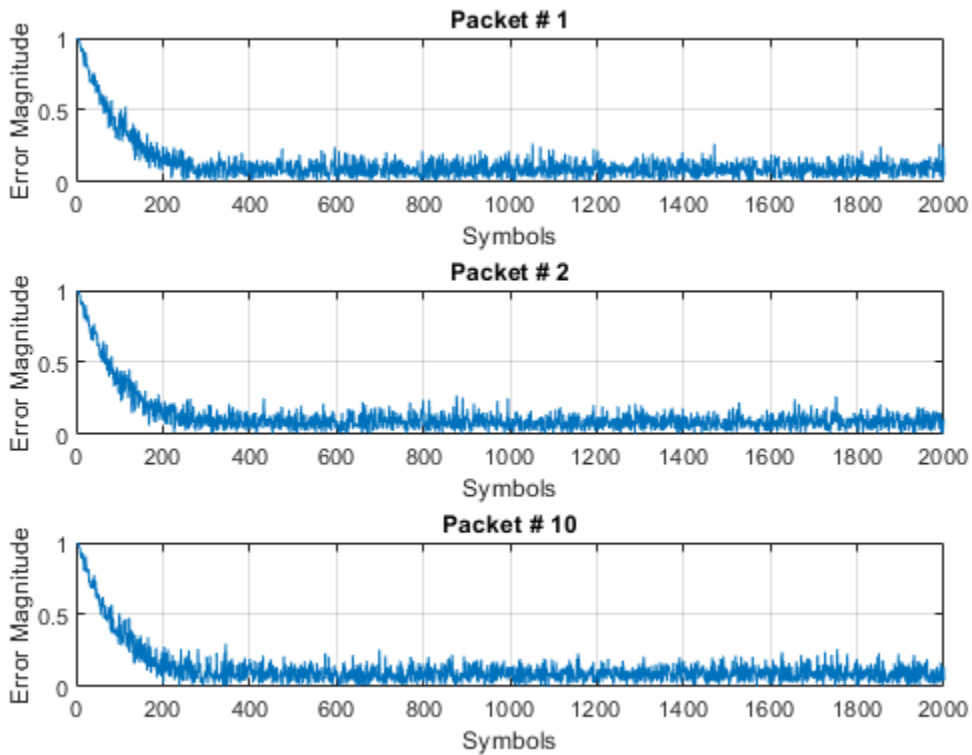
```
M = 4;
numTrainSymbols = 200;
numDataSymbols = 1800;
SNR = 20;
trainingSymbols = pskmod(randi([0 M-1],numTrainSymbols,1),M,pi/4);
numPkts = 10;
lineq = comm.LinearEqualizer('Algorithm','LMS', ...
    'NumTaps',5,'ReferenceTap',3,'StepSize',0.01);
```

Train the Equalizer at the Beginning of Each Packet With Reset

Use prepended training symbols when processing each packet. After processing each packet, reset the equalizer. This reset forces the equalizer to train the taps with no previous knowledge. Equalizer error signal plots for the first, second, and last packet show higher symbol errors at the start of each packet.

```
jj = 1;
figure
for ii = 1:numPkts
    b = randi([0 M-1],numDataSymbols,1);
    dataSym = pskmod(b,M,pi/4);
    packet = [trainingSymbols;dataSym];
    rx = awgn(packet,SNR);
```

```
[~,err] = lineq(rx,trainingSymbols);  
reset(lineq)  
if (ii ==1 || ii == 2 ||ii == numPkts)  
    subplot(3,1,jj)  
    plot(abs(err))  
    title(['Packet # ',num2str(ii)])  
    xlabel('Symbols')  
    ylabel('Error Magnitude')  
    axis([0,length(packet),0,1])  
    grid on;  
    jj = jj+1;  
end  
end
```



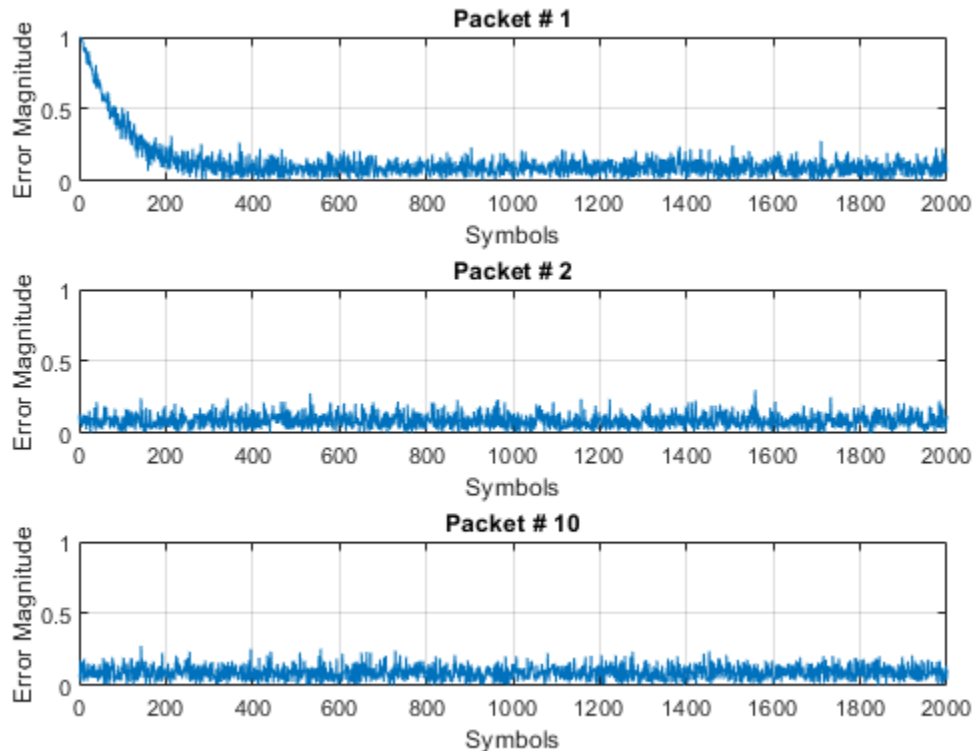
Train the Equalizer at the Beginning of Each Packet Without Reset

Process each packet using prepended training symbols. Do not reset the equalizer after each packet is processed. By not resetting after each packet, the equalizer retains tap weights from training prior packets. Equalizer error signal plots for the first, second, and last packet show that after the initial training on the first packet, subsequent packets have less symbol errors at the start of each packet.

```

release(lineq)
jj = 1;
figure
for ii = 1:numPkts
    b = randi([0 M-1],numDataSymbols,1);
    dataSym = pskmod(b,M,pi/4);
    packet = [trainingSymbols;dataSym];
    channel = 1;
    rx = awgn(packet*channel,SNR);
    [~,err] = lineq(rx,trainingSymbols);
    if (ii ==1 || ii == 2 ||ii == numPkts)
        subplot(3,1,jj)
        plot(abs(err))
        title(['Packet # ',num2str(ii)])
        xlabel('Symbols')
        ylabel('Error Magnitude')
        axis([0,length(packet),0,1])
        grid on;
        jj = jj+1;
    end
end

```



Train the Equalizer Periodically

Systems with signals subject to time-varying channels require periodic equalizer training to maintain lock on the channel variations. Specify a system that has 200 symbols of training for every 1800 data symbols. Between training, the equalizer does not update tap weights. The equalizer processes 200 symbols per packet.

```
Rs = 1e6;
fd = 20;
spp = 200; % Symbols per packet
b = randi([0 M-1],numDataSymbols,1);
dataSym = pskmod(b,M,pi/4);
packet = [trainingSymbols; dataSym];
stream = repmat(packet,10,1);
```



```
tx = (0:length(stream)-1)/Rs;
channel = exp(1i*2*pi*fd*tx);
rx = awgn(stream.*channel,SNR);
```

Set the `AdaptAfterTraining` property to `false` to stop the equalizer tap weight updates after the training phase.

```
release(lineq)
lineq.AdaptAfterTraining = false

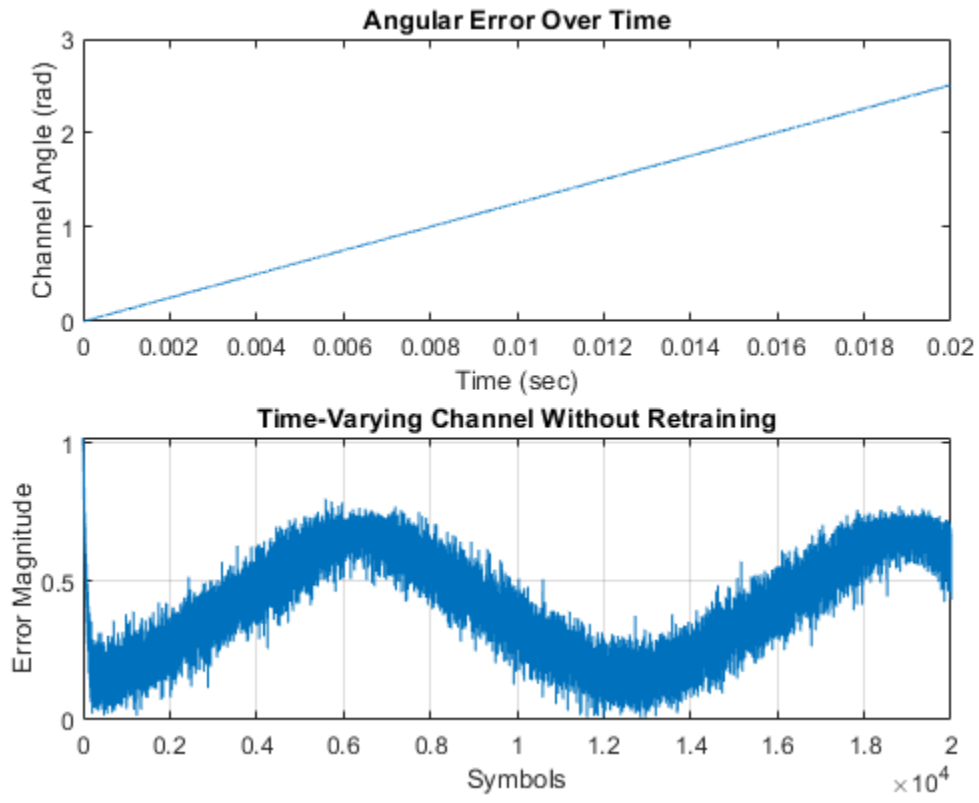
lineq =
  comm.LinearEqualizer with properties:

        Algorithm: 'LMS'
        NumTaps: 5
        StepSize: 0.0100
  Constellation: [1x4 double]
  ReferenceTap: 3
  InputDelay: 0
  InputSamplesPerSymbol: 1
  TrainingFlagInputPort: false
  AdaptAfterTraining: false
  InitialWeightsSource: 'Auto'
  WeightUpdatePeriod: 1
```

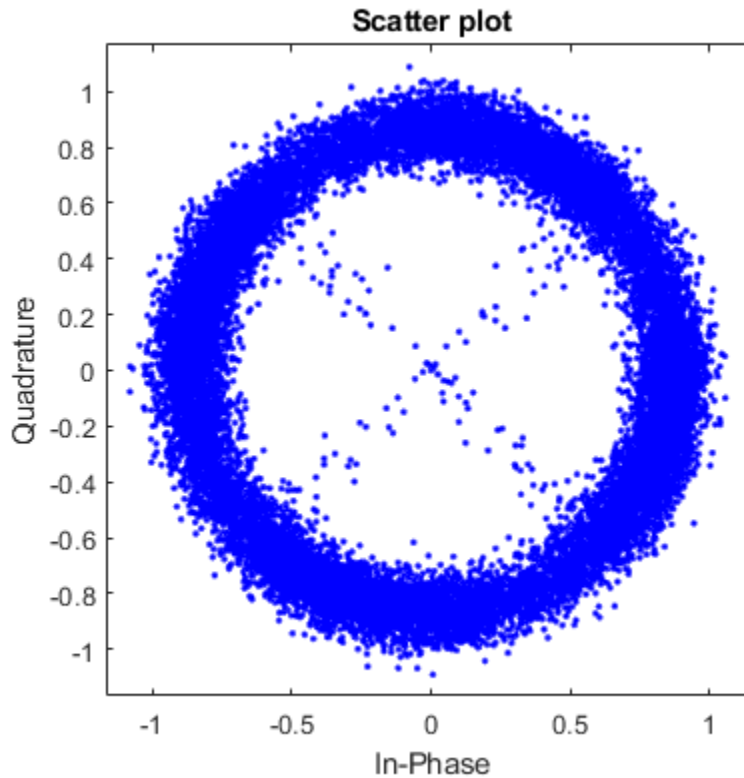
Equalize the impaired data. Plot the angular error from the channel, the equalizer error signal, and signal constellation. As the channel varies, the equalizer output does not remove the channel effects. The output constellation rotates out of sync, resulting in bit errors.

```
[y,err] = lineq(rx,trainingSymbols);

figure
subplot(2,1,1)
plot(tx, unwrap(angle(channel)))
xlabel('Time (sec)')
ylabel('Channel Angle (rad)')
title('Angular Error Over Time')
subplot(2,1,2)
plot(abs(err))
xlabel('Symbols')
ylabel('Error Magnitude')
grid on
title('Time-Varying Channel Without Retraining')
```



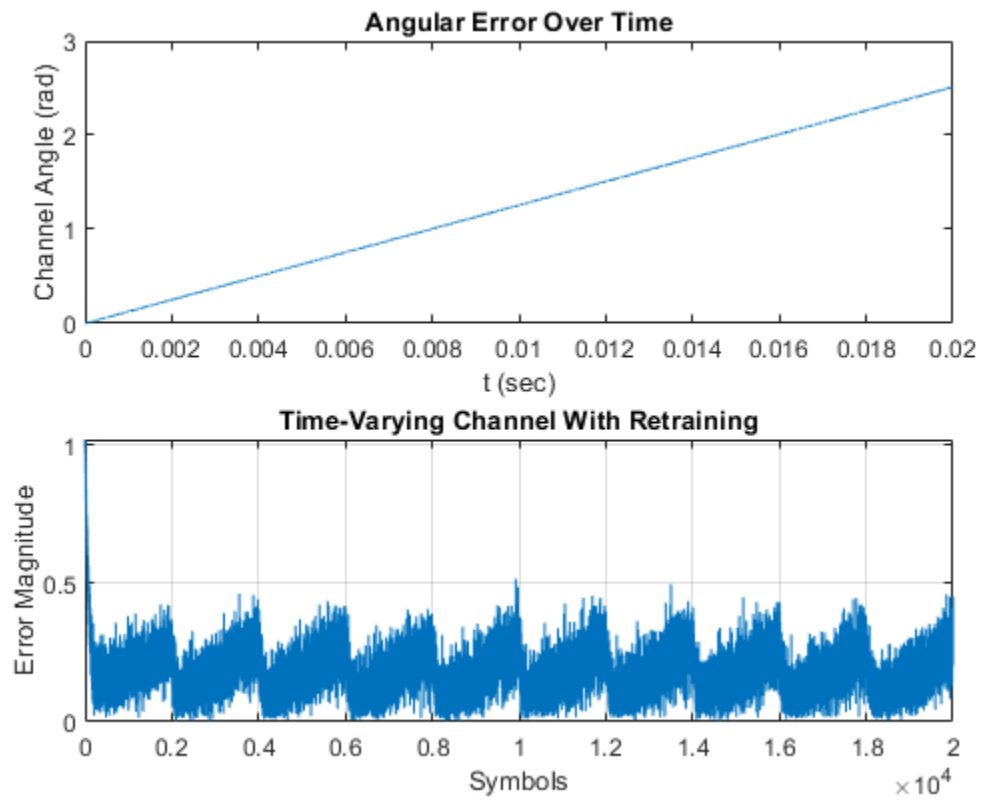
scatterplot(y)



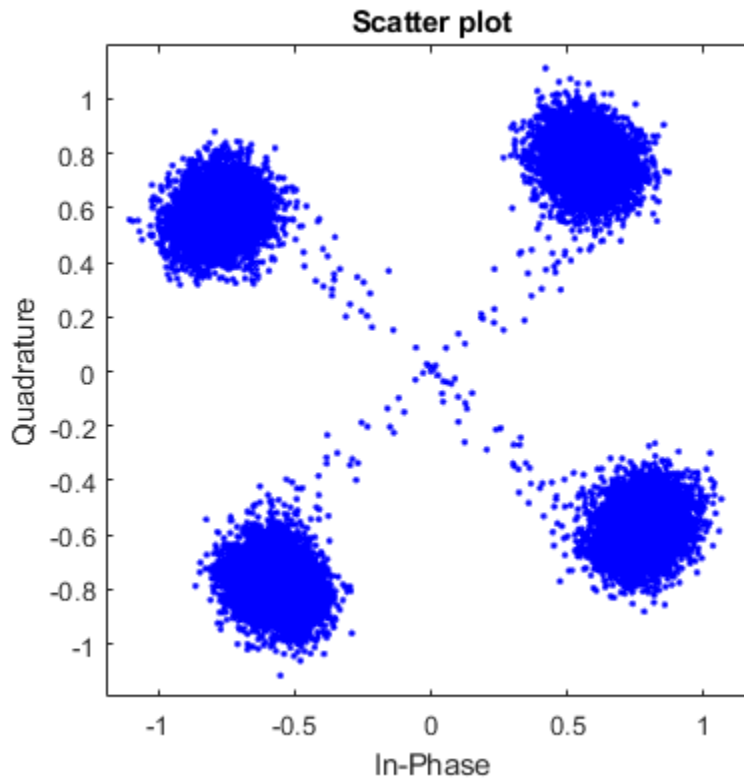
Set the `TrainingInputPort` property to `true` to configure the equalizer to retrain the taps when signaled by the `trainFlag` input. The equalizer trains only when `trainFlag` is `true`. After every 2000 symbols, the equalizer retrains the taps and keeps lock on variations of the channel. Plot the angular error from the channel, equalizer error signal, and signal constellation. As the channel varies, the equalizer output removes the channel effects. The output constellation does not rotate out of sync and bit errors are reduced.

```
release(lineq)
lineq.TrainingFlagInputPort = true;
symbolCnt = 0;
numPackets = length(rx)/spp;
trainFlag = true;
trainingPeriod = 2000;
eVec = zeros(size(rx));
```

```
yVec = zeros(size(rx));
for p=1:numPackets
    [yVec((p-1)*spp+1:p*spp,1),eVec((p-1)*spp+1:p*spp,1)] = ...
        lineq(rx((p-1)*spp+1:p*spp,1),trainingSymbols,trainFlag);
    symbolCnt = symbolCnt + spp;
    if symbolCnt >= trainingPeriod
        trainFlag = true;
        symbolCnt = 0;
    else
        trainFlag = false;
    end
end
figure
subplot(2,1,1)
plot(tx, unwrap(angle(channel)))
xlabel('t (sec)')
ylabel('Channel Angle (rad)')
title('Angular Error Over Time')
subplot(2,1,2)
plot(abs(eVec))
xlabel('Symbols')
ylabel('Error Magnitude')
grid on
title('Time-Varying Channel With Retraining')
```



scatterplot(yVec)



Linearly Equalize Symbols By Using EVM-Based Training

Recover QPSK symbols with a linear equalizer by using the constant modulus algorithm (CMA) and EVM-based taps training. When using blind equalizer algorithms, such as CMA, train the equalizer taps by using the `AdaptWeights` property to start and stop training. Helper functions are used to generate plots and apply phase correction.

Initialize system variables.

```
rng(123456);
M = 4; % QPSK
numSymbols = 100;
numPackets = 5000;
raylChan = comm.RayleighChannel('PathDelays',[0 1], ...
```

```

    'AveragePathGains',[0 -12], 'MaximumDopplerShift',1e-5);
SNR = 50;
adaptWeights = true;

```

Create the equalizer and EVM System objects. The equalizer System object specifies a linear equalizer by using the CMA adaptive algorithm. Call the helper function to initialize figure plots.

```

lineq = comm.LinearEqualizer('Algorithm','CMA', ...
    'NumTaps',5,'ReferenceTap',3, ...
    'StepSize',0.03,'AdaptWeightsSource','Input port')

```

```

lineq =
    comm.LinearEqualizer with properties:

        Algorithm: 'CMA'
        NumTaps: 5
        StepSize: 0.0300
        Constellation: [1x4 double]
        ReferenceTap: 3
        InputSamplesPerSymbol: 1
        AdaptWeightsSource: 'Input port'
        InitialWeightsSource: 'Auto'
        WeightUpdatePeriod: 1

```

```

info(lineq)

```

```

ans = struct with fields:
    Latency: 2

```

```

evm = comm.EVM('ReferenceSignalSource', ...
    'Estimated from reference constellation');
[errPlot, evmPlot, scatSym, adaptState] = initFigures(numPackets, lineq);

```

Equalization Loop

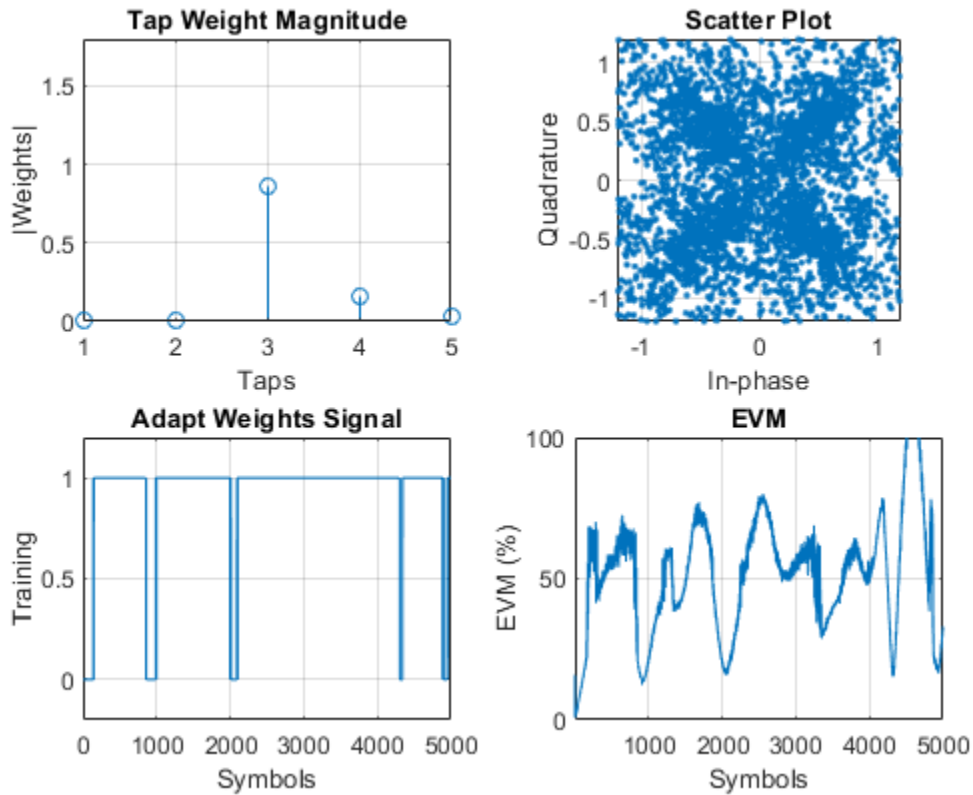
To implement the equalization loop:

- 1 Generate PSK data packets.
- 2 Apply Rayleigh fading and AWGN to the transmission data.
- 3 Apply equalization to the received data and phase correction to the equalizer output.

- 4 Estimate the EVM and toggle the `adaptWeights` flag to `true` or `false` based on the EVM level.
- 5 Update the figure plots.

```
for p=1:numPackets
    data = randi([0 M-1],numSymbols,1);
    tx = pskmod(data,M,pi/4);
    rx = awgn(raylChan(tx),SNR);
    rxDelay = finddelay(rx,tx);
    [y,err,wts] = lineq(rx,adaptWeights);
    y = phaseCorrection(y);
    evmEst = evm(y);
    adaptWeights = (evmEst > 20);

    updateFigures(errPlot,evmPlot,scatSym,adaptState, ...
        wts,y(end),evmEst,adaptWeights,p,numPackets)
end
```

rxDelay

rxDelay = 0

The figure plots show that, as the EVM varies, the equalizer toggles in and out of decision-directed weight adaptation mode.

Helper Functions

This helper function initializes figures that show a quad plot of simulation results.

```
function [errPlot,emvPlot,scatter,adaptState] = initFigures(numPkts,lineq)
yVec = nan(numPkts,1);
emvVec = nan(numPkts,1);
```

```
wVec = zeros(lineq.NumTaps,1);
adaptVec = nan(numPkts,1);

figure
subplot(2,2,1)
evmPlot = stem(wVec);
grid on; axis([1 lineq.NumTaps 0 1.8])
xlabel('Taps'); ylabel('|Weights|'); title('Tap Weight Magnitude')

subplot(2,2,2)
scatter = plot(yVec, '.');
axis square; axis([-1.2 1.2 -1.2 1.2]); grid on
xlabel('In-phase'); ylabel('Quadrature'); title('Scatter Plot');
subplot(2,2,3)
adaptState = plot(adaptVec);
grid on; axis([0 numPkts -0.2 1.2])
ylabel('Training'); xlabel('Symbols'); title('Adapt Weights Signal')
subplot(2,2,4)
errPlot = plot(evmVec);
grid on; axis([1 numPkts 0 100])
xlabel('Symbols'); ylabel('EVM (%)'); title('EVM')
end
```

This helper function updates figures.

```
function updateFigures(errPlot, evmPlot, scatSym, ...
    adaptState, w, y, evmEst, adaptWts, p, numFrames)
persistent yVec evmVec adaptVec

if p == 1
    yVec = nan(numFrames,1);
    evmVec = nan(numFrames,1);
    adaptVec = nan(numFrames,1);
end

yVec(p) = y;
evmVec(p) = evmEst;
adaptVec(p) = adaptWts;

errPlot.YData = abs(evmVec);
evmPlot.YData = abs(w);
scatSym.XData = real(yVec);
scatSym.YData = imag(yVec);
adaptState.YData = adaptVec;
```

```
drawnow limitrate
end
```

This helper function applies phase correction.

```
function y = phaseCorrection(y)
a = angle(y((real(y) > 0) & (imag(y) > 0)));
a(a < 0.1) = a(a < 0.1) + pi/2;
theta = mean(a) - pi/4;
y = y * exp(-1i*theta);
end
```

Managing Delays When Using Equalizers

For proper equalization, you must determine and account for system delays. As shown in the following example, you can use the `finddelay` function to determine the system delay. This example uses LMS linear equalization but the same approach is valid for the RLS and CMA adaptive algorithms and for decision feedback equalizers.

Linearly Equalize Delayed Signal

Simulate a system with delay between the transmitted symbols and received samples. Typical systems have transmitter and receiver filters that result in a delay. This delay must be accounted for to synchronize the system. In this example, the system delay is introduced without transmit and receive filters. Linear equalization, using the least mean squares (LMS) algorithm, recovers QPSK symbols.

Initialize simulation variables.

```
M = 4; % QPSK
numSymbols = 10000;
numTrainingSymbols = 1000;
mpChan = [1 0.5*exp(1i*pi/6) 0.1*exp(-1i*pi/8)];
systemDelay = dsp.Delay(20);
snr = 24;
```

Generate QPSK-modulated symbols. Apply multipath channel filtering, a system delay, and AWGN to the transmitted symbols.

```
data = randi([0 M-1],numSymbols,1);
tx = pskmod(data,M,pi/4); % OQPSK
delayedSym = systemDelay(filter(mpChan,1,tx));
rx = awgn(delayedSym,snr,'measured');
```

Create equalizer and EVM System objects. The equalizer System object specifies a linear equalizer that uses the LMS algorithm.

```
lineq = comm.LinearEqualizer('Algorithm','LMS', ...  
    'NumTaps',9,'ReferenceTap',5);  
evm = comm.EVM('ReferenceSignalSource', ...  
    'Estimated from reference constellation');
```

Equalize Without Adjusting Input Delay

Equalize the received symbols.

```
[y1,err1,wts1] = lineq(rx,tx(1:numTrainingSymbols,1));
```

Find the delay between the received symbols and the transmitted symbols by using the `finddelay` function.

```
rxDelay = finddelay(tx,rx)
```

```
rxDelay = 20
```

Display the equalizer information. The latency value indicates the delay introduced by the equalizer. Calculate the total delay as the sum of `rxDelay` and the equalizer latency.

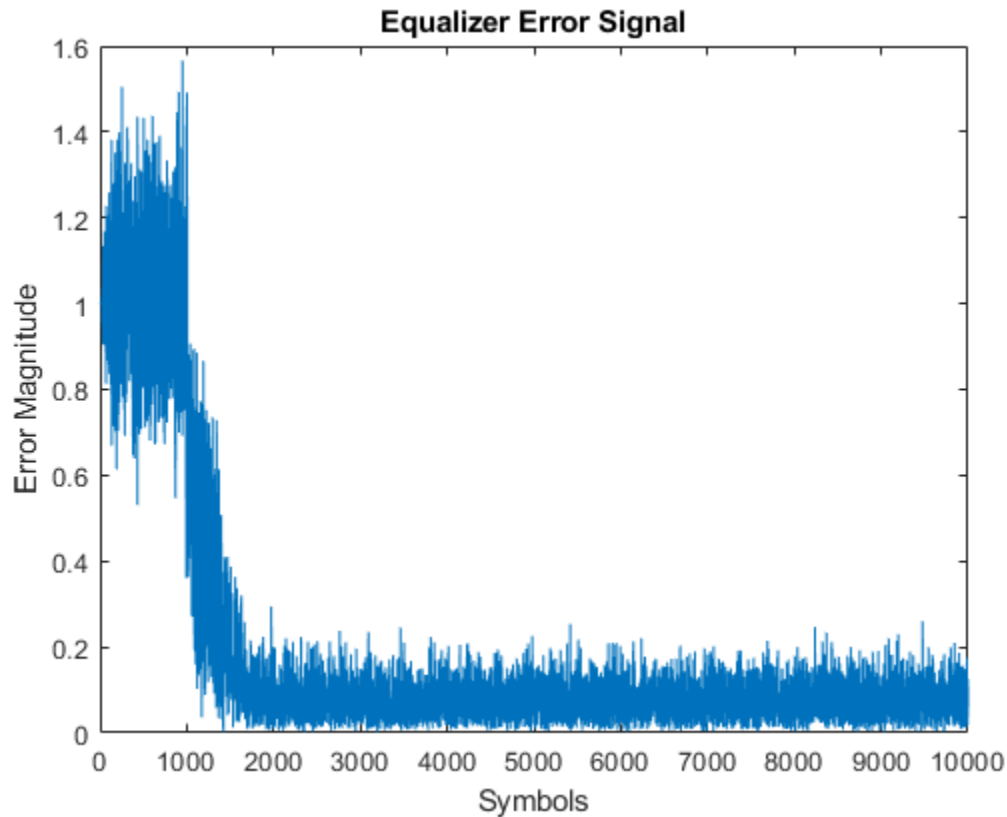
```
eqInfo = info(lineq)
```

```
eqInfo = struct with fields:  
    Latency: 4
```

```
totalDelay = rxDelay + eqInfo.Latency;
```

Until the equalizer output converges, the symbol error rate is high. Plot the error output, `err1`, to determine when the equalized output converges.

```
plot(abs(err1))  
xlabel('Symbols')  
ylabel('Error Magnitude')  
title('Equalizer Error Signal')
```



The plot shows excessive errors beyond the 1000 symbols training period. When demodulating symbols and computing symbol errors, to account for the unconverged output and the system delay between the equalizer output and transmitted symbols, skip the first 2000 symbols.

```
dataRec1 = pskdemod(y1(2000+totalDelay:end),M,pi/4);
symErrWithDelay = symerr(data(2000:end-totalDelay),dataRec1)
```

```
symErrWithDelay = 5999
```

```
evmWithDelay = evm(y1)
```

```
evmWithDelay = 29.5795
```

The error rate and EVM are high because the receive delay was not accounted for in the equalizer System object.

Adjust Input Delay in Equalizer

Equalize the received data by using the delay value to set the `InputDelay` property. Because `InputDelay` is a nontunable property, you must release the `lineq` System object to reconfigure the `InputDelay` property. Equalize the received symbols.

```
release(lineq)
lineq.InputDelay = rxDelay

lineq =
    comm.LinearEqualizer with properties:

        Algorithm: 'LMS'
        NumTaps: 9
        StepSize: 0.0100
        Constellation: [1x4 double]
        ReferenceTap: 5
        InputDelay: 20
        InputSamplesPerSymbol: 1
        TrainingFlagInputPort: false
        AdaptAfterTraining: true
        InitialWeightsSource: 'Auto'
        WeightUpdatePeriod: 1

[y2,err2,wts2] = lineq(rx,tx(1:numTrainingSymbols,1));
```

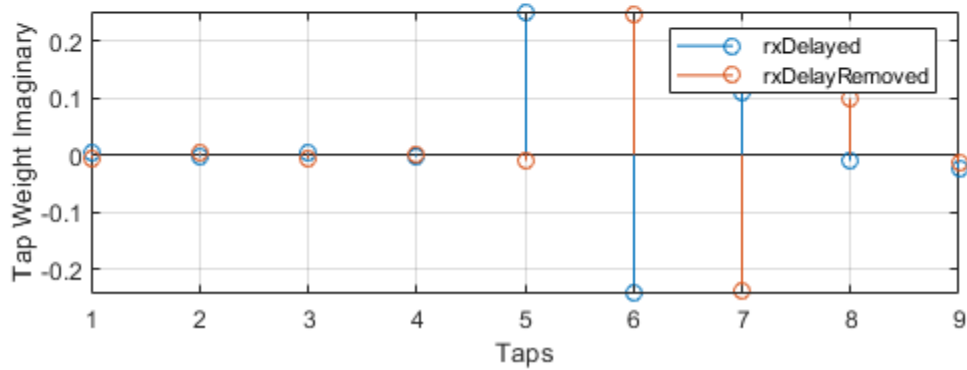
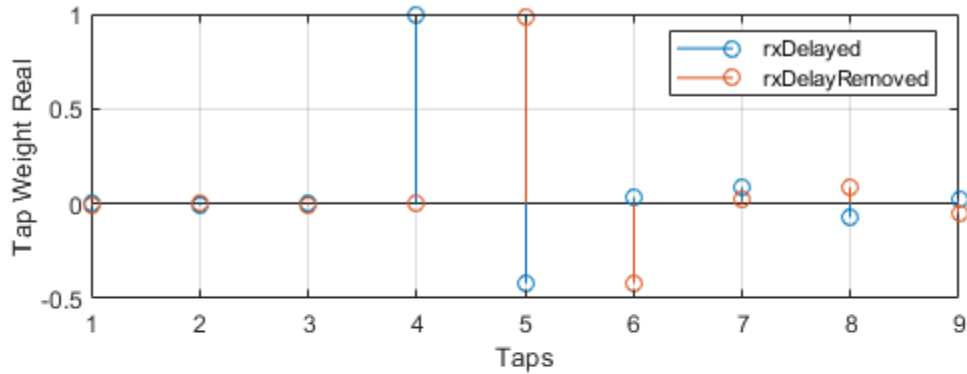
Plot the tap weights and equalized error magnitude. A stem plot shows the equalizer tap weights before and after the system delay is removed. A 2-D line plot shows the slower equalizer convergence for the delayed signal as compared to the signal with the delay removed.

```
subplot(2,1,1)
stem([real(wts1),real(wts2)])
xlabel('Taps')
ylabel('Tap Weight Real')
legend('rxDelayed','rxDelayRemoved')
grid on
subplot(2,1,2)
stem([imag(wts1),imag(wts2)])
xlabel('Taps')
ylabel('Tap Weight Imaginary')
```

```

legend('rxDelayed', 'rxDelayRemoved')
grid on

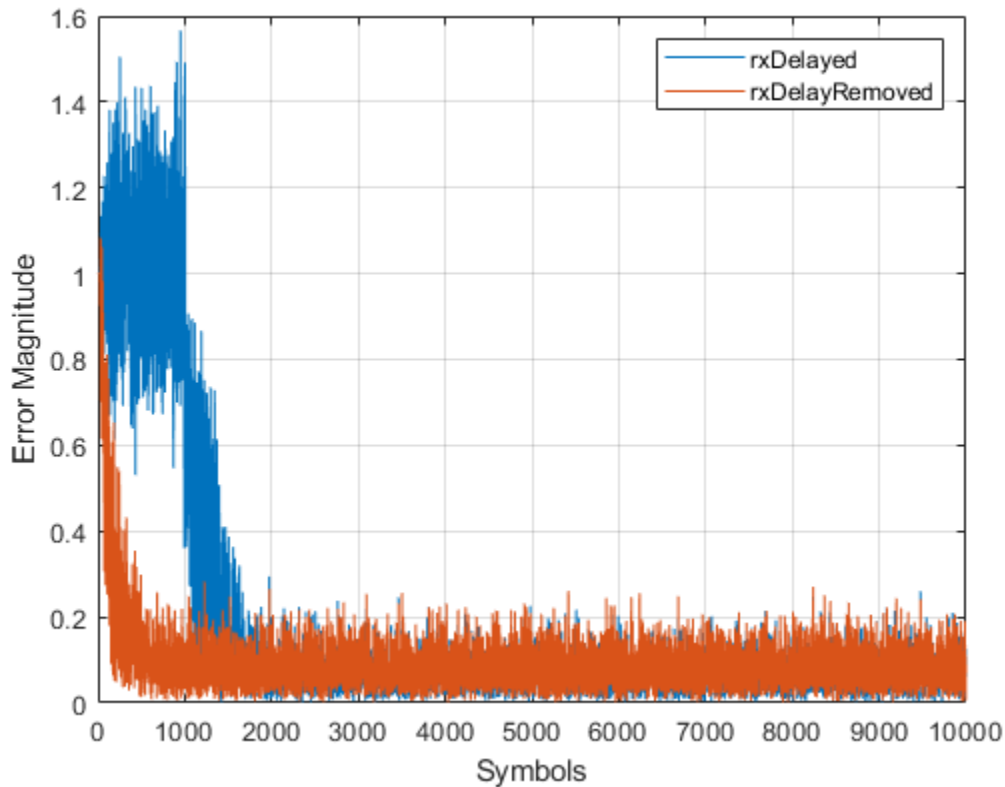
```



```

figure
plot([abs(err1),abs(err2)])
xlabel('Symbols')
ylabel('Error Magnitude')
legend('rxDelayed', 'rxDelayRemoved')
grid on

```



Plot error output of the equalized signals, `rxDelayed` and `rxDelayRemoved`. For the signal that has the delay removed, the equalizer converges during the 1000 symbol training period. When demodulating symbols and computing symbol errors, to account for the unconverged output and the system delay between the equalizer output and transmitted symbols, skip the first 500 symbols. Reconfiguring the equalizer to account for the system delay enables better equalization of the signal, and reduces symbol errors and the EVM.

```
eqInfo = info(lineq)
```

```
eqInfo = struct with fields:
    Latency: 4
```



```

totalDelay = rxDelay + eqInfo.Latency;
dataRec2 = pskdemod(y2(500+totalDelay:end),M,pi/4);
symErrDelayRemoved = symerr(data(500:end-totalDelay),dataRec2)

symErrDelayRemoved = 0

evmDelayRemoved = evm(y2(500+totalDelay:end))

evmDelayRemoved = 9.4435

```

Using Adaptive Equalizers in Simulink

Adaptive Equalization with Filtering and Fading Channel

This model shows the behavior of the selected adaptive equalizer in a communication link that has a fading channel. The transmitter and receiver have root raised cosine pulse shaped filtering. A subsystem block enables you to select between linear or decision feedback equalizers that use the least mean square (LMS) or recursive least square (RLS) adaptive algorithm.

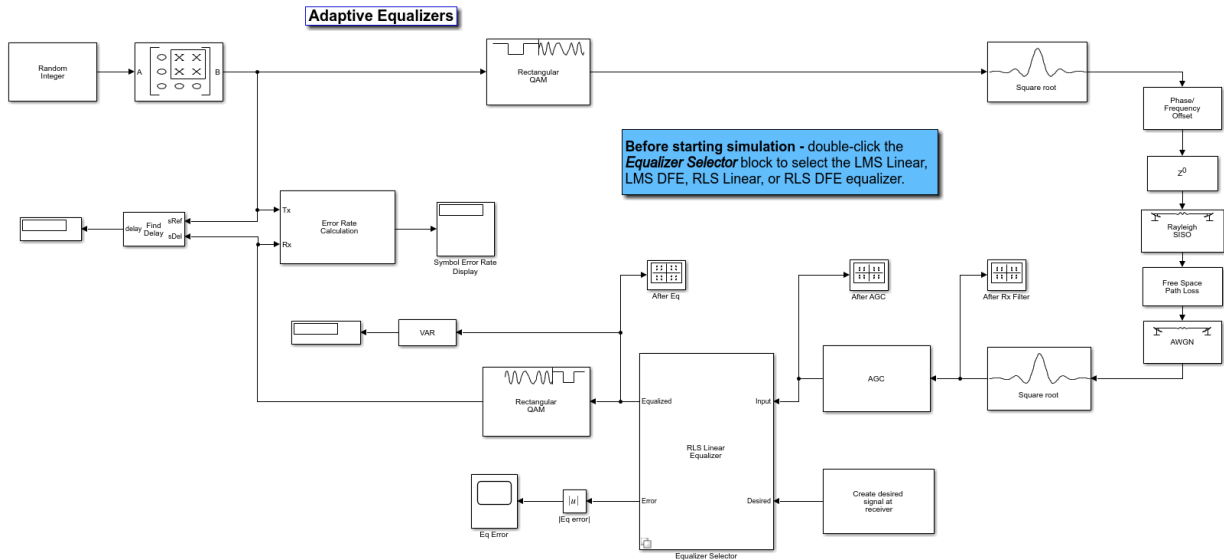
Model Structure

- The transmitter generates 16QAM random signal data that includes a training sequence and applies root raised cosine pulse shaped filtering.
- Channel impairments include multipath fading, Doppler shift, carrier frequency offset, variable integer delay, free space path loss, and AWGN.
- The receiver applies root raised cosine pulse shaped filtering, AGC, includes equalizer mode control to enable training and enables you to select the equalizer algorithm from these choices.

Selection	Equalizer Algorithm
LMS Linear	Linear least mean square equalizer
LMS DFE	Decision feedback least mean square equalizer
RLS Linear	Linear recursive least square equalizer
RLS DFE	Decision feedback recursive least square equalizer

- Scopes help you understand how the different equalizers and adaptive algorithms behave.

Explore Example Model



Experimenting with the model

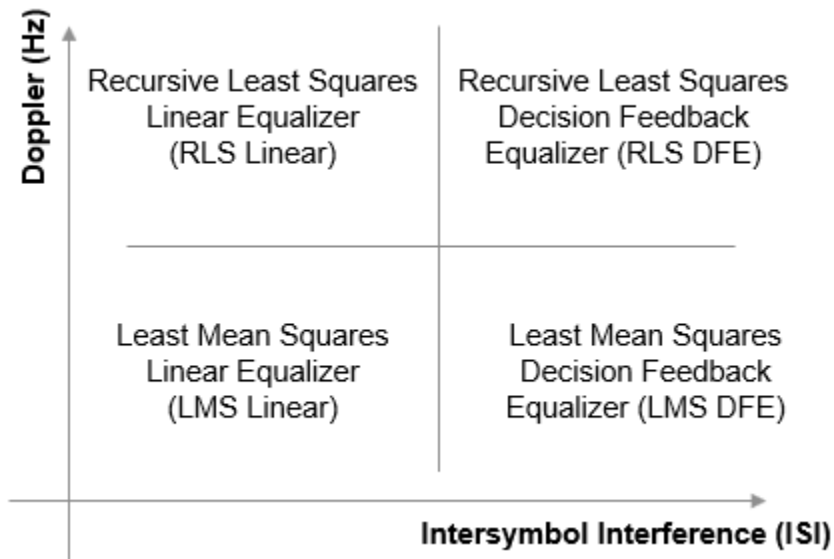
This model provides several ways for you to change settings and observe the results. The `InitFcn` found in `File>Model Properties>Callbacks` calls `cm_ex_adaptive_eq_with_fading_init` to initialize the model. This file enables you to vary settings in the model, including:

- System parameters, such as SNR.
- Pulse shaping filter parameters, such as rolloff and filter length
- Path loss value.
- Channel conditions: Rayleigh or Rician fading, channel path gains, channel path delays, and Doppler shift.
- Equalizer choice and configuration.

Model Considerations

This non-standards-based communication link is representative of a modern communications system.

- The optimal equalizer configuration depends on the channel conditions. The initialization file sets the Doppler shift and multipath fading channel parameters that highlight the capabilities of different equalizers.



- The decision feedback equalizer structure performs better than the linear equalizer structure for higher intersymbol interference.
- The RLS algorithm performs better than the LMS algorithm for higher Doppler frequencies.
- The LMS algorithm executes quickly, converges slowly, and its complexity grows linearly with the number of weights.
- The RLS algorithm converges quickly, its complexity grows approximately as the square of the number of weights. It can be unstable when the number of weights is large.
- The channels exercised for different equalizers have the following characteristics.

Selection	Channel Characteristics
LMS Linear	3 tap multipath fading channel with 10 Hz Doppler shift
LMS DFE	5 tap multipath fading channel with 25 Hz Doppler shift
RLS Linear	3 tap multipath fading channel with 50 Hz Doppler shift
RLS DFE	5 tap multipath fading channel with 100 Hz Doppler shift

- Initial settings for other channel impairments are the same for all equalizers. Carrier frequency offset value is set to 50 Hz. Free space path loss is set to 60 dB. Variable integer delay is set to 2 samples, which requires the equalizers to perform some timing recovery.

Deep channel fades and path loss can cause the equalizer input signal level to be much less than the desired output signal level and result in unacceptably long equalizer convergence time. The AGC block adjusts the magnitude of received signal to reduce the equalizer convergence time. You must adjust the optimal AGC output power level based on the modulation scheme selected. For 16QAM, a desired output power of 10 W is used.

Training of the equalizer is performed at the beginning of the simulation.

Running the Simulation

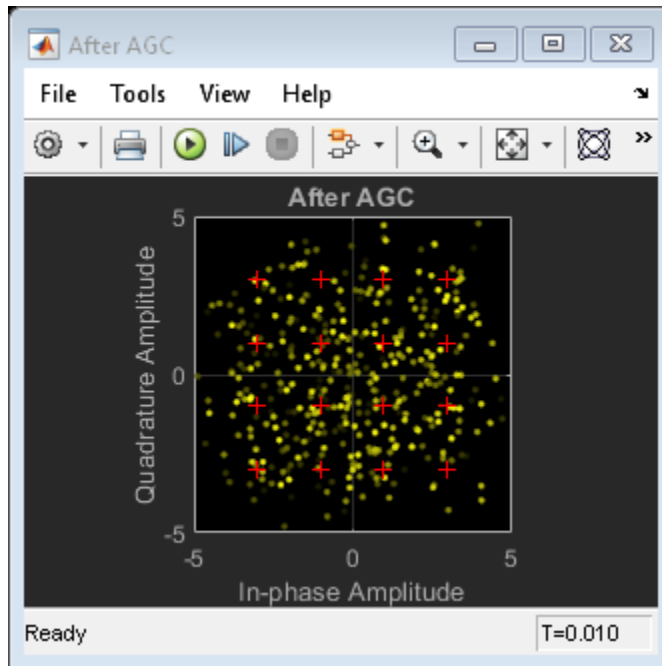
Running the simulation computes symbol error statistics and produces these figures:

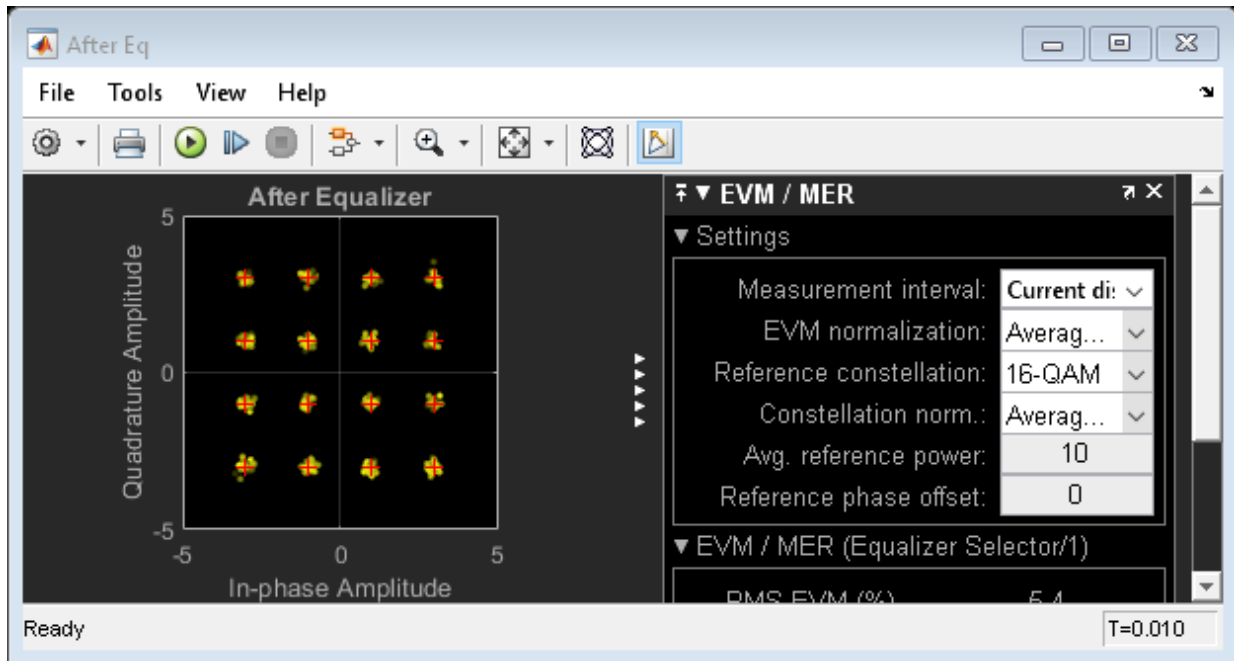
- A constellation diagram of the signal after the receive filter.
- A constellation diagram of the signal after AGC.
- A constellation diagram of the signal after equalization with signal quality measurements shown.
- An equalizer error plot.

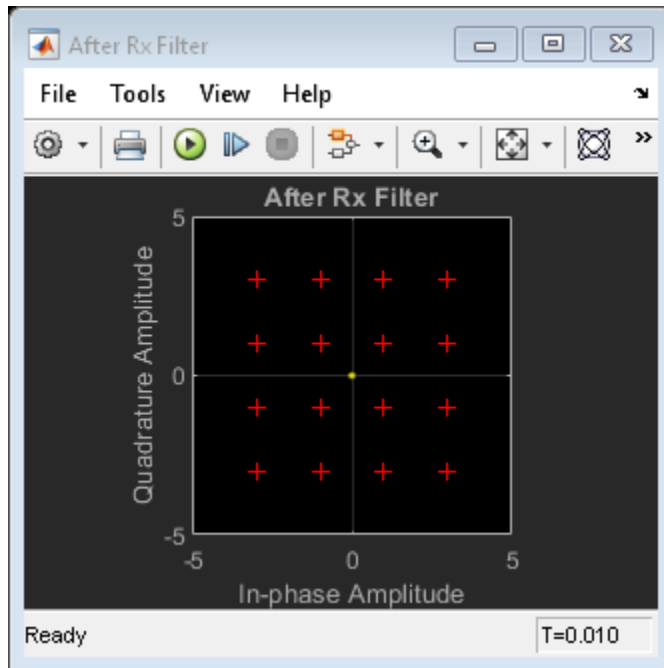
For the plots shown here, the equalizer algorithm selected is RLS Linear. Monitoring these figures, you can see that the received signal quality fluctuates as simulation time progresses.

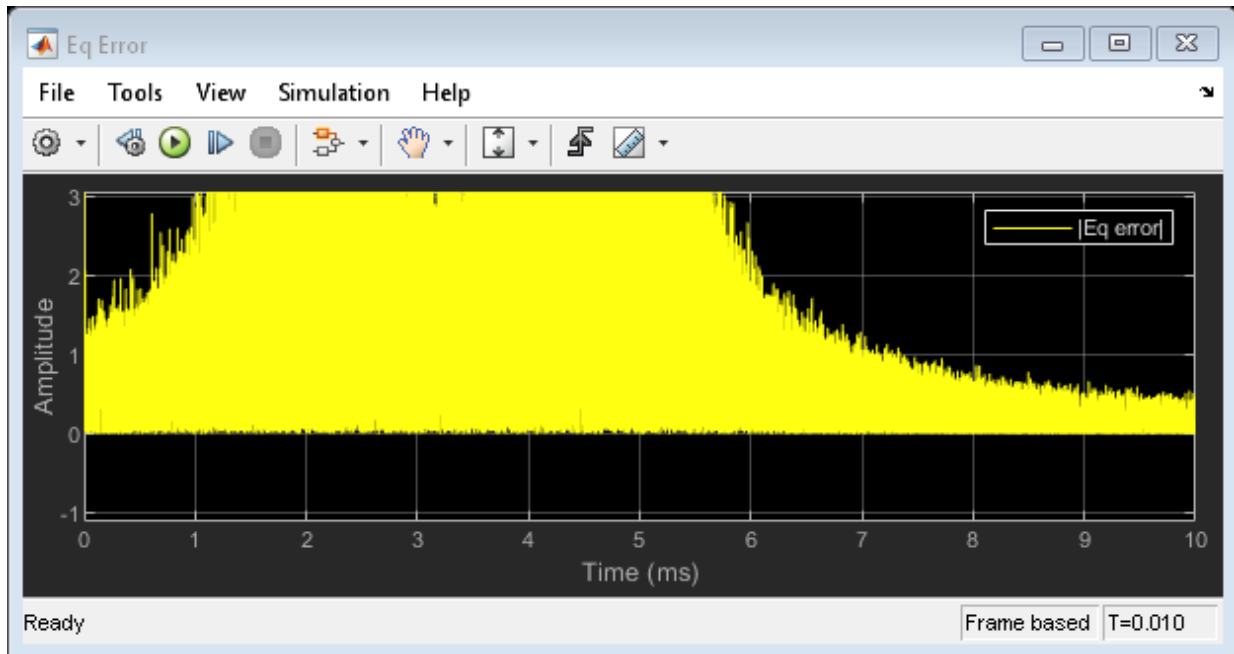
The *After Rx Filter* and *After AGC* constellation plots show the signal before equalization. *After AGC* shows the impact of the channel conditions on the transmitted signal. The *After Eq* plot shows the signal after equalization. The signal plotted in the constellation diagram after equalization shows the variation in signal quality based on the effectiveness of the equalization process. Throughout the simulation, the signal constellations plotted before equalization deviate noticeably from a 16QAM signal constellation. The *After Eq*

constellation improves or degrades as the equalizer error signal varies. The *Eq error* plotted in the *Eq Error* plot, indicates poor equalization at the start of the simulation. The error degrades at first then improves as the equalizer converges.









Further Exploration

Double-click the Equalizer Selector block and select a different equalizer. Run the simulation to see the performance of the various equalizer options. You can use the signal logger to compare the results from this experimentation. In the block diagram, right-click on signal wires and select Log Selected Signals. If you have enabled signal logging, after the simulation run finishes, open the Simulation Data Inspector to view the logged signals.

At the MATLAB™ command prompt, enter `edit cm_ex_adaptive_eq_with_fading_init.m` to open the initialization file, then modify a parameter and rerun the simulation. For example, adjust the channel characteristics (`params.maxDoppler`), `params.pathDelays`, and `params.pathGains`). The RLS

adaptive algorithm performs better than the LMS adaptive algorithm as the maximum Doppler is increased.

See Also

Objects

`comm.DecisionFeedbackEqualizer` | `comm.LinearEqualizer` |
`comm.MLSEEqualizer`

Blocks

Decision Feedback Equalizer | Linear Equalizer | MLSE Equalizer

More About

- “Equalization” on page 6-2
- “MLSE Equalizers” on page 6-46

MLSE Equalizers

In this section...
“Equalize a Vector Signal in MATLAB” on page 6-47
“Equalizing Signals in Continuous Operation Mode” on page 6-48
“Use a Preamble or a Postamble” on page 6-52
“Using MLSE Equalizers in Simulink” on page 6-54
“MLSE Equalization with Dynamically Changing Channel” on page 6-54

Maximum-Likelihood Sequence Estimation (MLSE) equalizers provide optimal equalization of time variations in the propagation channel characteristics. However, MLSE equalizers are sometimes less appealing because their computational complexity is higher than “Adaptive Equalizers” on page 6-6.

In Communications Toolbox, the `mlseeq` function, `comm.MLSEEqualizer` System object, and MLSE Equalizer block use the Viterbi algorithm to equalize a linearly modulated signal through a dispersive channel. These features output the maximum likelihood sequence estimate of the signal by using an estimate of the channel modeled as a finite input response (FIR) filter.

To decode a received signal, the MLSE equalizer:

- 1 Applies the FIR filter to the symbols in the input signal. The FIR filter tap weights correspond to the channel estimate.
- 2 Uses the Viterbi algorithm to compute the traceback paths and the state metric. These values are assigned to the symbols at each step of the Viterbi algorithm. The metrics are based on Euclidean distance.
- 3 Outputs the maximum likelihood sequence estimate of the signal as a sequence of complex numbers corresponding to the constellation points of the modulated signal.

An MLSE equalizer yields the best theoretically possible performance, but is computationally intensive.

For background material on MLSE equalizers, see “Selected References for Equalizers” on page 6-4.

Equalize a Vector Signal in MATLAB

You can use the `mlseeq` function or `comm.MLSEEqualizer` System object for MLSE equalization in MATLAB. The examples in this section call the `mlseeq` function. A similar workflow applies when using the `comm.MLSEEqualizer` System object. For examples that use the System object, see the `comm.MLSEEqualizer` System object reference page.

The `mlseeq` function has two operation modes:

- Continuous operation mode enables you to process a series of vectors by using repeated calls to `mlseeq`. The function saves its internal state information from one call to the next. To learn more, see [Equalizing in Continuous Operation Mode](#) on page 6-48.
- Reset operation mode enables you to specify a preamble and postamble that accompany your data. To learn more, see [Using a Preamble or Postamble](#) on page 6-52.

If you are not processing a series of vectors and do not need to specify a preamble or postamble, the operational modes are nearly identical. They differ in that continuous operation mode incurs a delay, while reset operation mode does not. The following example uses reset operation mode. If you modify the example to run using continuous operation mode, there will be delay in the equalized output. To learn more about this delay, see [Delays in Continuous Operation Mode](#) on page 6-49.

Use `mlseeq` to Equalize a Vector Signal

In its simplest form, the `mlseeq` function equalizes a vector of modulated data when you specify:

- The estimated coefficients of the channel (modeled as an FIR filter).
- The signal constellation for the modulation type.
- The Viterbi algorithm traceback depth. Larger values for the traceback depth can improve the results from the equalizer but increase the computation time.

Generate a PSK modulated signal with modulation order set to four.

```
M = 4;  
msg = pskmod([1 2 2 0 3 1 3 3 2 1 0 2 3 0 1]',M);
```

Filter the modulated signal through a distortion channel.

```
chcoeffs = [.986; .845; .237; .12345+.31i];  
filtmsg = filter(chcoeffs,1,msg);
```

Define the reference constellation, traceback length, and channel estimate for the MLSE equalizer. In this example, the exact channel is provided as the channel estimate.

```
const = pskmod([0:M-1],M);  
tblen = 10;  
chanest = chcoeffs;
```

Equalize the received signal.

```
msgEq = mlseeq(filtmsg,chanest,const,tblen,'rst');  
isequal(msg,msgEq)
```

```
ans = logical  
     1
```

Equalizing Signals in Continuous Operation Mode

If your data is partitioned into a series of vectors (that you process within a loop, for example), continuous operation mode is an appropriate way to use the `mlseeq` function. In continuous operation mode, `mlseeq` can save its internal state information for use in a subsequent invocation and can initialize by using previously stored state information. To choose continuous operation mode when invoking `mlseeq`, specify `'cont'` as an input argument.

Note Continuous operation mode incurs a delay, as described in *Delays in Continuous Operation Mode* on page 6-49. This mode cannot accommodate a preamble or postamble.

Procedure for Continuous Operation Mode

When using continuous operation mode within a loop, preallocate three empty matrix variables to store the state metrics, traceback states, and traceback inputs for the equalizer before the equalization loop starts. Inside the loop, invoke `mlseeq` using a syntax such as:

```
sm = [];  
ts = [];
```

```

ti = [];
for ...
    [y,sm,ts,ti] = mlseeq(x, chcoeffs, const, tblen, 'cont', nsamp, sm, ts, ti);
...
end

```

Using `sm`, `ts`, and `ti` as input arguments causes `mlseeq` to continue operating from where it finished in the previous iteration. Using `sm`, `ts`, and `ti` as output arguments causes `mlseeq` to update the state information at the end of the current iteration. In the first iteration, `sm`, `ts`, and `ti` start as empty matrices, so the first invocation of the `mlseeq` function initializes the metrics of all states to 0.

Delays in Continuous Operation Mode

Continuous operation mode with a traceback depth of `tblen` incurs an output delay of `tblen` symbols. The first `tblen` output symbols are unrelated to the input signal and the last `tblen` input symbols are unrelated to the output signal. For example, this command uses a traceback depth of 3. The first three output symbols are unrelated to the input signal of ones (1, 10).

```
y = mlseeq(ones(1,10), 1, [-7:2:7], 3, 'cont')
```

```

y =
    -7    -7    -7     1     1     1     1     1     1     1

```

It is important to keep track of delays introduced by different portions of the communications system. The “Use `mlseeq` to Equalize a Vector in Continuous Operation Mode” on page 6-49 example illustrates how to account for the delay when computing an error rate.

Use `mlseeq` to Equalize a Vector in Continuous Operation Mode

This example shows the procedure for using the continuous operation mode of the `mlseeq` function within a loop.

Initialize Variables

Specify run-time variables.

```

numsym = 200; % Number of symbols in each iteration
numiter = 25; % Number of iterations

```

```

M = 4; % Use 4-PSK modulation
qpskMod = comm.QPSKModulator('PhaseOffset', 0);

```

```
chcoeffs = [1 ; 0.25]; % Channel coefficients
chanest = chcoeffs; % Channel estimate
```

To initialize the equalizer, define parameters for the reference constellation, traceback length, number of samples per symbol, and the state variables `sm`, `ts`, and `ti`.

```
const = qpskMod((0:M-1)');
tblen = 10;
nsamp = 1;
sm = [];
ts = [];
ti = [];
```

Define variables to accumulate results from each iteration of the loop.

```
fullmodmsg = [];
fullfiltmsg = [];
fullrx = [];
```

Simulate the System by Using a Loop

Run the simulation in a loop that generates random data, modulates the data by using baseband PSK modulation, and filters the data. The `mlseeq` function equalizes the filtered data. The loop also updates the variables that accumulate results from each iteration of the loop.

```
for jj = 1:numiter
    msg = randi([0 M-1],numsym,1); % Random signal vector
    modmsg = qpskMod(msg); % PSK-modulated signal
    filtmsg = filter(chcoeffs,1,modmsg); % Filtered signal
    % Equalize the signal.
    [rx,sm,ts,ti] = mlseeq(filtmsg,chanest,const, ...
        tblen,'cont',nsamp,sm,ts,ti);
    % Update vectors with cumulative results.
    fullmodmsg = [fullmodmsg; modmsg];
    fullfiltmsg = [fullfiltmsg; filtmsg];
    fullrx = [fullrx; rx];
end
```

Computing an Error Rate and Plotting Results

Compute the symbol error rate from all iterations of the loop. The `symerr` function compares selected portions of the received and transmitted signals, not the entire signals. Because continuous operation mode incurs a delay whose length in samples is the

traceback depth (`tblen`) of the equalizer, exclude the first `tblen` samples from the received signal and the last `tblen` samples from the transmitted signal. Excluding samples that represent the delay of the equalizer ensures that the symbol error rate calculation compares samples from the received and transmitted signals that are meaningful and that truly correspond to each other.

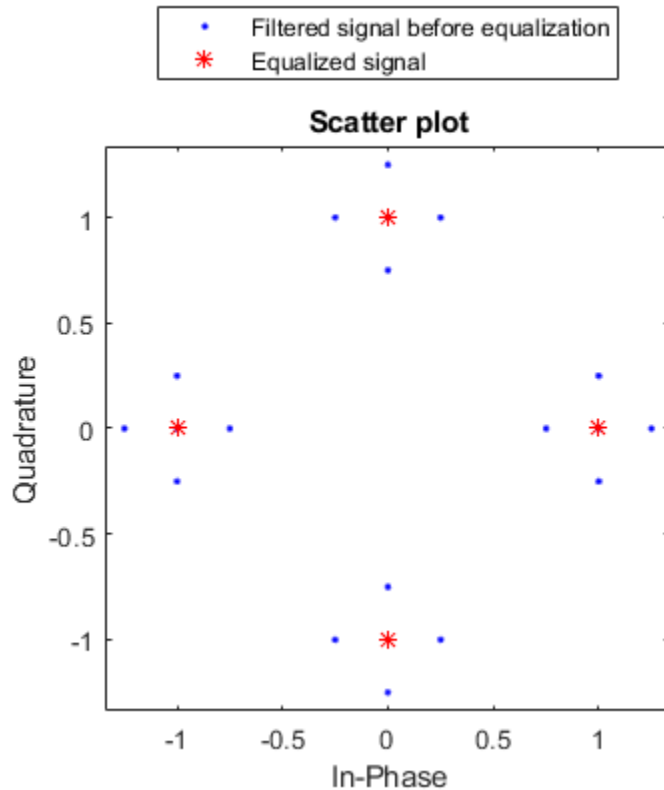
Taking the delay into account, compute the total number of symbol errors.

```
hErrorCalc = comm.ErrorRate('ReceiveDelay',10);  
err = step(hErrorCalc, fullmodmsg, fullrx);  
numsymerrs = err(1)
```

```
numsymerrs = 0
```

Plot the signal constellation before and after equalization. The points in the equalized signal coincide with the points of the ideal signal constellation for 4-PSK.

```
h = scatterplot(fullfiltmsg);  
hold on;  
scatterplot(fullrx,1,0,'r*',h);  
legend('Filtered signal before equalization','Equalized signal',...  
       'Location','NorthOutside');  
hold off;
```



Use a Preamble or a Postamble

Some systems include a sequence of known symbols at the beginning or end of a set of data. The known sequence at the beginning or end is called a *preamble* or *postamble*, respectively. The `mlseq` function can accommodate a preamble and postamble that are already incorporated into its input signal. When you invoke the function, you specify the preamble and postamble as integer vectors that represent the sequence of known symbols by indexing into the signal constellation vector. For example, a preamble vector of `[1 4 4]` and a 4-PSK signal constellation of `[1 j -1 -j]` indicates that the modulated signal begins with `[1 -j -j]`.

If your system uses a preamble without a postamble, use a postamble vector of `[]` when invoking `mlseq`. If your system uses a postamble without a preamble, use a preamble vector of `[]`.

Recover Message Containing Preamble

Recover a message that includes a preamble, equalize the signal, and check the symbol error rate.

Specify the modulation order, equalizer traceback depth, number of samples per symbol, preamble, and message length.

```
M = 4;
tblen = 16;
nsamp = 1;
preamble = [3;1];
msgLen = 500;
```

Generate the reference constellation.

```
const = pskmod(0:3,4);
```

Generate a message by using random data and prepend the preamble to the message. Modulate the random data.

```
msgData = randi([0 M-1],msgLen,1);
msgData = [preamble; msgData];
msgSym = pskmod(msgData,M);
```

Filter the data through a distortion channel and add Gaussian noise to the signal.

```
chcoeffs = [0.623; 0.489+0.234i; 0.398i; 0.21];
chanest = chcoeffs;
msgFilt = filter(chcoeffs,1,msgSym);
msgRx = awgn(msgFilt,9,'measured');
```

Equalize the received signal. To configure the equalizer, provide the channel estimate, reference constellation, equalizer traceback depth, operating mode, number of samples per symbol, and preamble. The same preamble symbols appear at the beginning of the message vector and in the syntax for `mlseq`. Because the system does not use a postamble, an empty vector is specified as the last input argument in this `mlseq` syntax.

Check the symbol error rate of the equalized signal. Run-to-run results vary due to use of random numbers.

```
eqSym = mlseeq(msgRx,chanest,const,tblen,'rst',nsamp,preamble,[]);  
[nsymerrs,ser] = symerr(msgSym,eqSym)  
  
nsymerrs = 8  
ser = 0.0159
```

Using MLSE Equalizers in Simulink

The MLSE Equalizer block uses the Viterbi algorithm to equalize a linearly modulated signal through a dispersive channel. The block outputs the maximum likelihood sequence estimate (MLSE) of the signal by using your estimate of the channel modeled as a finite input response (FIR) filter. When using the MLSE Equalizer block, you specify the channel estimate and the signal constellation of the input signal. You can also specify an expected input signal preamble and postamble as input parameters to the MLSE Equalizer block.

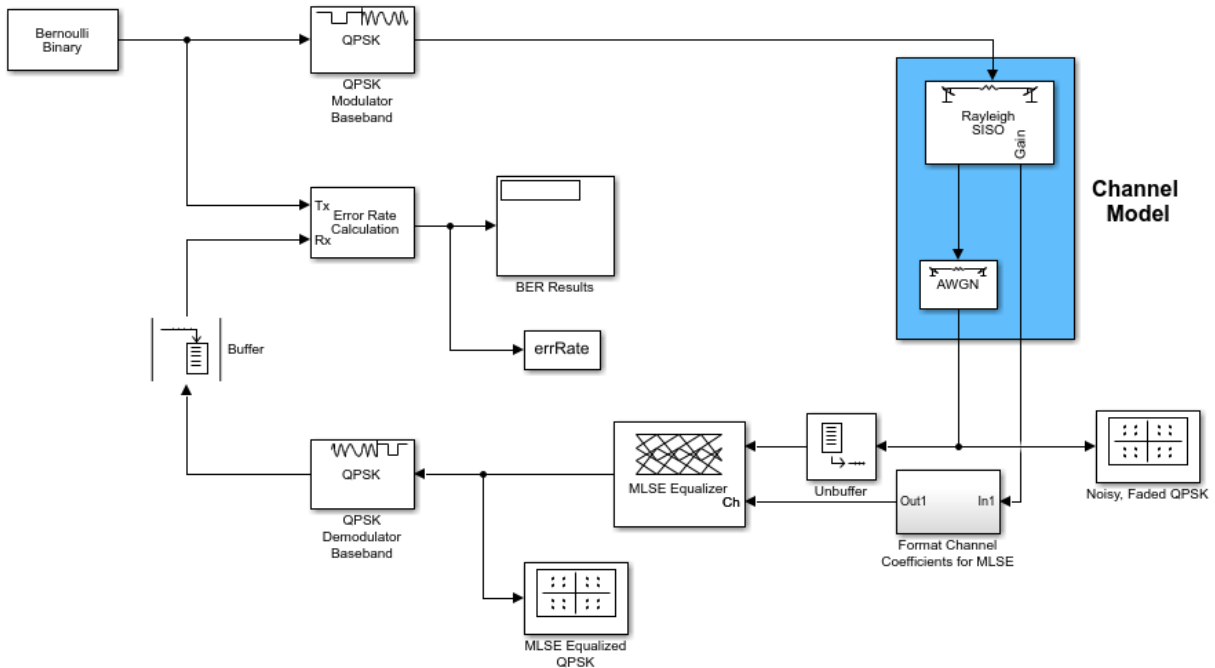
MLSE Equalization with Dynamically Changing Channel

Use a Maximum Likelihood Sequence Estimation (MLSE) equalizer to equalize the effects of a multipath Rayleigh fading channel. The MLSE equalizer inputs data that has passed through a time varying dispersive channel and an estimate of the channel. The channel estimate contains dynamically evolving channel coefficients of a two-path Rayleigh fading channel.

Model Structure

- The transmitter generates QPSK random signal data.
- Channel impairments include multipath fading and AWGN.
- The receiver applies MLSE equalization and QPSK demodulation.
- The model uses scopes and a BER calculation to show the system behavior.

Explore Example Model



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Experimenting with the model

The Bernoulli Binary Generator block sample time of 5e-6 seconds corresponds to a bit rate of 200 kbps and a QPSK symbol rate of 100 ksym/sec.

The Multipath Rayleigh Fading Channel block settings are:

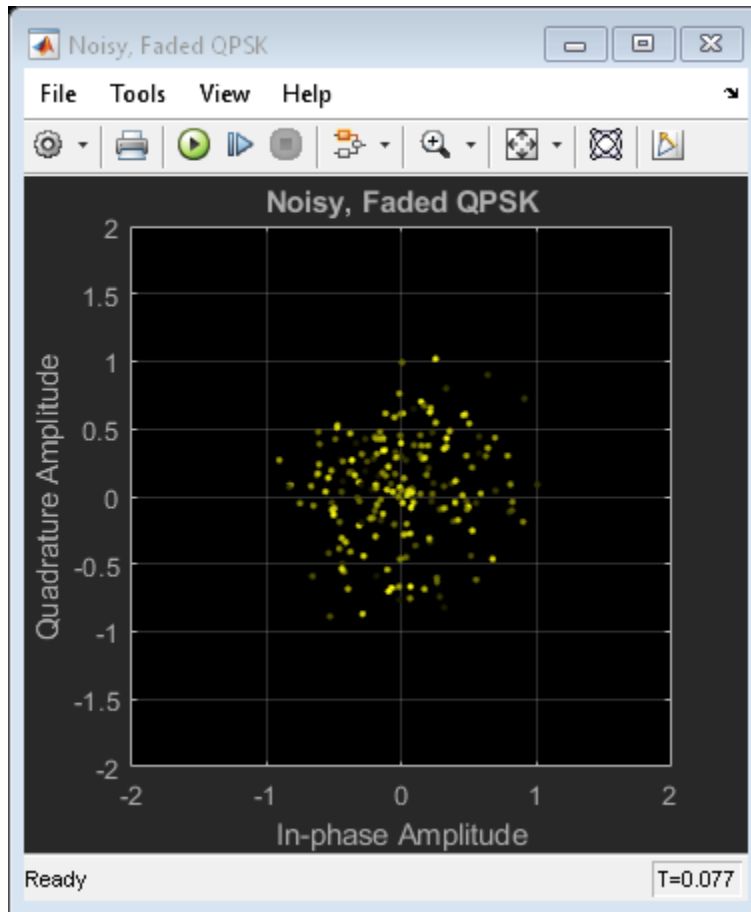
- Maximum Doppler shift is 30 Hz.
- Discrete path delay is [0 1e-5], which corresponds to two consecutive sample times of the input QPSK symbol data. This delay reflects the simplest delay vector for a two-path channel.
- Average path gain is [0 -10].
- Average path gains are normalized to 0 dB so that the average power input to the AWGN block is 1 W.

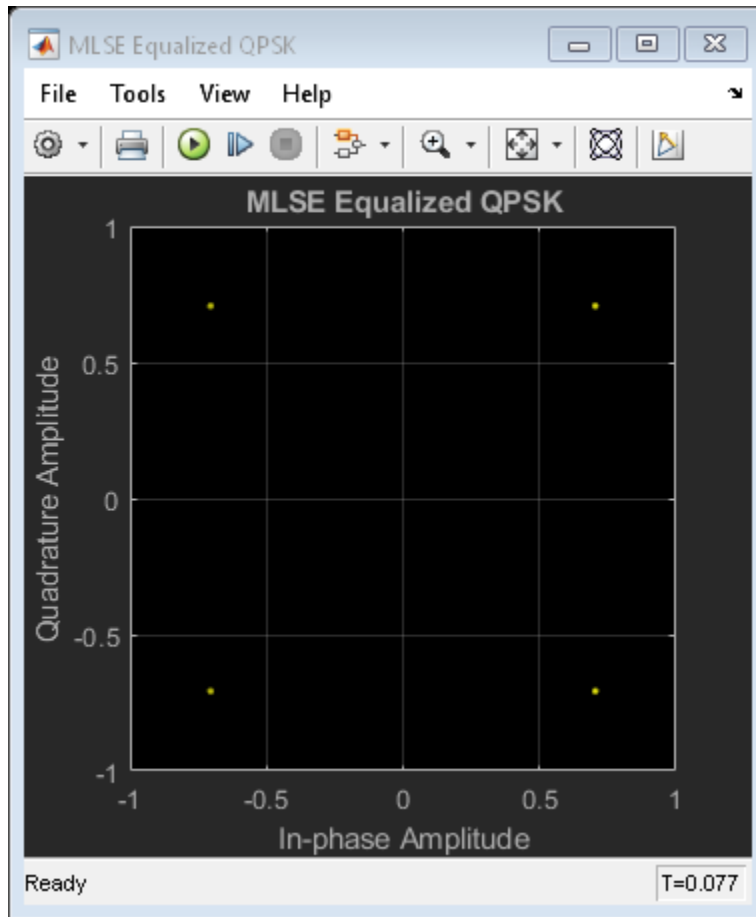
The MLSE Equalizer block has the Traceback depth set to 10. Vary this depth to study its effect on Bit Error rate (BER).

The QPSK demodulator accepts an N-by-1 input frame and generates a 2N-by-1 output frame. This output frame and a traceback depth of 10 results in a delay of 20 bits. The model performs frame-based processing on frames that have 100 samples per frame. Due to the frame-based processing, there is an inherent delay of 100 bits in the model. The combined receive delay of 120 is set in the Receive delay parameter of the Error Rate Calculation block, aligning the samples.

The computed BER is displayed. Constellation plots show the constellation before and after equalization.

BER = 0.033645





See Also

Objects

`comm.DecisionFeedbackEqualizer` | `comm.LinearEqualizer` | `comm.MLSEEqualizer`

Blocks

Decision Feedback Equalizer | Linear Equalizer | MLSE Equalizer

More About

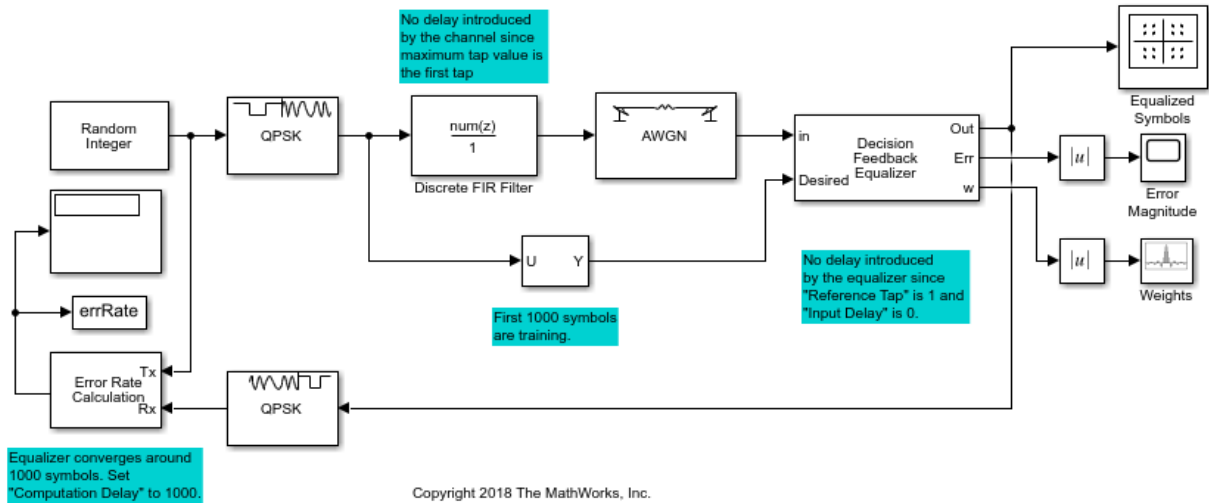
- “Equalization” on page 6-2
- “Adaptive Equalizers” on page 6-6

Equalizer Examples (new & old)

- “DF Equalize QPSK-Modulated Signal in Simulink” on page 7-2
- “Linearly Equalize QPSK-Modulated Signal in Simulink” on page 7-6
- “Adaptive Equalization with Filtering and Fading Channel” on page 7-10
- “MLSE Equalization with Dynamically Changing Channel” on page 7-18
- “Adaptive Equalization” on page 7-22
- “Equalize BSPK Signal” on page 7-32
- “Compare RLS and LMS Algorithms” on page 7-36

DF Equalize QPSK-Modulated Signal in Simulink

Apply decision feedback equalization using the least mean squares (LMS) algorithm to recover QPSK symbols passed through an AWGN channel.



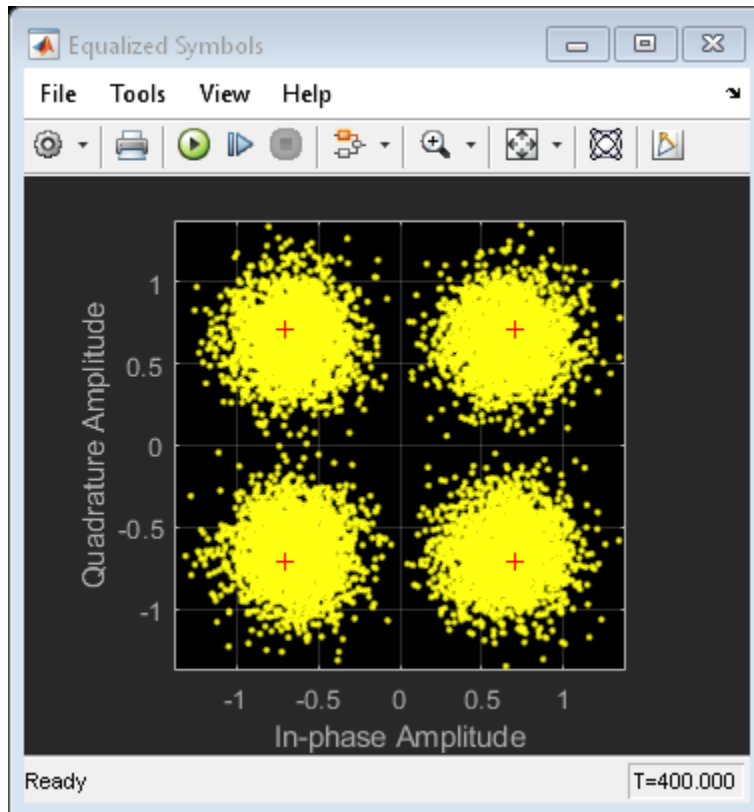
Copyright 2018 The MathWorks, Inc.

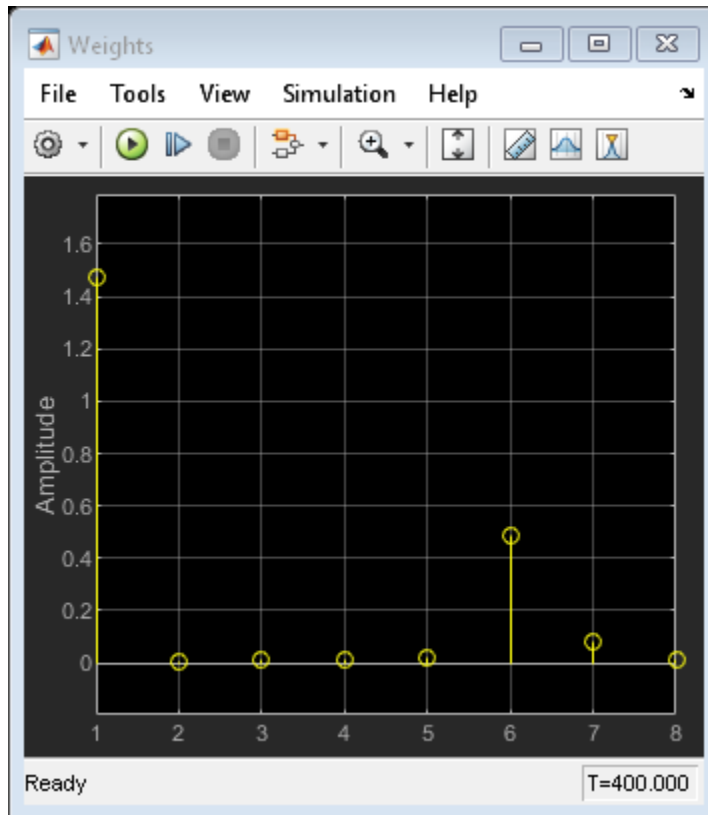
The `slex_dfreq_qpsk_signal` model generates an $M=4$ sequence using the Random Integer Generator block. The sequence is modulated with the M-PSK Modulator Baseband block, filtered with the Discrete FIR Filter block, and then impaired with the AWGN block. The Decision Feedback Equalizer block equalizes the data sequence, the data is demodulated with the M-PSK Demodulator Baseband block, and the bit error rate is computed. The signal path out of the modulator is split to a Selector block, which provides the first 1000 symbols of the modulated signal to the equalizer as an initial training sequence.

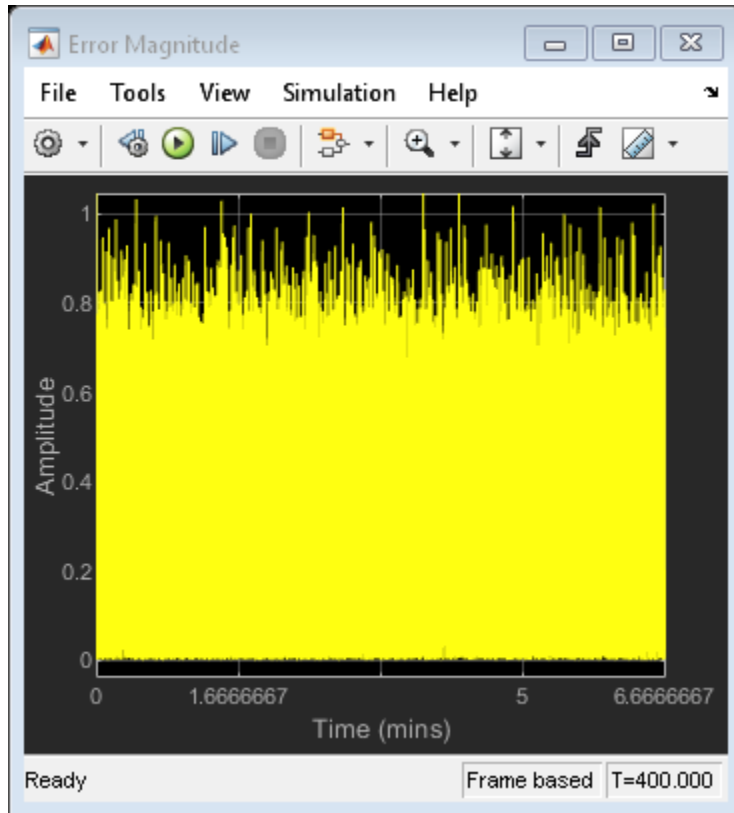
No delay is introduced between the transmitted and received signal because the maximum tap value is the first tap of the discrete FIR filter and the equalizer reference tap is 1. The equalizer converges after around 1000 symbols so this value is used for the computation delay of the Error Rate Calculation block.

The computed error rate is displayed and plots show the equalized constellation, equalized tap weights, and signal error magnitude.

Computed error rate = 0.0011111

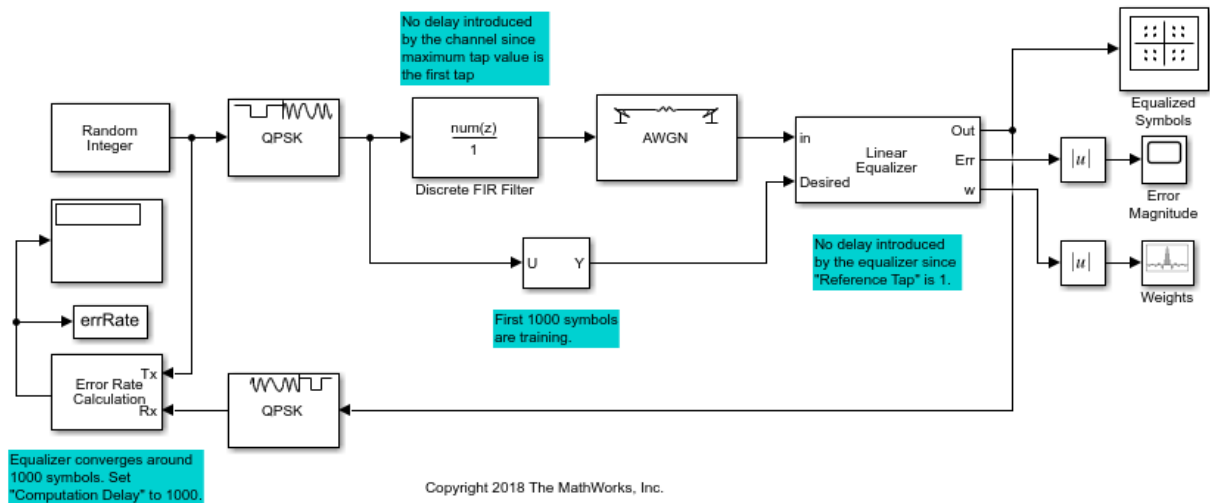






Linearly Equalize QPSK-Modulated Signal in Simulink

Apply linear equalization using the least mean squares (LMS) algorithm to recover QPSK symbols passed through an AWGN channel.

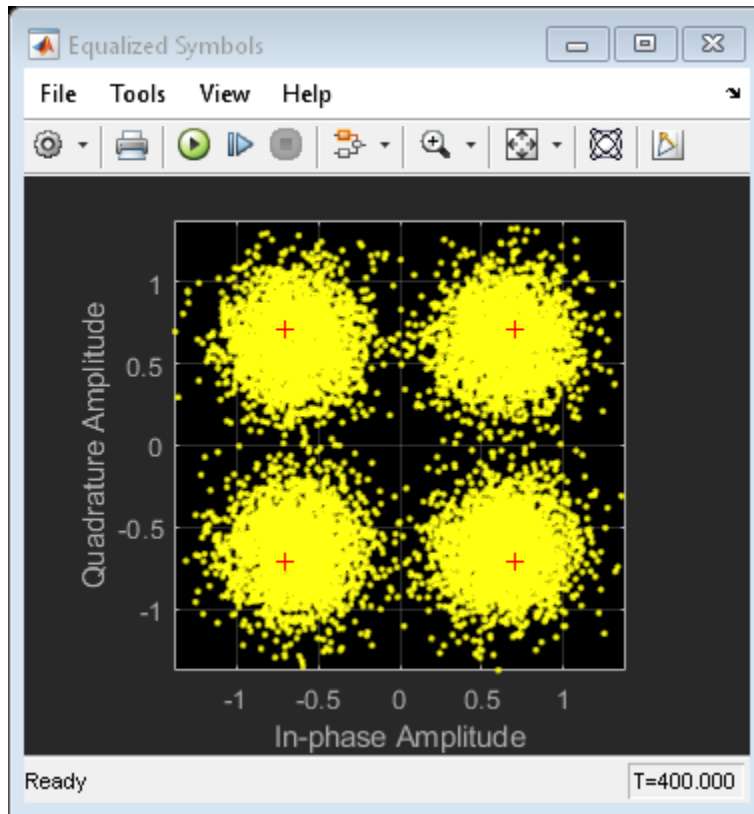


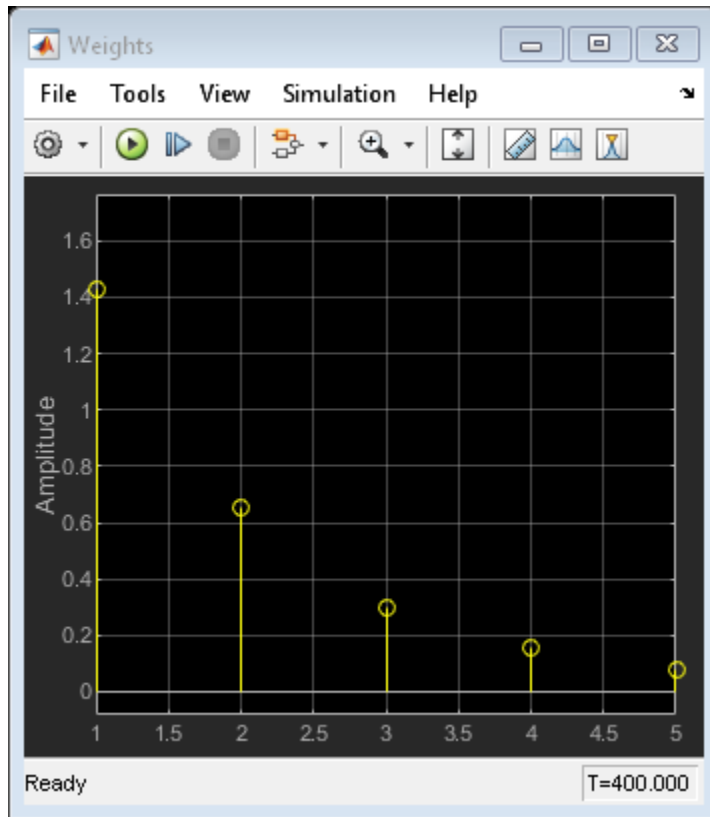
The `slex_lineq_qpsk_signal` model generates an $M=4$ sequence using the Random Integer Generator block. The sequence is modulated with the M-PSK Modulator Baseband block, filtered with the Discrete FIR Filter block, and then impaired with the AWGN block. The Linear Equalizer block equalizes the data sequence, the data is demodulated with the M-PSK Demodulator Baseband block, and the bit error rate is computed. The signal path out of the modulator is split to a Selector block, which provides the first 1000 symbols of the modulated signal to the equalizer as an initial training sequence.

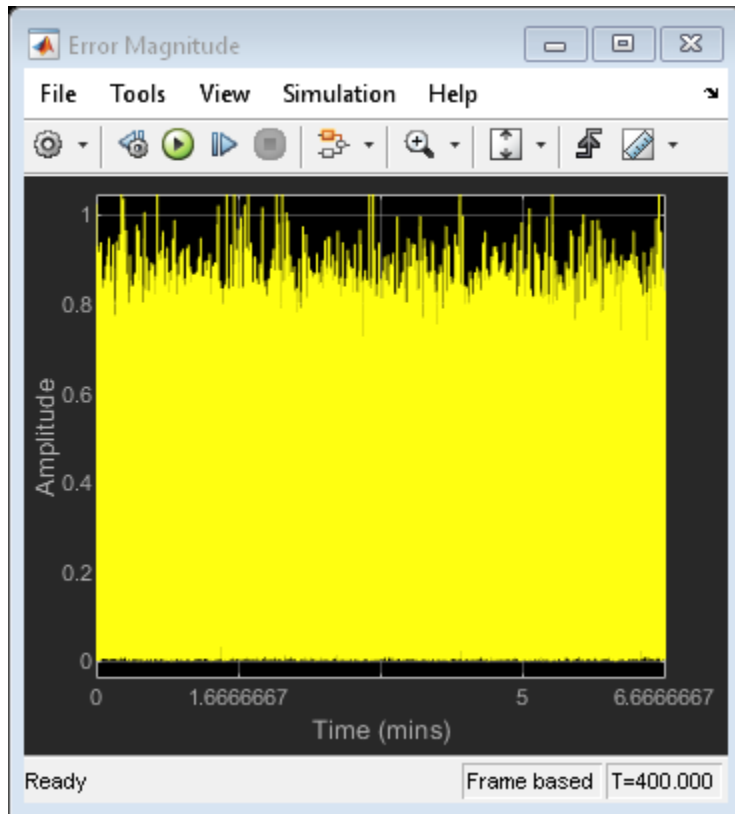
No delay is introduced between the transmitted and received signal because the maximum tap value is the first tap of the discrete FIR filter and the equalizer reference tap is 1. The equalizer converges after around 1000 symbols so this value is used for the computation delay of the Error Rate Calculation block.

The computed error rate is displayed and plots show the equalized constellation, equalized tap weights, and signal error magnitude.

Computed error rate = 0.0024444







Adaptive Equalization with Filtering and Fading Channel

This model shows the behavior of the selected adaptive equalizer in a communication link that has a fading channel. The transmitter and receiver have root raised cosine pulse shaped filtering. A subsystem block enables you to select between linear or decision feedback equalizers that use the least mean square (LMS) or recursive least square (RLS) adaptive algorithm.

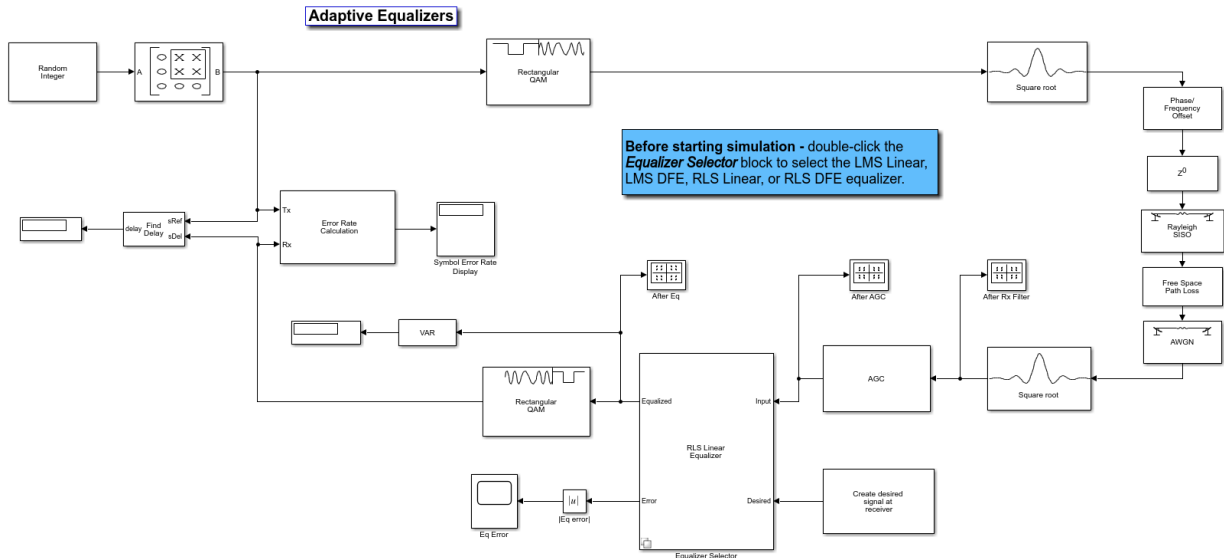
Model Structure

- The transmitter generates 16QAM random signal data that includes a training sequence and applies root raised cosine pulse shaped filtering.
- Channel impairments include multipath fading, Doppler shift, carrier frequency offset, variable integer delay, free space path loss, and AWGN.
- The receiver applies root raised cosine pulse shaped filtering, AGC, includes equalizer mode control to enable training and enables you to select the equalizer algorithm from these choices.

Selection	Equalizer Algorithm
LMS Linear	Linear least mean square equalizer
LMS DFE	Decision feedback least mean square equalizer
RLS Linear	Linear recursive least square equalizer
RLS DFE	Decision feedback recursive least square equalizer

- Scopes help you understand how the different equalizers and adaptive algorithms behave.

Explore Example Model



Experimenting with the model

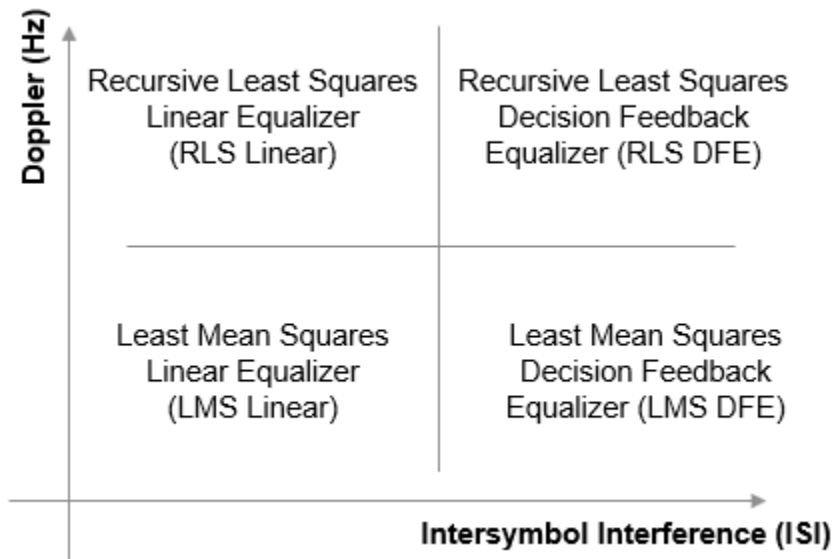
This model provides several ways for you to change settings and observe the results. The `InitFcn` found in `File>Model Properties>Callbacks` calls `cm_ex_adaptive_eq_with_fading_init` to initialize the model. This file enables you to vary settings in the model, including:

- System parameters, such as SNR.
- Pulse shaping filter parameters, such as rolloff and filter length
- Path loss value.
- Channel conditions: Rayleigh or Rician fading, channel path gains, channel path delays, and Doppler shift.
- Equalizer choice and configuration.

Model Considerations

This non-standards-based communication link is representative of a modern communications system.

- The optimal equalizer configuration depends on the channel conditions. The initialization file sets the Doppler shift and multipath fading channel parameters that highlight the capabilities of different equalizers.



- The decision feedback equalizer structure performs better than the linear equalizer structure for higher intersymbol interference.
- The RLS algorithm performs better than the LMS algorithm for higher Doppler frequencies.
- The LMS algorithm executes quickly, converges slowly, and its complexity grows linearly with the number of weights.
- The RLS algorithm converges quickly, its complexity grows approximately as the square of the number of weights. It can be unstable when the number of weights is large.
- The channels exercised for different equalizers have the following characteristics.

Selection	Channel Characteristics
LMS Linear	3 tap multipath fading channel with 10 Hz Doppler shift
LMS DFE	5 tap multipath fading channel with 25 Hz Doppler shift
RLS Linear	3 tap multipath fading channel with 50 Hz Doppler shift
RLS DFE	5 tap multipath fading channel with 100 Hz Doppler shift

- Initial settings for other channel impairments are the same for all equalizers. Carrier frequency offset value is set to 50 Hz. Free space path loss is set to 60 dB. Variable integer delay is set to 2 samples, which requires the equalizers to perform some timing recovery.

Deep channel fades and path loss can cause the equalizer input signal level to be much less than the desired output signal level and result in unacceptably long equalizer convergence time. The AGC block adjusts the magnitude of received signal to reduce the equalizer convergence time. You must adjust the optimal AGC output power level based on the modulation scheme selected. For 16QAM, a desired output power of 10 W is used.

Training of the equalizer is performed at the beginning of the simulation.

Running the Simulation

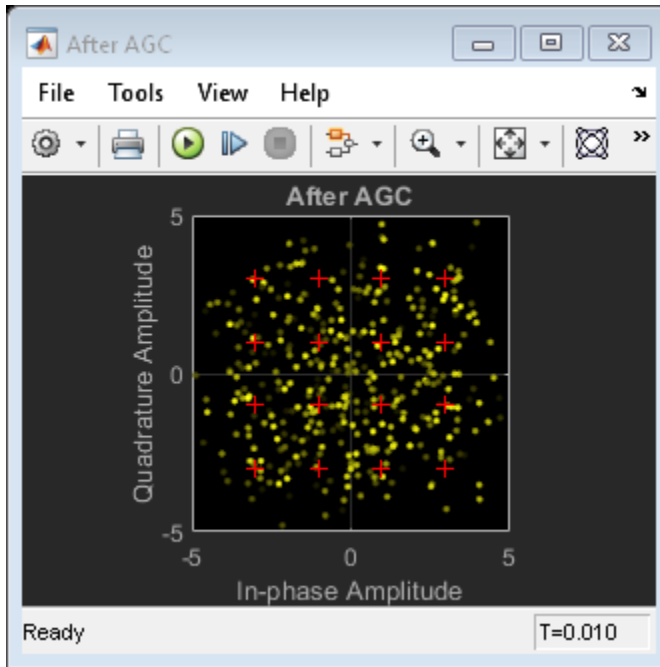
Running the simulation computes symbol error statistics and produces these figures:

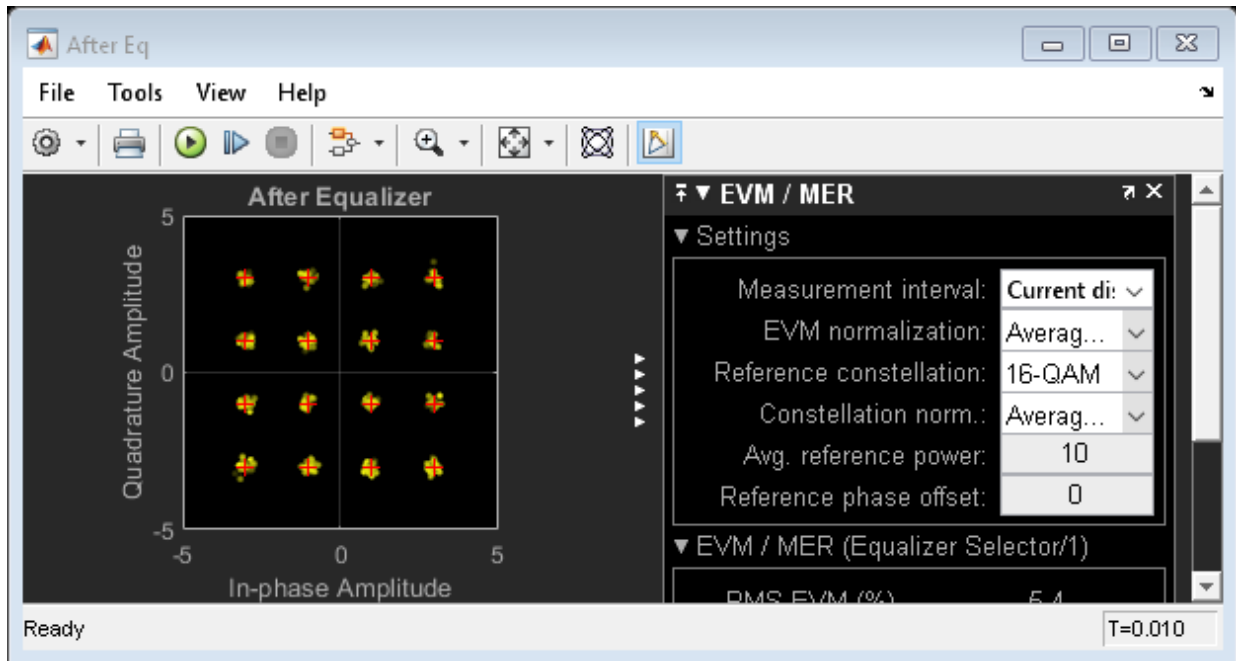
- A constellation diagram of the signal after the receive filter.
- A constellation diagram of the signal after AGC.
- A constellation diagram of the signal after equalization with signal quality measurements shown.
- An equalizer error plot.

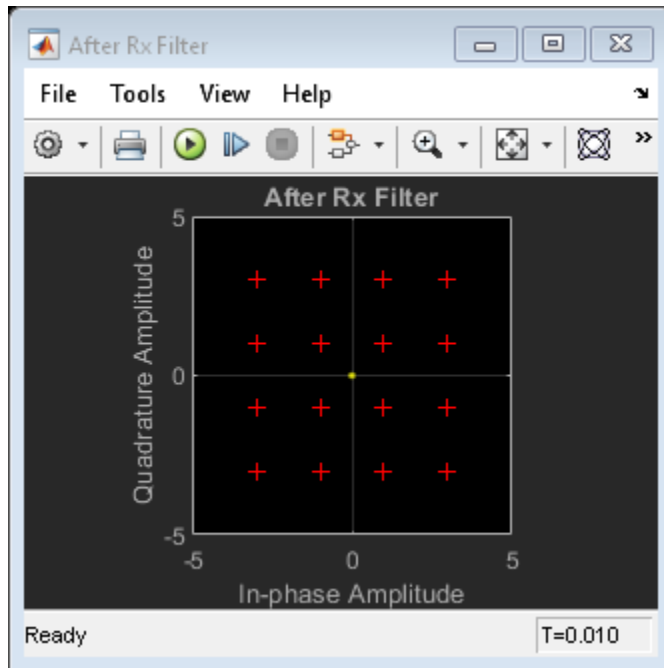
For the plots shown here, the equalizer algorithm selected is RLS Linear. Monitoring these figures, you can see that the received signal quality fluctuates as simulation time progresses.

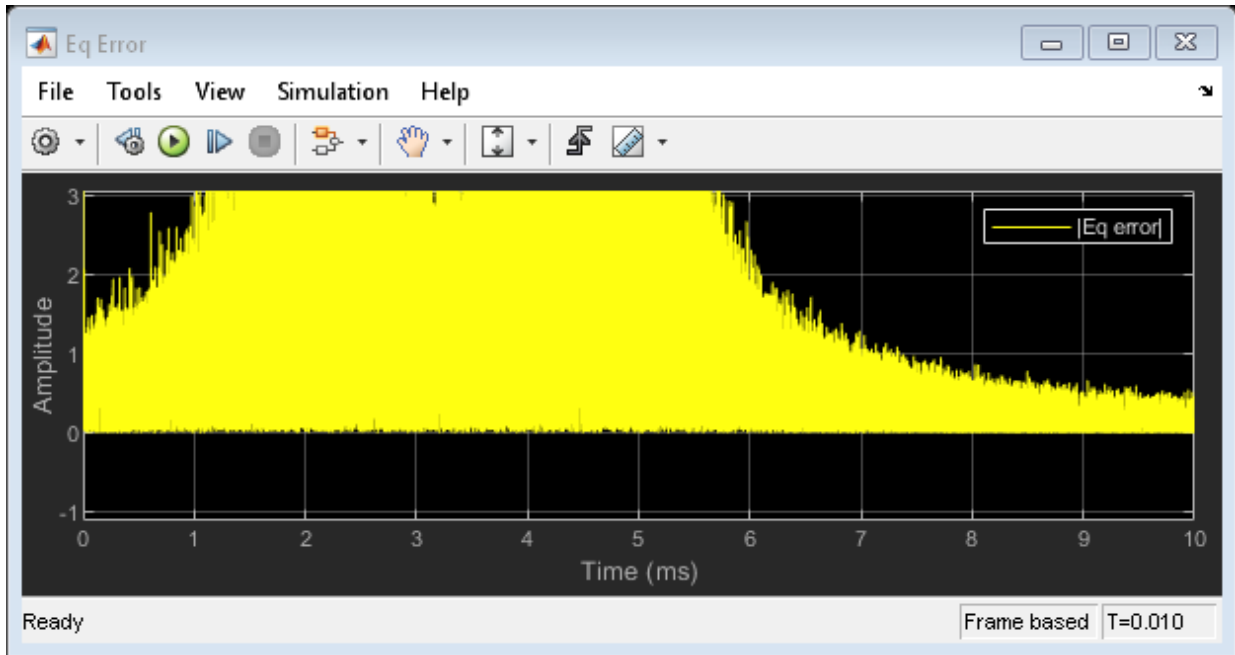
The *After Rx Filter* and *After AGC* constellation plots show the signal before equalization. *After AGC* shows the impact of the channel conditions on the transmitted signal. The *After Eq* plot shows the signal after equalization. The signal plotted in the constellation diagram after equalization shows the variation in signal quality based on the effectiveness of the equalization process. Throughout the simulation, the signal constellations plotted before equalization deviate noticeably from a 16QAM signal constellation. The *After Eq*

constellation improves or degrades as the equalizer error signal varies. The *Eq error* plotted in the *Eq Error* plot, indicates poor equalization at the start of the simulation. The error degrades at first then improves as the equalizer converges.









Further Exploration

Double-click the Equalizer Selector block and select a different equalizer. Run the simulation to see the performance of the various equalizer options. You can use the signal logger to compare the results from this experimentation. In the block diagram, right-click on signal wires and select Log Selected Signals. If you have enabled signal logging, after the simulation run finishes, open the Simulation Data Inspector to view the logged signals.

At the MATLAB™ command prompt, enter `edit cm_ex_adaptive_eq_with_fading_init.m` to open the initialization file, then modify a parameter and rerun the simulation. For example, adjust the channel characteristics (`params.maxDoppler`, `params.pathDelays`, and `params.pathGains`). The RLS adaptive algorithm performs better than the LMS adaptive algorithm as the maximum Doppler is increased.

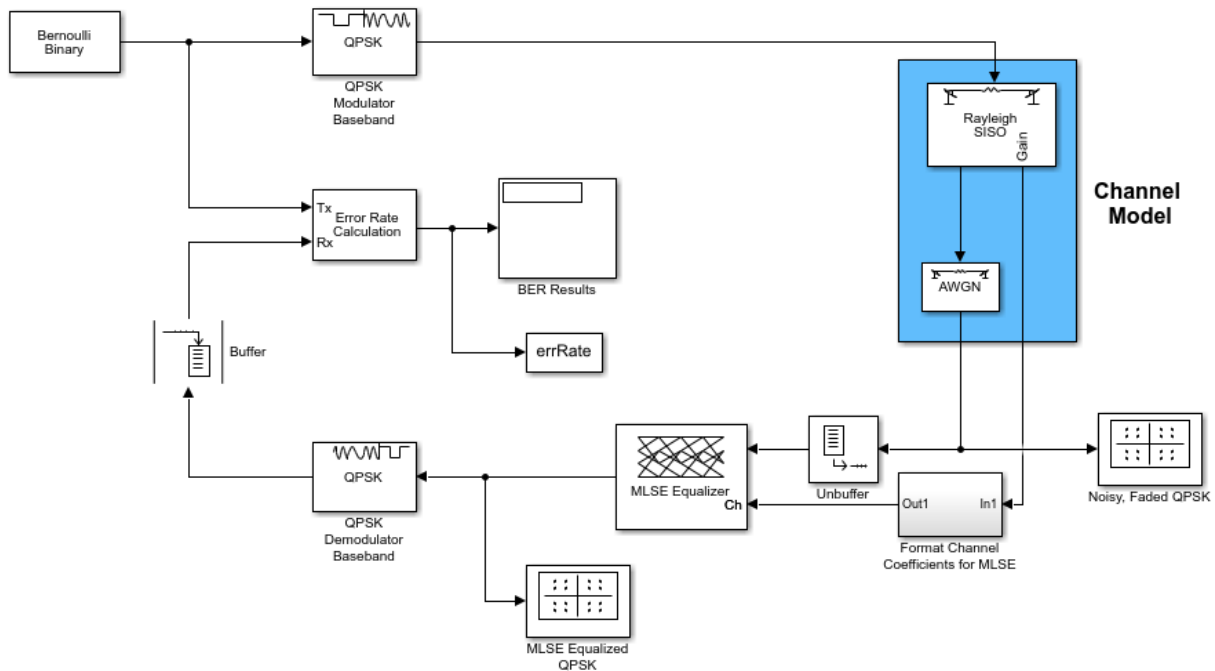
MLSE Equalization with Dynamically Changing Channel

Use a Maximum Likelihood Sequence Estimation (MLSE) equalizer to equalize the effects of a multipath Rayleigh fading channel. The MLSE equalizer inputs data that has passed through a time varying dispersive channel and an estimate of the channel. The channel estimate contains dynamically evolving channel coefficients of a two-path Rayleigh fading channel.

Model Structure

- The transmitter generates QPSK random signal data.
- Channel impairments include multipath fading and AWGN.
- The receiver applies MLSE equalization and QPSK demodulation.
- The model uses scopes and a BER calculation to show the system behavior.

Explore Example Model



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Experimenting with the model

The Bernoulli Binary Generator block sample time of 5e-6 seconds corresponds to a bit rate of 200 kbps and a QPSK symbol rate of 100 ksym/sec.

The Multipath Rayleigh Fading Channel block settings are:

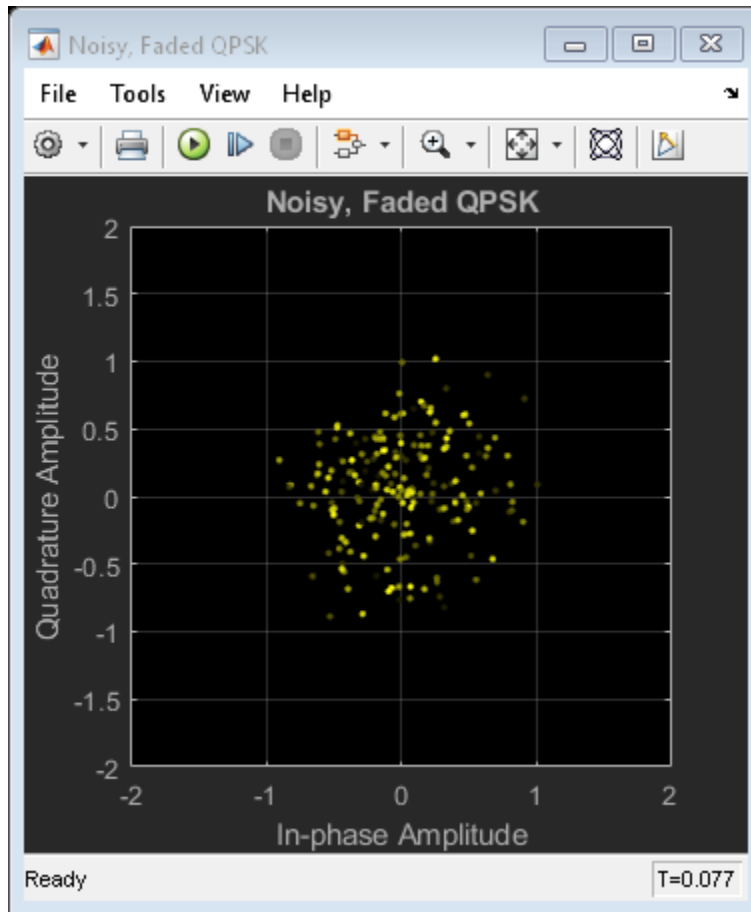
- Maximum Doppler shift is 30 Hz.
- Discrete path delay is [0 1e-5], which corresponds to two consecutive sample times of the input QPSK symbol data. This delay reflects the simplest delay vector for a two-path channel.
- Average path gain is [0 -10].
- Average path gains are normalized to 0 dB so that the average power input to the AWGN block is 1 W.

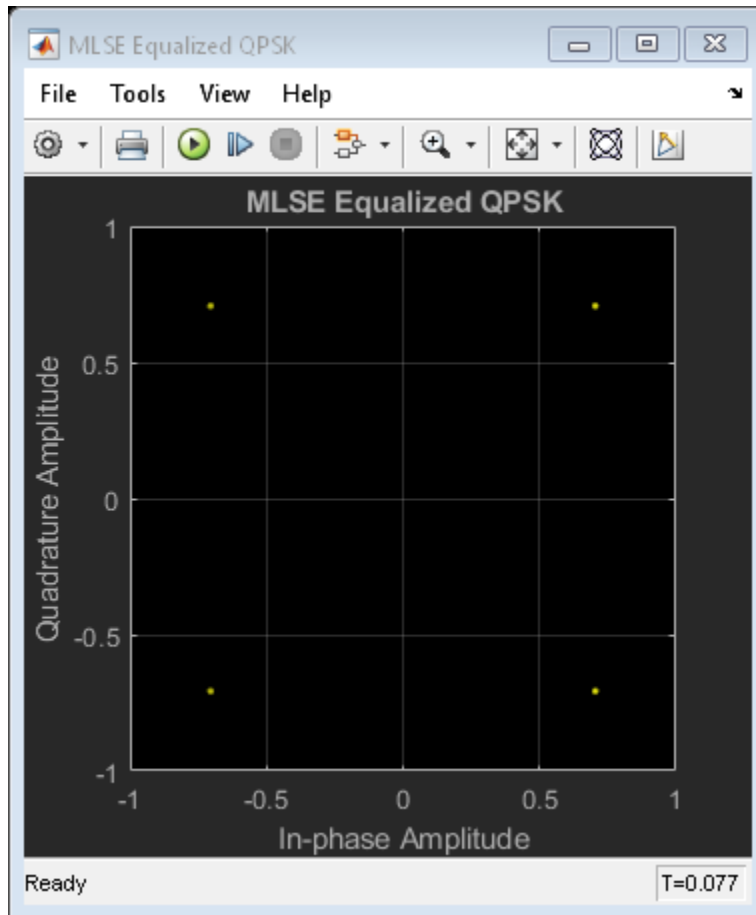
The MLSE Equalizer block has the Traceback depth set to 10. Vary this depth to study its effect on Bit Error rate (BER).

The QPSK demodulator accepts an N-by-1 input frame and generates a 2N-by-1 output frame. This output frame and a traceback depth of 10 results in a delay of 20 bits. The model performs frame-based processing on frames that have 100 samples per frame. Due to the frame-based processing, there is an inherent delay of 100 bits in the model. The combined receive delay of 120 is set in the Receive delay parameter of the Error Rate Calculation block, aligning the samples.

The computed BER is displayed. Constellation plots show the constellation before and after equalization.

BER = 0.033645





Adaptive Equalization

This example shows how to model a communication link with PSK modulation, raised cosine pulse shaping, multipath fading, and adaptive equalization.

The example sets up three equalization scenarios, and calls a separate script to execute the processing loop multiple times for each scenario. Each call corresponds to a transmission block. The pulse shaping and multipath fading channel retain state information from one block to the next. For visualizing the impact of channel fading on adaptive equalizer convergence, the simulation resets the equalizer state every block.

To experiment with different simulation settings, you can edit the example. For instance, you can set the `ResetBeforeFiltering` property of the equalizer object to 0, which will cause the equalizer to retain state from one block to the next.

Transmission Block

Set parameters related to the transmission block which is composed of three parts: training sequence, payload, and tail sequence. All three use the same PSK scheme; the training and tail sequences are used for equalization. We use the default random number generator to ensure the repeatability of the results.

```
Rsym      = 1e6; % Symbol rate (Hz)
nTrain    = 100; % Number of training symbols
nPayload  = 400; % Number of payload symbols
nTail     = 20;  % Number of tail symbols
% Set random number generator for repeatability
hStream   = RandStream.create('mt19937ar', 'seed', 12345);
```

PSK Modulation

Configure the PSK modulation and demodulation System objects.

```
bitsPerSym = 2; % Number of bits per PSK symbol
M = 2^bitsPerSym; % Modulation order
hPSKMod = comm.PSKModulator(M, ...
    'PhaseOffset',0, ...
    'SymbolMapping','Binary');
hPSKDemod = comm.PSKDemodulator(M, ...
    'PhaseOffset',0, ...
    'SymbolMapping','Binary');

PSKConstellation = constellation(hPSKMod).'; % PSK constellation
```

Training and Tail Sequences

Generate the training and tail sequences.

```
xTrainData = randi(hStream, [0 M-1], nTrain, 1);
xTailData  = randi(hStream, [0 M-1], nTail, 1);
xTrain     = step(hPSKMod,xTrainData);
xTail      = step(hPSKMod,xTailData);
```

Transmit and Receive Filters

Configure raised cosine transmit and receive filter System objects. The filters incorporate upsampling and downsampling, respectively.

```
chanFilterSpan = 8; % Filter span in symbols
sampPerSymChan = 4; % Samples per symbol through channels
hTxFilt = comm.RaisedCosineTransmitFilter( ...
    'RolloffFactor',0.25, ...
    'FilterSpanInSymbols',chanFilterSpan, ...
    'OutputSamplesPerSymbol',sampPerSymChan);

hRxFilt = comm.RaisedCosineReceiveFilter( ...
    'RolloffFactor',0.25, ...
    'FilterSpanInSymbols',chanFilterSpan, ...
    'InputSamplesPerSymbol',sampPerSymChan, ...
    'DecimationFactor',sampPerSymChan);

% Calculate the samples per symbol after the receive filter
sampPerSymPostRx = sampPerSymChan/hRxFilt.DecimationFactor;
% Calculate the delay in samples from both channel filters
chanFilterDelay = chanFilterSpan*sampPerSymPostRx;
```

AWGN Channel

Configure an AWGN channel System object with the NoiseMethod property set to Signal to noise ratio (Es/No) and Es/No set to 20 dB.

```
hAWGNChan = comm.AWGNChannel( ...
    'NoiseMethod','Signal to noise ratio (Es/No)', ...
    'EsNo',20, ...
    'SamplesPerSymbol',sampPerSymChan);
```

Simulation 1: Linear Equalization for Frequency-Flat Fading

Begin with single-path, frequency-flat fading channel. For this channel, the receiver uses a simple 1-tap LMS (least mean square) equalizer, which implements automatic gain and phase control.

The script `commadapteqloop.m` runs multiple times. Each run corresponds to a transmission block. The equalizer resets its state and weight every transmission block. To retain state from one block to the next, you can set the `ResetBeforeFiltering` property of the equalizer object to `false`.

Before the first run, `commadapteqloop.m` displays the Rayleigh channel System object and the properties of the equalizer object. For each run, a MATLAB figure shows signal processing visualizations. The red circles in the signal constellation plots correspond to symbol errors. In the "Weights" plot, blue and magenta lines correspond to real and imaginary parts, respectively.

```
simName = 'Linear equalization for frequency-flat fading'; % Used to label figure window

% Configure a frequency-flat Rayleigh channel System object with the
% RandomStream property set to 'mt19937ar with seed' for repeatability.
hRayleighChan = comm.RayleighChannel( ...
    'SampleRate',Rsym*sampPerSymChan, ...
    'MaximumDopplerShift',30);

% Configure an adaptive equalizer object
nWeights = 1; % Single weight
stepSize = 0.1; % Step size for LMS algorithm
alg = lms(stepSize); % Adaptive algorithm object
eqObj = lineareq(nWeights,alg,PSKConstellation); % Equalizer object
% Delay in symbols from the equalizer
eqDelayInSym = (eqObj.RefTap-1)/sampPerSymPostRx;

% Link simulation
nBlocks = 50; % Number of transmission blocks in simulation
for block = 1:nBlocks
    commadapteqloop;
end

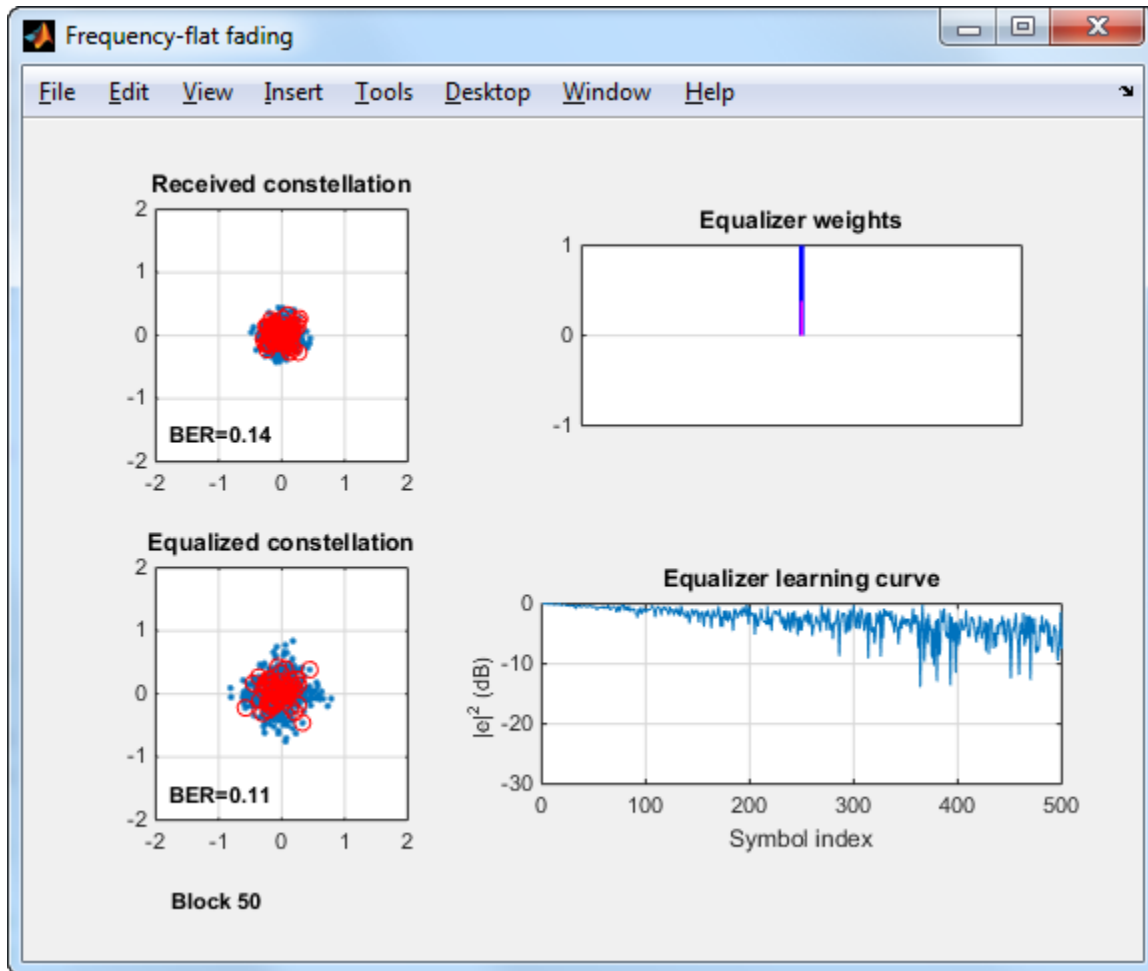
System: comm.RayleighChannel

Properties:
    SampleRate: 4000000
    PathDelays: 0
    AveragePathGains: 0
    NormalizePathGains: true
```



```
MaximumDopplerShift: 30
  DopplerSpectrum: [1x1 struct]
    RandomStream: 'mt19937ar with seed'
    Seed: 73
PathGainsOutputPort: false

    EqType: 'Linear Equalizer'
    AlgType: 'LMS'
    nWeights: 1
    nSampPerSym: 1
    RefTap: 1
    SigConst: [1.0000 + 0.0000i 0.0000 + 1.0000i -1.0000 + 0.0000i -0.0000 - 1.0000i]
    StepSize: 0.1000
    LeakageFactor: 1
    Weights: 0
    WeightInputs: 0
ResetBeforeFiltering: 1
NumSamplesProcessed: 0
```



Simulation 2: Linear Equalization for Frequency-Selective Fading

Simulate a three-path, frequency-selective Rayleigh fading channel. The receiver uses an 8-tap linear RLS (recursive least squares) equalizer with symbol-spaced taps.

```
simName = 'Linear equalization for frequency-selective fading';

% Reset transmit and receive filters
reset(hTxFilt);
reset(hRxFilt);
```

```

% Set the Rayleigh channel System object to be frequency-selective
release(hRayleighChan);
hRayleighChan.PathDelays = [0 0.9 1.5]/Rsym;
hRayleighChan.AveragePathGains = [0 -3 -6];

% Configure an adaptive equalizer
nWeights = 8;
forgetFactor = 0.99; % RLS algorithm forgetting factor
alg = rls(forgetFactor); % RLS algorithm object
eqObj = lineareq(nWeights,alg,PSKConstellation);
eqObj.RefTap = 3; % Reference tap
eqDelayInSym = (eqObj.RefTap-1)/sampPerSymPostRx;

% Link simulation and store BER values
BERvect = zeros(1,nBlocks);
for block = 1:nBlocks
    commadapteqloop;
    BERvect(block) = BEREq;
end
avgBER2 = mean(BERvect)

System: comm.RayleighChannel

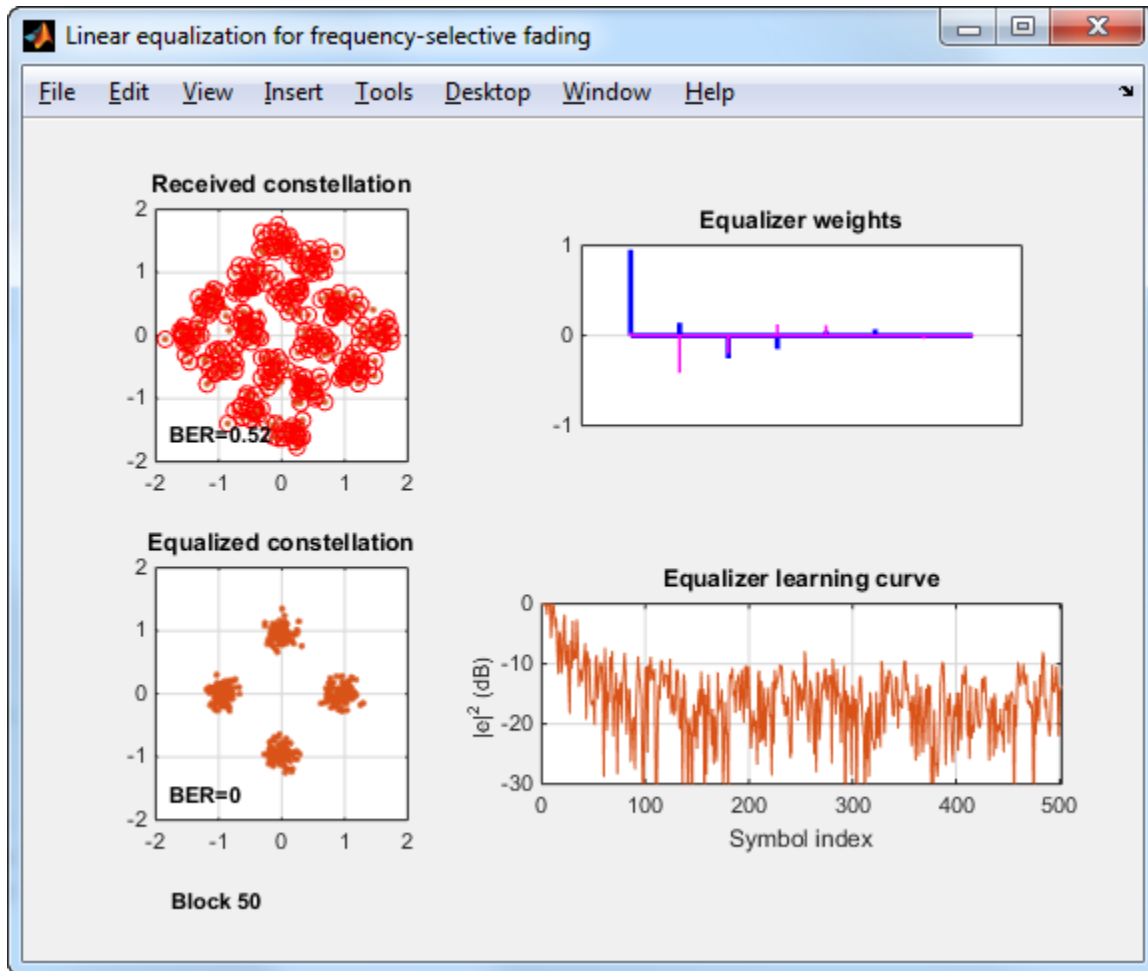
Properties:
    SampleRate: 4000000
    PathDelays: [0 9e-07 1.5e-06]
    AveragePathGains: [0 -3 -6]
    NormalizePathGains: true
    MaximumDopplerShift: 30
    DopplerSpectrum: [1x1 struct]
        RandomStream: 'mt19937ar with seed'
        Seed: 73
    PathGainsOutputPort: false

    EqType: 'Linear Equalizer'
    AlgType: 'RLS'
    nWeights: 8
    nSampPerSym: 1
    RefTap: 3
    SigConst: [1.0000 + 0.0000i 0.0000 + 1.0000i -1.0000 + 0.0000i -0.0000 - 1.0000i]
    ForgetFactor: 0.9900
    InvCorrInit: 0.1000
    InvCorrMatrix: [8x8 double]
        Weights: [0 0 0 0 0 0 0 0]
        WeightInputs: [0 0 0 0 0 0 0 0]
    ResetBeforeFiltering: 1
    NumSamplesProcessed: 0

avgBER2 =

    3.0000e-04

```



Simulation 3: Decision feedback Equalization (DFE) for Frequency-Selective Fading

The receiver uses a DFE with a six-tap fractionally spaced forward filter (two samples per symbol) and two feedback weights. The DFE uses the same RLS algorithm as in Simulation 2. The receive filter structure is reconstructed to account for the increased number of samples per symbol.

```

simName = 'Decision feedback equalization (DFE) for frequency-selective fading';

% Reset transmit filter and adjust receive filter decimation factor
reset(hTxFilt);
release(hRxFilt);
hRxFilt.DecimationFactor = 2;
sampPerSymPostRx = sampPerSymChan/hRxFilt.DecimationFactor;
chanFilterDelay = chanFilterSpan*sampPerSymPostRx;

% Reset fading channel
reset(hRayleighChan);

% Configure an adaptive equalizer object
nFwdWeights = 6; % Number of feedforward equalizer weights
nFbkWeights = 2; % Number of feedback filter weights
eqObj = dfe(nFwdWeights, nFbkWeights,alg,PSKConstellation,sampPerSymPostRx);
eqObj.RefTap = 3;
eqDelayInSym = (eqObj.RefTap-1)/sampPerSymPostRx;

for block = 1:nBlocks
    commadapteqloop;
    BERvect(block) = BEREq;
end
avgBER3 = mean(BERvect)

System: comm.RayleighChannel

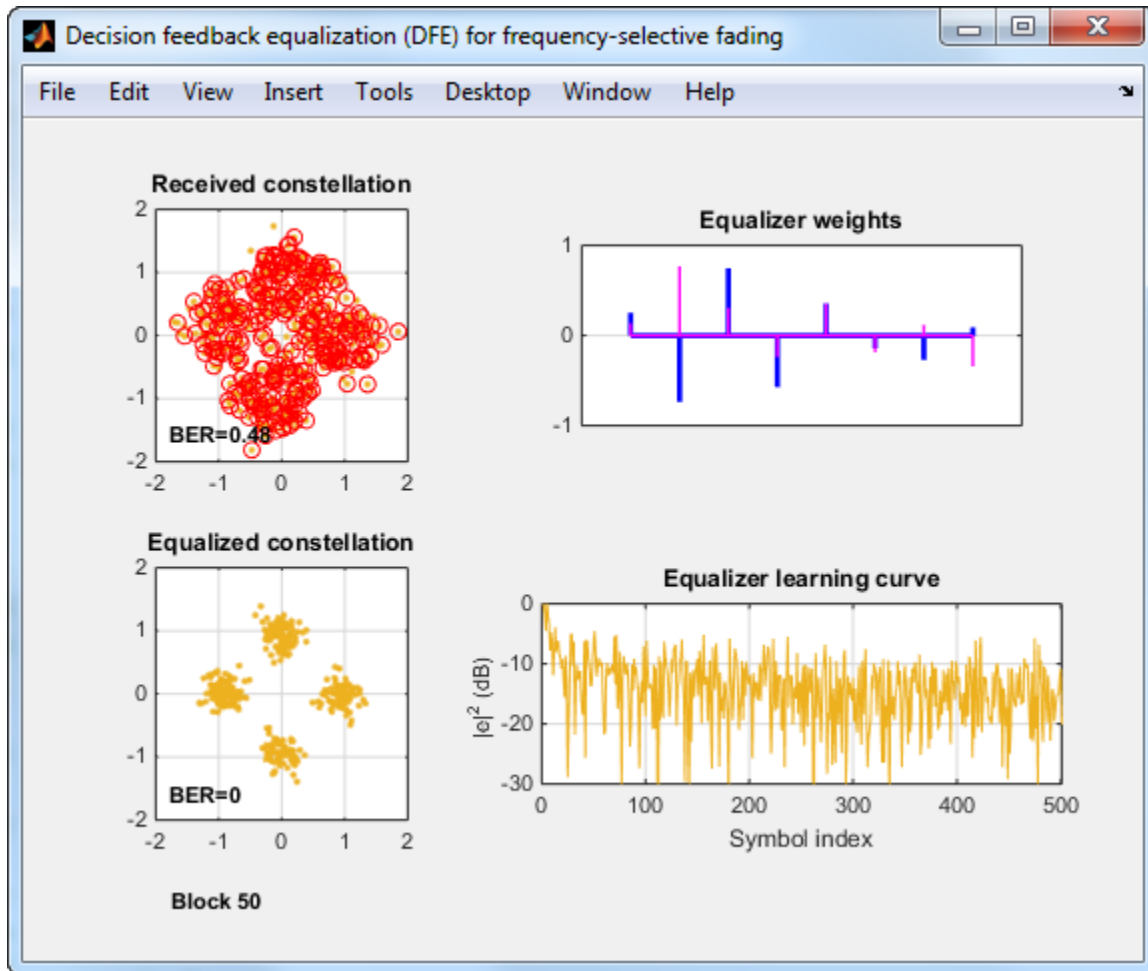
Properties:
    SampleRate: 4000000
    PathDelays: [0 9e-07 1.5e-06]
    AveragePathGains: [0 -3 -6]
    NormalizePathGains: true
    MaximumDopplerShift: 30
    DopplerSpectrum: [1x1 struct]
    RandomStream: 'mt19937ar with seed'
    Seed: 73
    PathGainsOutputPort: false

    EqType: 'Decision Feedback Equalizer'
    AlgType: 'RLS'
    nWeights: [6 2]
    nSampPerSym: 2
    RefTap: 3
    SigConst: [1.0000 + 0.0000i 0.0000 + 1.0000i -1.0000 + 0.0000i -0.0000 - 1.0000i]
    ForgetFactor: 0.9900
    InvCorrInit: 0.1000
    InvCorrMatrix: [8x8 double]
    Weights: [0 0 0 0 0 0 0 0]
    WeightInputs: [0 0 0 0 0 0 0 0]
    ResetBeforeFiltering: 1
    NumSamplesProcessed: 0

```

avgBER3 =

0



Summary

This example showed the relative performance of linear and decision feedback equalizers in both frequency-flat and frequency-selective fading channels. It showed how a one-tap equalizer is sufficient to compensate for a frequency-flat channel, but that a frequency-selective channel requires an equalizer with multiple taps. Finally, it showed that a

decision feedback equalizer is superior to a linear equalizer in a frequency-selective channel.

Appendix

This example uses the following script and helper functions:

- `commadapteqloop.m`
- `commadapteq_checkvars.m`
- `commadapteq_graphics.m`

Equalize BSPK Signal

Equalize a BPSK signal using a linear equalizer with an least mean square (LMS) algorithm.

Generate random binary data and apply BPSK modulation.

```
data = randi([0 1],1000,1);  
modData = pskmod(data,2);
```

Apply two-tap static fading to the modulated signal.

```
rxSig = conv(modData,[0.5 0.05]);
```

Create an LMS adaptive algorithm object with a step size of 0.06.

```
alg = lms(0.06);
```

Create a linear equalizer object having 8 taps using the previously created algorithm object. Set the reference tap index to 4.

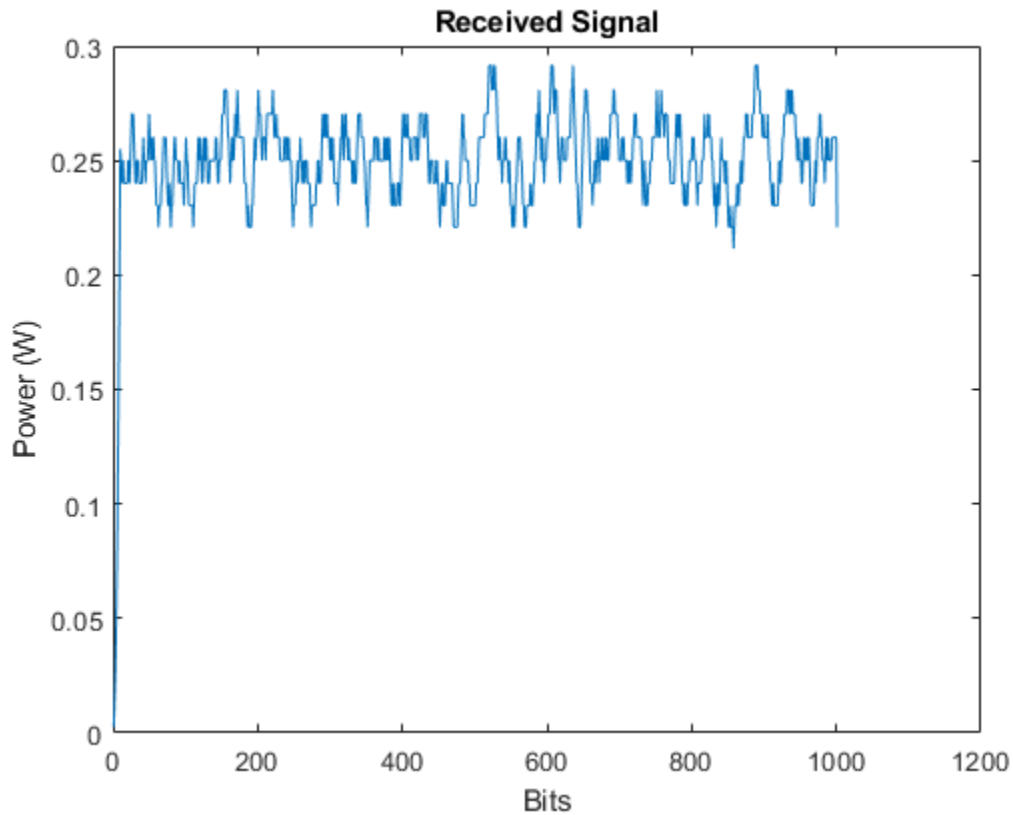
```
eqlms = lineareq(8,alg);  
eqlms.RefTap = 4;
```

Equalize the received signal, rxSig, while using the first 200 data bits as a training sequence.

```
trSeq = data(1:200);  
[eqSig,~,e] = equalize(eqlms,rxSig,trSeq);
```

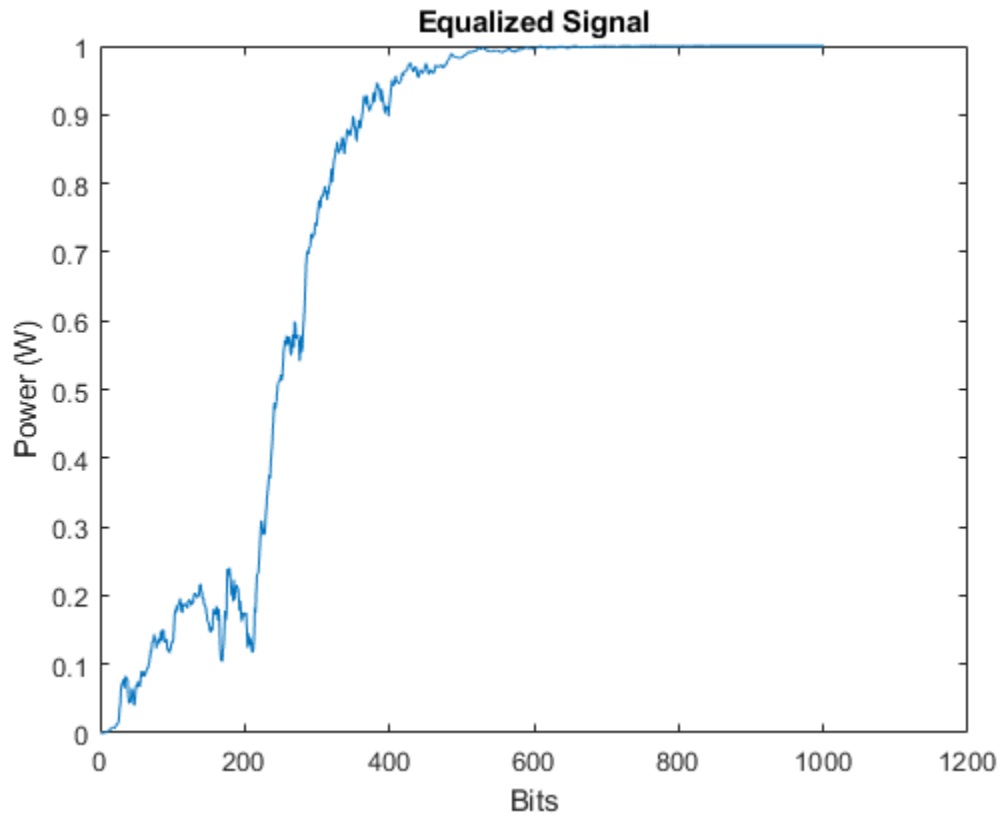
Filter and plot the power of the received (nonequalized) signal. The magnitude of the signal has been attenuated by the channel.

```
rxSigPwr = filter(0.1*ones(10,1),1,abs(rxSig)).^2;  
plot(rxSigPwr)  
title('Received Signal')  
xlabel('Bits')  
ylabel('Power (W)')
```

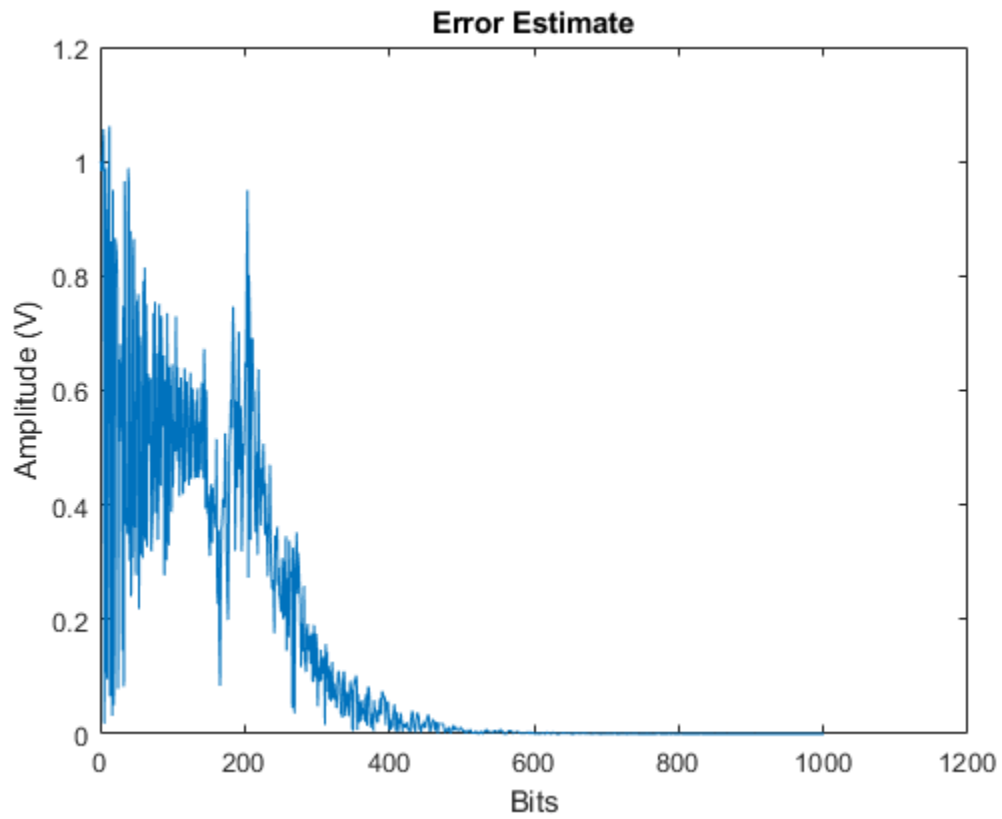
Plot the equalized signal. The signal reaches the intended power level of 1 W.

```
eqSigPwr = filter(0.1*ones(10,1),1,abs(eqSig)).^2;  
plot(eqSigPwr)  
title('Equalized Signal')  
xlabel('Bits')  
ylabel('Power (W)')
```



Plot the magnitude of the error estimate, e . The error decreases until it is nearly zero after 400 bits.

```
plot(abs(e))  
title('Error Estimate')  
xlabel('Bits')  
ylabel('Amplitude (V)')
```



Compare RLS and LMS Algorithms

Equalize a QAM signal passed through a frequency-selective fading channel using RLS and LMS algorithms. Compare the performance of the two algorithms.

Specify the modulation order. Generate the corresponding QAM reference constellation.

```
M = 16;  
sigConst = qammod(0:M-1,M,'UnitAveragePower',true);
```

Create a frequency-selective static channel having three taps.

```
rchan = comm.RayleighChannel('SampleRate',1000, ...  
    'PathDelays',[0 1e-3 2e-3],'AveragePathGains',[0 -3 -6], ...  
    'MaximumDopplerShift',0, ...  
    'RandomStream','mt19937ar with seed','Seed',73);
```

RLS Equalizer

Create an RLS equalizer object.

```
eqrls = lineareq(6,rls(0.99,0.1));  
eqrls.SigConst = sigConst;  
eqrls.ResetBeforeFiltering = 0;
```

Generate and QAM modulate a random training sequence. Pass the sequence through the Rayleigh fading channel. Pass the received signal and the training signal through the equalizer to set the equalizer tap weights.

```
trainData = randi([0 M-1],200,1);  
trainSig = qammod(trainData,M,'UnitAveragePower',true);  
rxSig = rchan(trainSig);  
[~,~,errorSig] = equalize(eqrls,rxSig,trainSig);
```

Plot the magnitude of the error estimate.

```
plot(abs(errorSig))  
title('Error Estimate, RLS Equalizer')  
xlabel('Symbols')  
ylabel('Amplitude')
```

The error is nearly eliminated within 200 symbols.

Transmit a QAM signal through a frequency-selective channel. Equalize the received signal using the previously 'trained' RLS equalizer. Measure the time required to execute the processing loop.

```
tic
for k = 1:20
    data = randi([0 M-1],1000,1); % Random message
    txSig = qammod(data,M,'UnitAveragePower',true);

    % Introduce channel distortion.
    rxSig = rchan(txSig);

    % Equalize the received signal.
    eqSig = equalize(eqlms,rxSig);

end
rlstime = toc;
```

Plot the constellation diagram of the received and equalized signals.

```
h = scatterplot(rxSig,1,0,'c. ');
hold on
scatterplot(eqSig,1,0,'b.',h)
legend('Received Signal','Equalized Signal')
title('RLS Equalizer')
hold off
```

The equalizer removed the effects of the fading channel.

LMS Equalizer

Repeat the equalization process with an LMS equalizer. Create an LMS equalizer object.

```
eqlms = lineareq(6,lms(0.03));
eqlms.SigConst = sigConst;
eqlms.ResetBeforeFiltering = 0;
```

Train the LMS equalizer.

```
trainData = randi([0 M-1],1000,1);
trainSig = qammod(trainData,M,'UnitAveragePower',true);
rxSig = rchan(trainSig);
[~,~,errorSig] = equalize(eqlms,rxSig,trainSig);
```

Plot the magnitude of the error estimate.

```
plot(abs(errorSig))
title('Error Estimate, LMS Equalizer')
xlabel('Symbols')
ylabel('Amplitude')
```

Training the LMS equalizer requires 1000 symbols.

Transmit a QAM signal through the same frequency-selective channel. Equalize the received signal using the previously 'trained' LMS equalizer. Measure the time required to execute the processing loop.

```
tic
for k = 1:20
    data = randi([0 M-1],1000,1); % Random message
    txSig = qammod(data,M,'UnitAveragePower',true);

    % Introduce channel distortion.
    rxSig = rchan(txSig);

    % Equalize the received signal.
    eqSig = equalize(eqlms,rxSig);

end
lmstime = toc;
```

Plot the constellation diagram of the received and equalized signals.

```
h = scatterplot(rxSig,1,0,'c. ');
hold on
scatterplot(eqSig,1,0,'b.',h)
legend('Received Signal','Equalized Signal')
title('LMS Equalizer')
```

The equalizer removes the effects of the fading channel.

Compare the loop execution time for the two equalizer algorithms.

```
[rlstime lmstime]
```

The LMS algorithm is more computationally efficient as it took 50% of the time to execute the processing loop. However, the training sequence required by the LMS algorithm is 5 times longer.

System Design

- “Source Coding” on page 8-2
- “Error Detection and Correction” on page 8-16
- “Interleaving” on page 8-154
- “Digital Modulation” on page 8-172
- “Analog Passband Modulation” on page 8-197
- “Phase-Locked Loops” on page 8-204
- “Multiple-Input Multiple-Output (MIMO)” on page 8-207
- “Huffman Coding” on page 8-210
- “Differential Pulse Code Modulation” on page 8-213
- “Compand a Signal” on page 8-217
- “Arithmetic Coding” on page 8-219
- “Quantization” on page 8-221

Source Coding

In this section...

“Represent Partitions” on page 8-2
“Represent Codebooks” on page 8-3
“Determine Which Interval Each Input Is In” on page 8-3
“Optimize Quantization Parameters” on page 8-4
“Differential Pulse Code Modulation” on page 8-5
“Optimize DPCM Parameters” on page 8-7
“Comband a Signal” on page 8-8
“Huffman Coding” on page 8-10
“Arithmetic Coding” on page 8-12
“Quantize a Signal” on page 8-13

Represent Partitions

Scalar quantization is a process that maps all inputs within a specified range to a common value. This process maps inputs in a different range of values to a different common value. In effect, scalar quantization digitizes an analog signal. Two parameters determine a quantization: a partition on page 8-2 and a codebook on page 8-3.

A quantization partition defines several contiguous, nonoverlapping ranges of values within the set of real numbers. To specify a partition in the MATLAB environment, list the distinct endpoints of the different ranges in a vector.

For example, if the partition separates the real number line into the four sets

- $\{x: x \leq 0\}$
- $\{x: 0 < x \leq 1\}$
- $\{x: 1 < x \leq 3\}$
- $\{x: 3 < x\}$

then you can represent the partition as the three-element vector

```
partition = [0,1,3];
```


The length of the partition vector is one less than the number of partition intervals.

Represent Codebooks

A codebook tells the quantizer which common value to assign to inputs that fall into each range of the partition. Represent a codebook as a vector whose length is the same as the number of partition intervals. For example, the vector

```
codebook = [-1, 0.5, 2, 3];
```

is one possible codebook for the partition $[0, 1, 3]$.

Determine Which Interval Each Input Is In

The `quantiz` function also returns a vector that tells which interval each input is in. For example, the output below says that the input entries lie within the intervals labeled 0, 6, and 5, respectively. Here, the 0th interval consists of real numbers less than or equal to 3; the 6th interval consists of real numbers greater than 8 but less than or equal to 9; and the 5th interval consists of real numbers greater than 7 but less than or equal to 8.

```
partition = [3,4,5,6,7,8,9];
index = quantiz([2 9 8],partition)
```

The output is

```
index =
     0
     6
     5
```

If you continue this example by defining a codebook vector such as

```
codebook = [3,3,4,5,6,7,8,9];
```

then the equation below relates the vector `index` to the quantized signal `quants`.

```
quants = codebook(index+1);
```

This formula for `quants` is exactly what the `quantiz` function uses if you instead phrase the example more concisely as below.

```
partition = [3,4,5,6,7,8,9];
codebook = [3,3,4,5,6,7,8,9];
[index,quants] = quantiz([2 9 8],partition,codebook);
```

Optimize Quantization Parameters

- “Section Overview” on page 8-4
- “Example: Optimizing Quantization Parameters” on page 8-4

Section Overview

Quantization distorts a signal. You can reduce distortion by choosing appropriate partition and codebook parameters. However, testing and selecting parameters for large signal sets with a fine quantization scheme can be tedious. One way to produce partition and codebook parameters easily is to optimize them according to a set of so-called *training data*.

Note The training data you use should be typical of the kinds of signals you will actually be quantizing.

Example: Optimizing Quantization Parameters

The `lloyds` function optimizes the partition and codebook according to the Lloyd algorithm. The code below optimizes the partition and codebook for one period of a sinusoidal signal, starting from a rough initial guess. Then it uses these parameters to quantize the original signal using the initial guess parameters as well as the optimized parameters. The output shows that the mean square distortion after quantizing is much less for the optimized parameters. The `quantiz` function automatically computes the mean square distortion and returns it as the third output parameter.

```
% Start with the setup from 2nd example in "Quantizing a Signal."
t = [0:.1:2*pi];
sig = sin(t);
partition = [-1:.2:1];
codebook = [-1.2:.2:1];
% Now optimize, using codebook as an initial guess.
[partition2,codebook2] = lloyds(sig,codebook);
[index,quants,distor] = quantiz(sig,partition,codebook);
[index2,quant2,distor2] = quantiz(sig,partition2,codebook2);
% Compare mean square distortions from initial and optimized
[distor, distor2] % parameters.
```

The output is

ans =

0.0148 0.0024

Differential Pulse Code Modulation

- “Section Overview” on page 8-5
- “DPCM Terminology” on page 8-5
- “Represent Predictors” on page 8-5
- “Example: DPCM Encoding and Decoding” on page 8-6

Section Overview

The quantization in the section “Quantize a Signal” on page 8-13 requires no *a priori* knowledge about the transmitted signal. In practice, you can often make educated guesses about the present signal based on past signal transmissions. Using such educated guesses to help quantize a signal is known as *predictive quantization*. The most common predictive quantization method is differential pulse code modulation (DPCM).

The functions `dpcmenco`, `dpcmdeco`, and `dpcmopt` can help you implement a DPCM predictive quantizer with a linear predictor.

DPCM Terminology

To determine an encoder for such a quantizer, you must supply not only a partition and codebook as described in “Represent Partitions” on page 8-2 and “Represent Codebooks” on page 8-3, but also a *predictor*. The predictor is a function that the DPCM encoder uses to produce the educated guess at each step. A linear predictor has the form

$$y(k) = p(1)x(k-1) + p(2)x(k-2) + \dots + p(m-1)x(k-m+1) + p(m)x(k-m)$$

where x is the original signal, $y(k)$ attempts to predict the value of $x(k)$, and p is an m -tuple of real numbers. Instead of quantizing x itself, the DPCM encoder quantizes the *predictive error*, $x-y$. The integer m above is called the *predictive order*. The special case when $m = 1$ is called *delta modulation*.

Represent Predictors

If the guess for the k th value of the signal x , based on earlier values of x , is

$$y(k) = p(1)x(k-1) + p(2)x(k-2) + \dots + p(m-1)x(k-m+1) + p(m)x(k-m)$$

then the corresponding predictor vector for toolbox functions is

$$\text{predictor} = [0, p(1), p(2), p(3), \dots, p(m-1), p(m)]$$

Note The initial zero in the predictor vector makes sense if you view the vector as the polynomial transfer function of a finite impulse response (FIR) filter.

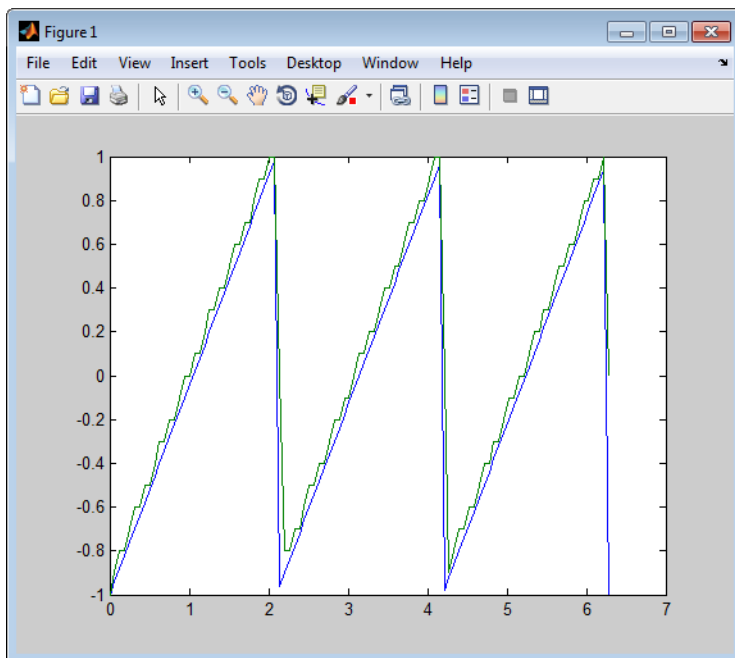
Example: DPCM Encoding and Decoding

A simple special case of DPCM quantizes the difference between the signal's current value and its value at the previous step. Thus the predictor is just $y(k) = x(k-1)$. The code below implements this scheme. It encodes a sawtooth signal, decodes it, and plots both the original and decoded signals. The solid line is the original signal, while the dashed line is the recovered signals. The example also computes the mean square error between the original and decoded signals.

```
predictor = [0 1]; % y(k)=x(k-1)
partition = [-1:.1:.9];
codebook = [-1:.1:1];
t = [0:pi/50:2*pi];
x = sawtooth(3*t); % Original signal
% Quantize x using DPCM.
encodedx = dpcmenco(x,codebook,partition,predictor);
% Try to recover x from the modulated signal.
decodedx = dpcmdeco(encodedx,codebook,predictor);
plot(t,x,t,decodedx,'--')
legend('Original signal','Decoded signal','Location','NorthOutside');
distor = sum((x-decodedx).^2)/length(x) % Mean square error
```

The output is

```
distor =
    0.0327
```



Optimize DPCM Parameters

- “Section Overview” on page 8-7
- “Example: Comparing Optimized and Nonoptimized DPCM Parameters” on page 8-8

Section Overview

The section “Optimize Quantization Parameters” on page 8-4 describes how to use training data with the `lloyd` function to help find quantization parameters that will minimize signal distortion.

This section describes similar procedures for using the `dpcmopt` function in conjunction with the two functions `dpcmenco` and `dpcmdeco`, which first appear in the previous section.

Note The training data you use with `dpcmopt` should be typical of the kinds of signals you will actually be quantizing with `dpcmenco`.

Example: Comparing Optimized and Nonoptimized DPCM Parameters

This example is similar to the one in the last section. However, where the last example created `predictor`, `partition`, and `codebook` in a straightforward but haphazard way, this example uses the same codebook (now called `initcodebook`) as an initial guess for a new *optimized* codebook parameter. This example also uses the predictive order, 1, as the desired order of the new optimized predictor. The `dpcmopt` function creates these optimized parameters, using the sawtooth signal `x` as training data. The example goes on to quantize the training data itself; in theory, the optimized parameters are suitable for quantizing other data that is similar to `x`. Notice that the mean square distortion here is much less than the distortion in the previous example.

```
t = [0:pi/50:2*pi];
x = sawtooth(3*t); % Original signal
initcodebook = [-1:.1:1]; % Initial guess at codebook
% Optimize parameters, using initial codebook and order 1.
[predictor,codebook,partition] = dpcmopt(x,1,initcodebook);
% Quantize x using DPCM.
encodedx = dpcmenco(x,codebook,partition,predictor);
% Try to recover x from the modulated signal.
decodedx = dpcmdeco(encodedx,codebook,predictor);
distor = sum((x-decodedx).^2)/length(x) % Mean square error
```

The output is

```
distor =

    0.0063
```

Compand a Signal

- “Section Overview” on page 8-8
- “Example: μ -Law Compander” on page 8-9

Section Overview

In certain applications, such as speech processing, it is common to use a logarithm computation, called a *compressor*, before quantizing. The inverse operation of a compressor is called an *expander*. The combination of a compressor and expander is called a *compander*.

The `compand` function supports two kinds of companders: μ -law and A-law companders. Its reference page lists both compressor laws.

Example: μ -Law Compressor

The code below quantizes an exponential signal in two ways and compares the resulting mean square distortions. First, it uses the `quantiz` function with a partition consisting of length-one intervals. In the second trial, `compand` implements a μ -law compressor, `quantiz` quantizes the compressed data, and `compand` expands the quantized data. The output shows that the distortion is smaller for the second scheme. This is because equal-length intervals are well suited to the logarithm of `sig`, but not well suited to `sig`. The figure shows how the compressor changes `sig`.

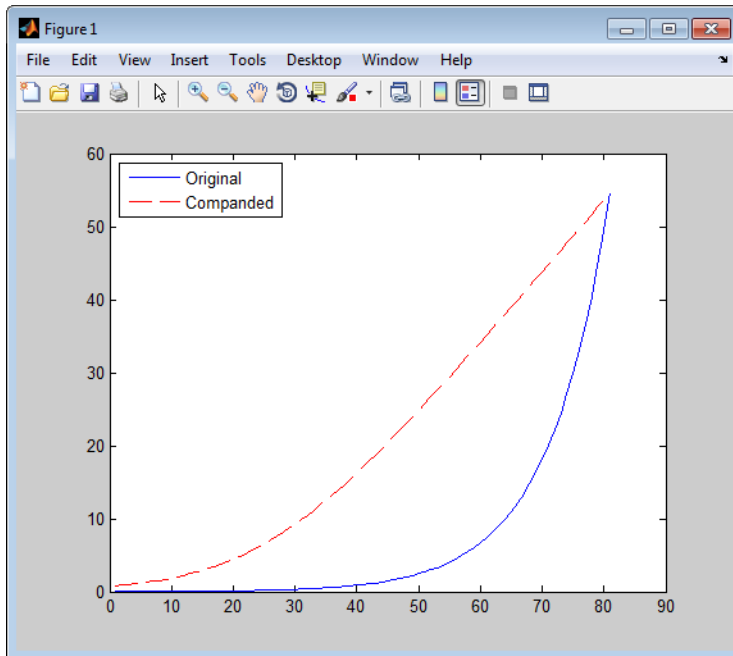
```
Mu = 255; % Parameter for mu-law compressor
sig = -4:.1:4;
sig = exp(sig); % Exponential signal to quantize
V = max(sig);
% 1. Quantize using equal-length intervals and no compressor.
[index,quants,distor] = quantiz(sig,0:floor(V),0:ceil(V));

% 2. Use same partition and codebook, but compress
% before quantizing and expand afterwards.
compsig = compand(sig,Mu,V,'mu/compressor');
[index,quants] = quantiz(compsig,0:floor(V),0:ceil(V));
newsig = compand(quants,Mu,max(quants),'mu/expander');
distor2 = sum((newsig-sig).^2)/length(sig);
[distor, distor2] % Display both mean square distortions.

plot(sig); % Plot original signal.
hold on;
plot(compsig,'r--'); % Plot compressed signal.
legend('Original','Compressed','Location','NorthWest')
```

The output and figure are below.

```
ans =
    0.5348    0.0397
```



Huffman Coding

- “Section Overview” on page 8-10
- “Create a Huffman Code Dictionary in MATLAB” on page 8-11
- “Create and Decode a Huffman Code Using MATLAB” on page 8-12

Section Overview

Huffman coding offers a way to compress data. The average length of a Huffman code depends on the statistical frequency with which the source produces each symbol from its alphabet. A Huffman code dictionary, which associates each data symbol with a codeword, has the property that no codeword in the dictionary is a prefix of any other codeword in the dictionary.

The `huffmandict`, `huffmanenco`, and `huffmandeco` functions support Huffman coding and decoding.

Note For long sequences from sources having skewed distributions and small alphabets, arithmetic coding compresses better than Huffman coding. To learn how to use arithmetic coding, see “Arithmetic Coding” on page 8-12.

Create a Huffman Code Dictionary in MATLAB

Huffman coding requires statistical information about the source of the data being encoded. In particular, the `p` input argument in the `huffmandict` function lists the probability with which the source produces each symbol in its alphabet.

For example, consider a data source that produces 1s with probability 0.1, 2s with probability 0.1, and 3s with probability 0.8. The main computational step in encoding data from this source using a Huffman code is to create a dictionary that associates each data symbol with a codeword. The commands below create such a dictionary and then show the codeword vector associated with a particular value from the data source.

```
symbols = [1 2 3]; % Data symbols
p = [0.1 0.1 0.8]; % Probability of each data symbol
dict = huffmandict(symbols,p) % Create the dictionary.
dict{1,:} % Show one row of the dictionary.
```

The output below shows that the most probable data symbol, 3, is associated with a one-digit codeword, while less probable data symbols are associated with two-digit codewords. The output also shows, for example, that a Huffman encoder receiving the data symbol 1 should substitute the sequence 11.

```
dict =
```

```
    [1]    [1x2 double]
    [2]    [1x2 double]
    [3]    [          0]
```

```
ans =
```

```
    1
```

```
ans =
```

```
    1    1
```

Create and Decode a Huffman Code Using MATLAB

The example below performs Huffman encoding and decoding, using a source whose alphabet has three symbols. Notice that the `huffmanenco` and `huffmandeco` functions use the dictionary that `huffmandict` created.

```
sig = repmat([3 3 1 3 3 3 3 3 2 3],1,50); % Data to encode
symbols = [1 2 3]; % Distinct data symbols appearing in sig
p = [0.1 0.1 0.8]; % Probability of each data symbol
dict = huffmandict(symbols,p); % Create the dictionary.
hcode = huffmanenco(sig,dict); % Encode the data.
dhsig = huffmandeco(hcode,dict); % Decode the code.
```

Arithmetic Coding

- “Section Overview” on page 8-12
- “Represent Arithmetic Coding Parameters” on page 8-12
- “Create and Decode an Arithmetic Code Using MATLAB” on page 8-13

Section Overview

Arithmetic coding offers a way to compress data and can be useful for data sources having a small alphabet. The length of an arithmetic code, instead of being fixed relative to the number of symbols being encoded, depends on the statistical frequency with which the source produces each symbol from its alphabet. For long sequences from sources having skewed distributions and small alphabets, arithmetic coding compresses better than Huffman coding.

The `arithenco` and `arithdeco` functions support arithmetic coding and decoding.

Represent Arithmetic Coding Parameters

Arithmetic coding requires statistical information about the source of the data being encoded. In particular, the `counts` input argument in the `arithenco` and `arithdeco` functions lists the frequency with which the source produces each symbol in its alphabet. You can determine the frequencies by studying a set of test data from the source. The set of test data can have any size you choose, as long as each symbol in the alphabet has a nonzero frequency.

For example, before encoding data from a source that produces 10 x's, 10 y's, and 80 z's in a typical 100-symbol set of test data, define

```
counts = [10 10 80];
```

Alternatively, if a larger set of test data from the source contains 22 x's, 23 y's, and 185 z's, then define

```
counts = [22 23 185];
```

Create and Decode an Arithmetic Code Using MATLAB

The example below performs arithmetic encoding and decoding, using a source whose alphabet has three symbols.

```
seq = repmat([3 3 1 3 3 3 3 2 3],1,50);
counts = [10 10 80];
code = arithenco(seq,counts);
dseq = arithdeco(code,counts,length(seq));
```

Quantize a Signal

- “Scalar Quantization Example 1” on page 8-13
- “Scalar Quantization Example 2” on page 8-14

Scalar Quantization Example 1

The code below shows how the `quantiz` function uses `partition` and `codebook` to map a real vector, `samp`, to a new vector, `quantized`, whose entries are either -1, 0.5, 2, or 3.

```
partition = [0,1,3];
codebook = [-1, 0.5, 2, 3];
samp = [-2.4, -1, -.2, 0, .2, 1, 1.2, 1.9, 2, 2.9, 3, 3.5, 5];
[index,quantized] = quantiz(samp,partition,codebook);
quantized
```

The output is below.

```
quantized =
```

```
Columns 1 through 6
```

```
-1.0000    -1.0000   -1.0000   -1.0000    0.5000    0.5000
```

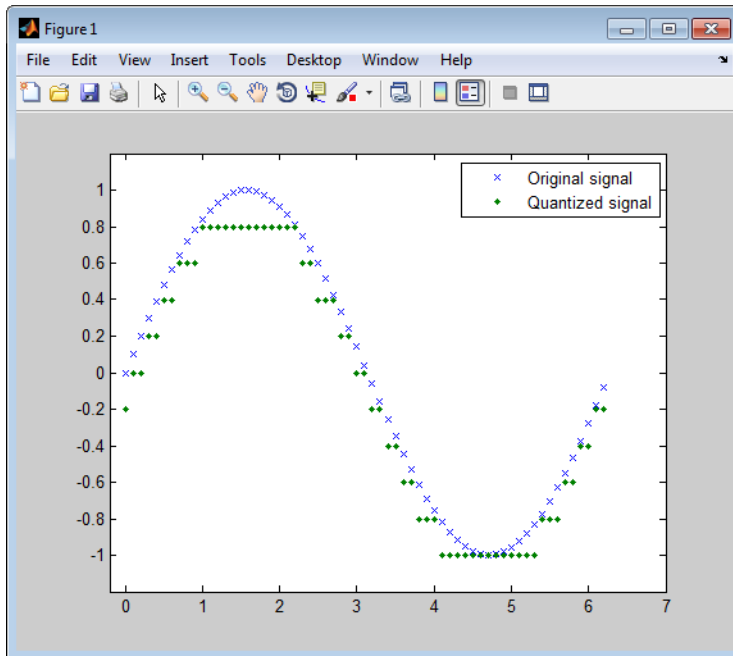
```
Columns 7 through 12
```

```
2.0000    2.0000    2.0000    2.0000    2.0000    3.0000
Column 13
3.0000
```

Scalar Quantization Example 2

This example illustrates the nature of scalar quantization more clearly. After quantizing a sampled sine wave, it plots the original and quantized signals. The plot contrasts the x's that make up the sine curve with the dots that make up the quantized signal. The vertical coordinate of each dot is a value in the vector codebook.

```
t = [0:.1:2*pi]; % Times at which to sample the sine function
sig = sin(t); % Original signal, a sine wave
partition = [-1:.2:1]; % Length 11, to represent 12 intervals
codebook = [-1.2:.2:1]; % Length 12, one entry for each interval
[index,quants] = quantiz(sig,partition,codebook); % Quantize.
plot(t,sig,'x',t,quants, '.')
legend('Original signal','Quantized signal');
axis([-1.2 7 -1.2 1.2])
```



Error Detection and Correction

In this section...
“Cyclic Redundancy Check Codes” on page 8-16
“Block Codes” on page 8-20
“Convolutional Codes” on page 8-37
“Linear Block Codes” on page 8-71
“Hamming Codes” on page 8-81
“BCH Codes” on page 8-89
“Reed-Solomon Codes” on page 8-96
“LDPC Codes” on page 8-107
“Galois Field Computations” on page 8-107
“Galois Fields of Odd Characteristic” on page 8-138

Cyclic Redundancy Check Codes

- “CRC-Code Features” on page 8-16
- “CRC Non-Direct Algorithm” on page 8-17
- “Example Using CRC Non-Direct Algorithm” on page 8-18
- “CRC Direct Algorithm” on page 8-19
- “Selected Bibliography for CRC Coding” on page 8-19

CRC-Code Features

Cyclic redundancy check (CRC) coding is an error-control coding technique for detecting errors that occur when a message is transmitted. Unlike block or convolutional codes, CRC codes do not have a built-in error-correction capability. Instead, when a communications system detects an error in a received message word, the receiver requests the sender to retransmit the message word.

In CRC coding, the transmitter applies a rule to each message word to create extra bits, called the *checksum*, or *syndrome*, and then appends the checksum to the message word. After receiving a transmitted word, the receiver applies the same rule to the received word. If the resulting checksum is nonzero, an error has occurred, and the transmitter should resend the message word.

Open the Error Detection and Correction library by double-clicking its icon in the main Communications Toolbox block library. Open the CRC sublibrary by double-clicking on its icon in the Error Detection and Correction library.

Communications Toolbox supports CRC Coding using Simulink blocks, System objects, or MATLAB objects.

For a list of the CRC coding routines the Communications Toolbox supports see Cyclic Redundancy Check Coding.

CRC Non-Direct Algorithm

The CRC non-direct algorithm accepts a binary data vector, corresponding to a polynomial M , and appends a checksum of r bits, corresponding to a polynomial C . The concatenation of the input vector and the checksum then corresponds to the polynomial $T = M \cdot x^r + C$, since multiplying by x^r corresponds to shifting the input vector r bits to the left. The algorithm chooses the checksum C so that T is divisible by a predefined polynomial P of degree r , called the *generator polynomial*.

The algorithm divides T by P , and sets the checksum equal to the binary vector corresponding to the remainder. That is, if $T = Q \cdot P + R$, where R is a polynomial of degree less than r , the checksum is the binary vector corresponding to R . If necessary, the algorithm prepends zeros to the checksum so that it has length r .

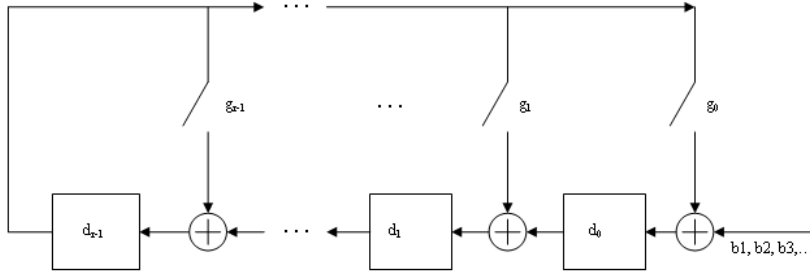
The CRC generation feature, which implements the transmission phase of the CRC algorithm, does the following:

- 1** Left shifts the input data vector by r bits and divides the corresponding polynomial by P .
- 2** Sets the checksum equal to the binary vector of length r , corresponding to the remainder from step 1.
- 3** Appends the checksum to the input data vector. The result is the output vector.

The CRC detection feature computes the checksum for its entire input vector, as described above.

The CRC algorithm uses binary vectors to represent binary polynomials, in descending order of powers. For example, the vector $[1 \ 1 \ 0 \ 1]$ represents the polynomial $x^3 + x^2 + 1$.

Note The implementation described in this section is one of many valid implementations of the CRC algorithm. Different implementations can yield different numerical results.

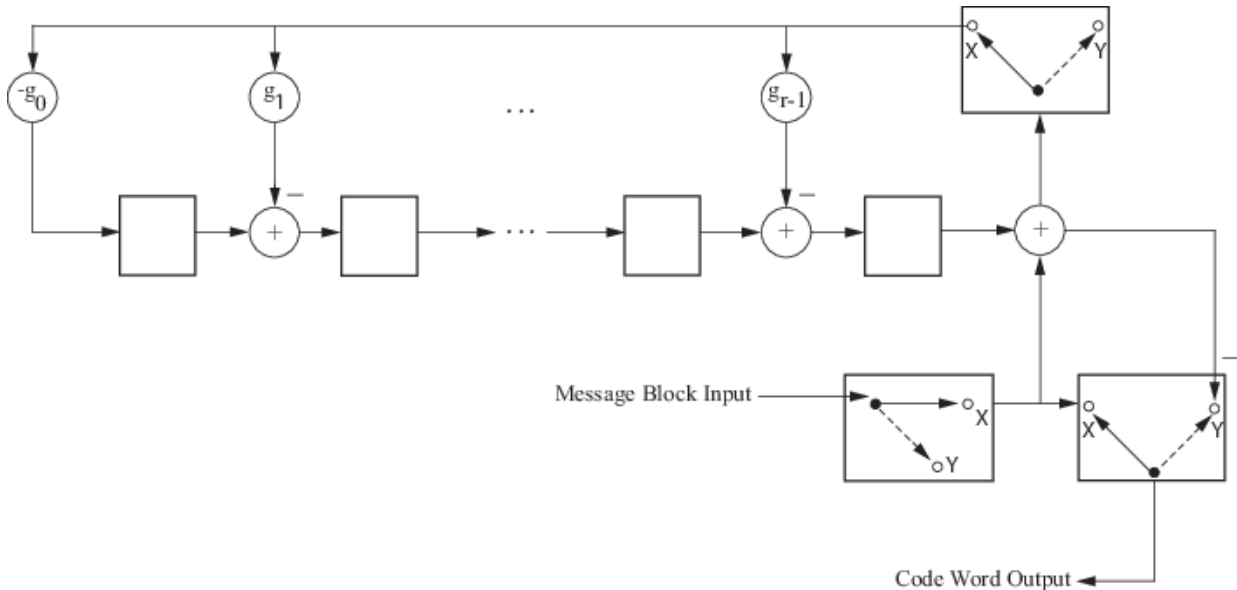


Bits enter the linear feedback shift register (LFSR) from the lowest index bit to the highest index bit. The sequence of input message bits represents the coefficients of a message polynomial in order of decreasing powers. The message vector is augmented with r zeros to flush out the LFSR, where r is the degree of the generator polynomial. If the output from the leftmost register stage $d(1)$ is a 1, then the bits in the shift register are XORed with the coefficients of the generator polynomial. When the augmented message sequence is completely sent through the LFSR, the register contains the checksum $[d(1) d(2) \dots d(r)]$. This is an implementation of binary long division, in which the message sequence is the divisor (numerator) and the polynomial is the dividend (denominator). The CRC checksum is the remainder of the division operation.

Example Using CRC Non-Direct Algorithm

Suppose the input frame is $[1 \ 1 \ 0 \ 0 \ 1 \ 1 \ 0]'$, corresponding to the polynomial $M = x^6 + x^5 + x^2 + x$, and the generator polynomial is $P = x^3 + x^2 + 1$, of degree $r = 3$. By polynomial division, $M \cdot x^3 = (x^6 + x^3 + x) \cdot P + x$. The remainder is $R = x$, so that the checksum is then $[0 \ 1 \ 0]'$. An extra 0 is added on the left to make the checksum have length 3.

CRC Direct Algorithm



where

Message Block Input is m_0, m_1, \dots, m_{k-1}

Code Word Output is $c_0, c_1, \dots, c_{n-1} = \underbrace{m_0, m_1, \dots, m_{k-1}}_X, \underbrace{d_0, d_1, \dots, d_{n-k-1}}_Y$

The initial step of the direct CRC encoding occurs with the three switches in position X. The algorithm feeds k message bits to the encoder. These bits are the first k bits of the code word output. Simultaneously, the algorithm sends k bits to the linear feedback shift register (LFSR). When the system completely feeds the k th message bit to the LFSR, the switches move to position Y. Here, the LFSR contains the mathematical remainder from the polynomial division. These bits are shifted out of the LFSR and they are the remaining bits (checksum) of the code word output.

Selected Bibliography for CRC Coding

- [1] Sklar, Bernard., *Digital Communications: Fundamentals and Applications*, Englewood Cliffs, NJ, Prentice Hall, 1988.

[2] Wicker, Stephen B., *Error Control Systems for Digital Communication and Storage*, Upper Saddle River, NJ, Prentice Hall, 1995.

Block Codes

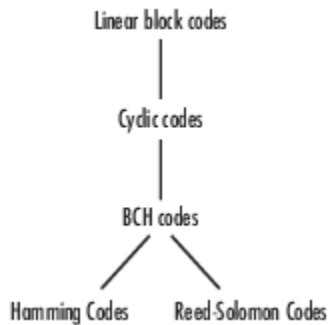
- “Block-Coding Features” on page 8-20
- “Terminology” on page 8-22
- “Data Formats for Block Coding” on page 8-22
- “Using Block Encoders and Decoders Within a Model” on page 8-24
- “Examples of Block Coding” on page 8-25
- “Notes on Specific Block-Coding Techniques” on page 8-27
- “Shortening, Puncturing, and Erasures” on page 8-30
- “Reed-Solomon Code in Integer Format” on page 8-34
- “Find a Generator Polynomial” on page 8-34
- “Performing Other Block Code Tasks” on page 8-35
- “Selected Bibliography for Block Coding” on page 8-36

Block-Coding Features

Error-control coding techniques detect, and possibly correct, errors that occur when messages are transmitted in a digital communication system. To accomplish this, the encoder transmits not only the information symbols but also extra redundant symbols. The decoder interprets what it receives, using the redundant symbols to detect and possibly correct whatever errors occurred during transmission. You might use error-control coding if your transmission channel is very noisy or if your data is very sensitive to noise. Depending on the nature of the data or noise, you might choose a specific type of error-control coding.

Block coding is a special case of error-control coding. Block-coding techniques map a fixed number of message symbols to a fixed number of code symbols. A block coder treats each block of data independently and is a memoryless device. Communications Toolbox contains block-coding capabilities by providing Simulink blocks, System objects, and MATLAB functions.

The class of block-coding techniques includes categories shown in the diagram below.



Communications Toolbox supports general linear block codes. It also process cyclic, BCH, Hamming, and Reed-Solomon codes (which are all special kinds of linear block codes). Blocks in the product can encode or decode a message using one of the previously mentioned techniques. The Reed-Solomon and BCH decoders indicate how many errors they detected while decoding. The Reed-Solomon coding blocks also let you decide whether to use symbols or bits as your data.

Note The blocks and functions in Communications Toolbox are designed for error-control codes that use an alphabet having 2 or 2^m symbols.

Communications Toolbox Support Functions

Functions in Communications Toolbox can support simulation blocks by

- Determining characteristics of a technique, such as error-correction capability or possible message lengths
- Performing lower-level computations associated with a technique, such as
 - Computing a truth table
 - Computing a generator or parity-check matrix
 - Converting between generator and parity-check matrices
 - Computing a generator polynomial

For more information about error-control coding capabilities, see Block Codes on page 8-20.

Terminology

Throughout this section, the information to be encoded consists of *message* symbols and the code that is produced consists of *codewords*.

Each block of K message symbols is encoded into a codeword that consists of N message symbols. K is called the message length, N is called the codeword length, and the code is called an $[N,K]$ code.

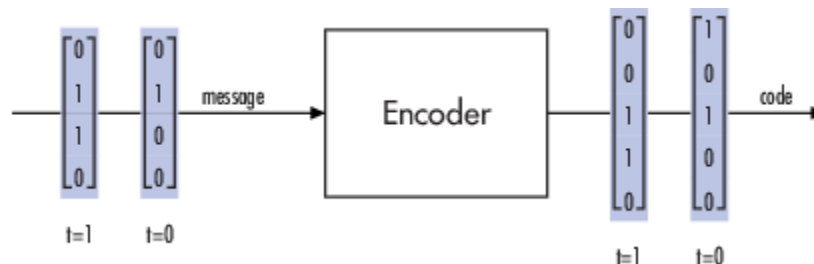
Data Formats for Block Coding

Each message or codeword is an ordered grouping of symbols. Each block in the Block Coding sublibrary processes one word in each time step, as described in the following section, “Binary Format (All Coding Methods)” on page 8-22. Reed-Solomon coding blocks also let you choose between binary and integer data, as described in “Integer Format (Reed-Solomon Only)” on page 8-23.

Binary Format (All Coding Methods)

You can structure messages and codewords as binary *vector* signals, where each vector represents a message word or a codeword. At a given time, the encoder receives an entire message word, encodes it, and outputs the entire codeword. The message and code signals operate over the same sample time.

This example illustrates the encoder receiving a four-bit message and producing a five-bit codeword at time 0. It repeats this process with a new message at time 1.



For all coding techniques *except* Reed-Solomon using binary input, the message vector must have length K and the corresponding code vector has length N . For Reed-Solomon codes with binary input, the symbols for the code are binary sequences of length M , corresponding to elements of the Galois field $GF(2^M)$. In this case, the message vector must have length $M*K$ and the corresponding code vector has length $M*N$. The Binary-

Input RS Encoder block and the Binary-Output RS Decoder block use this format for messages and codewords.

If the input to a block-coding block is a frame-based vector, it must be a column vector instead of a row vector.

To produce sample-based messages in the binary format, you can configure the Bernoulli Binary Generator block so that its **Probability of a zero** parameter is a vector whose length is that of the signal you want to create. To produce frame-based messages in the binary format, you can configure the same block so that its **Probability of a zero** parameter is a scalar and its **Samples per frame** parameter is the length of the signal you want to create.

Using Serial Signals

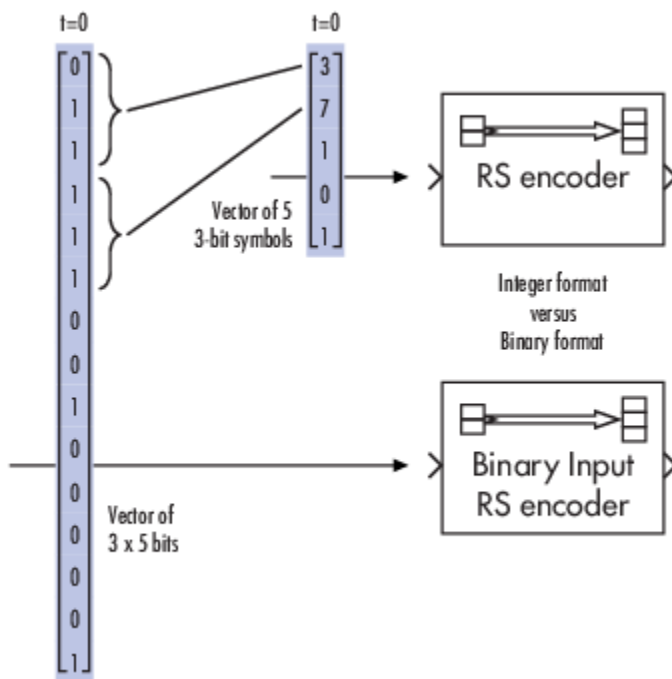
If you prefer to structure messages and codewords as scalar signals, where several samples jointly form a message word or codeword, you can use the Buffer and Unbuffer blocks. Buffering involves latency and multirate processing. If your model computes error rates, the initial delay in the coding-buffering combination influences the **Receive delay** parameter in the Error Rate Calculation block. If you are unsure about the sample times of signals in your model, click the **Display** menu and select **Sample Time > Colors**. Alternatively, you can attach Probe blocks to connector lines to help evaluate sample timing, buffering and delays.

Integer Format (Reed-Solomon Only)

A message word for an $[N,K]$ Reed-Solomon code consists of $M \cdot K$ bits, which you can interpret as K symbols from 0 to 2^M . The symbols are binary sequences of length M , corresponding to elements of the Galois field $GF(2^M)$, in descending order of powers. The integer format for Reed-Solomon codes lets you structure messages and codewords as *integer* signals instead of binary signals. (The input must be a frame-based column vector.)

Note In this context, Simulink expects the *first* bit to be the most significant bit in the symbol. “First” means the smallest index in a vector or the smallest time for a series of scalars.

The following figure illustrates the equivalence between binary and integer signals for a Reed-Solomon encoder. The case for the decoder is similar.



To produce sample-based messages in the integer format, you can configure the Random Integer Generator block so that **M-ary number** and **Initial seed** parameters are vectors of the desired length and all entries of the **M-ary number** vector are 2^M . To produce frame-based messages in the integer format, you can configure the same block so that its **M-ary number** and **Initial seed** parameters are scalars and its **Samples per frame** parameter is the length of the signal you want to create.

Using Block Encoders and Decoders Within a Model

Once you have configured the coding blocks, a few tips can help you place them correctly within your model:

- If a block has multiple outputs, the first one is always the stream of coding data.

The Reed-Solomon and BCH blocks have an error counter as a second output.

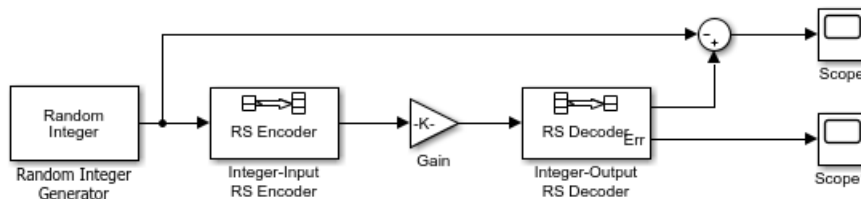
- Be sure that the signal sizes are appropriate for the mask parameters. For example, if you use the Binary Cyclic Encoder block and set **Message length K** to 4, the input signal must be a vector of length 4.

If you are unsure about the size of signals in your model, clicking the **Display** menu select **Signals and Ports >Signal Dimension**.

Examples of Block Coding

Example: Reed-Solomon Code in Integer Format

This example uses a Reed-Solomon code in integer format. It illustrates the appropriate vector lengths of the code and message signals for the coding blocks. It also exhibits error correction, using a simple way of introducing errors into each codeword.



Open the model by typing `doc_rscoding` at the MATLAB command line. To build the model, gather and configure these blocks:

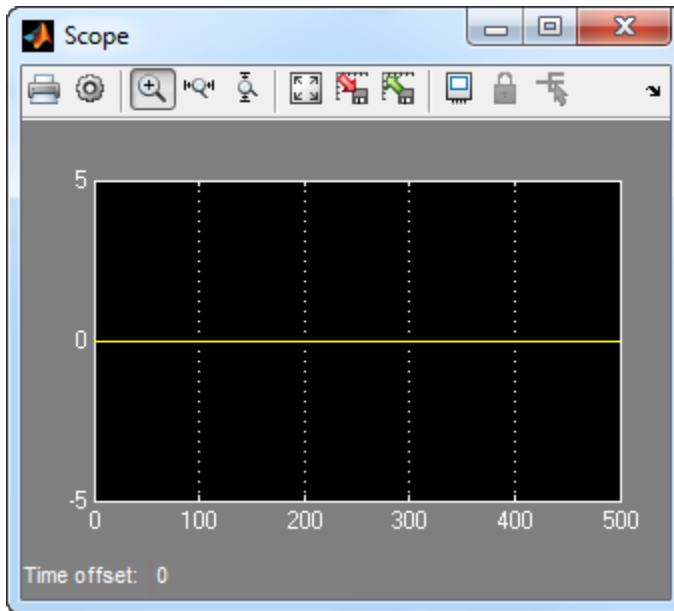
- Random Integer Generator, in the Comm Sources library
 - Set **M-ary number** to 15.
 - Set **Initial seed** to a positive number, `randn` is chosen here.
 - Check the **Frame-based outputs** check box.
 - Set **Samples per frame** to 5.
- Integer-Input RS Encoder
 - Set **Codeword length N** to 15.
 - Set **Message length K** to 5.
- Gain, in the Simulink Math Operations library
 - Set **Gain** to `[0; 0; 0; 0; 0; ones(10,1)]`.
- Integer-Output RS Decoder
 - Set **Codeword length N** to 15.
 - Set **Message length K** to 5.

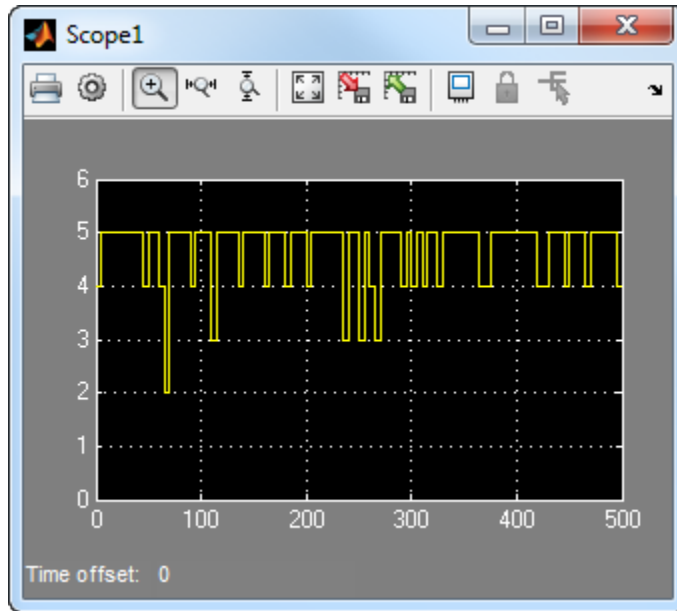
- Scope, in the Simulink Sinks library. Get two copies of this block.
- Sum, in the Simulink Math Operations library
 - Set **List of signs** to | - +

Connect the blocks as in the preceding figure. From the **Simulation** menu of the model window, select **Model Configuration Parameters**. In the Configuration Parameters dialog box, set **Stop Time** to 500.

The vector length numbers appear on the connecting lines only if you click the **Display** menu and select **Signals & Ports > Signal Dimensions**. The encoder accepts a vector of length 5 (which is K in this case) and produces a vector of length 15 (which is N in this case). The decoder does the opposite.

Running the model produces the following scope images. The plotted error count will vary based on the **Initial Seed** value used in the Random Integer Generator block. (To make the axis range exactly match that of the first scope, right-click the plot area in the scope and select **Axes Properties**.)





Number of Errors Before Correction

The second plot is the number of errors that the decoder detected while trying to recover the message. Often the number is five because the Gain block replaces the first five symbols in each codeword with zeros. However, the number of errors is less than five whenever a correct codeword contains one or more zeros in the first five places.

The first plot is the difference between the original message and the recovered message; since the decoder was able to correct all errors that occurred, each of the five data streams in the plot is zero.

Notes on Specific Block-Coding Techniques

Although the Block Coding sublibrary is somewhat uniform in its look and feel, the various coding techniques are not identical. This section describes special options and restrictions that apply to parameters and signals for the coding technique categories in this sublibrary. Coding techniques discussed below include - Generic Linear Block code, Cyclic code, Hamming code, BCH code, and Reed-Solomon code.

Generic Linear Block Codes

Encoding a message using a generic linear block code requires a generator matrix. Decoding the code requires the generator matrix and possibly a truth table. To use the Binary Linear Encoder and Binary Linear Decoder blocks, you must understand the **Generator matrix** and **Error-correction truth table** parameters.

Generator Matrix - The process of encoding a message into an $[N,K]$ linear block code is determined by a K -by- N generator matrix G . Specifically, a 1 -by- K message vector v is encoded into the 1 -by- N codeword vector vG . If G has the form $[I_k, P]$ or $[P, I_k]$, where P is some K -by- $(N-K)$ matrix and I_k is the K -by- K identity matrix, G is said to be in *standard form*. (Some authors, such as Clark and Cain [2], use the first standard form, while others, such as Lin and Costello [3], use the second.) The linear block-coding blocks in this product require the **Generator matrix** mask parameter to be in standard form.

Decoding Table - A decoding table tells a decoder how to correct errors that may have corrupted the code during transmission. Hamming codes can correct any single-symbol error in any codeword. Other codes can correct, or partially correct, errors that corrupt more than one symbol in a given codeword.

The Binary Linear Decoder block allows you to specify a decoding table in the **Error-correction truth table** parameter. Represent a decoding table as a matrix with N columns and 2^{N-K} rows. Each row gives a correction vector for one received codeword vector.

You can avoid specifying a decoding table explicitly, by setting the **Error-correction truth table** parameter to \emptyset . When **Error-correction truth table** is \emptyset , the block computes a decoding table using the `syndtable` function.

Cyclic Codes

For cyclic codes, the codeword length N must have the form 2^M-1 , where M is an integer greater than or equal to 3.

Generator Polynomials - Cyclic codes have special algebraic properties that allow a polynomial to determine the coding process completely. This so-called generator polynomial is a degree- $(N-K)$ divisor of the polynomial x^N-1 . Van Lint [5] explains how a generator polynomial determines a cyclic code.

The Binary Cyclic Encoder and Binary Cyclic Decoder blocks allow you to specify a generator polynomial as the second mask parameter, instead of specifying K there. The blocks represent a generator polynomial using a vector that lists the coefficients of the polynomial in order of *ascending* powers of the variable. You can find generator polynomials for cyclic codes using the `cyclpoly` function.

If you do not want to specify a generator polynomial, set the second mask parameter to the value of K .

Hamming Codes

For Hamming codes, the codeword length N must have the form $2^M - 1$, where M is an integer greater than or equal to 3. The message length K must equal $N - M$.

Primitive Polynomials - Hamming codes rely on algebraic fields that have 2^M elements (or, more generally, p^M elements for a prime number p). Elements of such fields are named *relative to a distinguished element of the field that is called a primitive element*. The minimal polynomial of a primitive element is called a *primitive polynomial*. The Hamming Encoder and Hamming Decoder blocks allow you to specify a primitive polynomial for the finite field that they use for computations. If you want to specify this polynomial, do so in the second mask parameter field. The blocks represent a primitive polynomial using a vector that lists the coefficients of the polynomial in order of *ascending* powers of the variable. You can find generator polynomials for Galois fields using the `gfprimfd` function.

If you do not want to specify a primitive polynomial, set the second mask parameter to the value of K .

BCH Codes

BCH codes are cyclic error-correcting codes that are constructed using finite fields. For these codes, the codeword length N must have the form $2^M - 1$, where M is an integer from 3 to 9. The message length K is restricted to particular values that depend on N . To see which values of K are valid for a given N , see the `comm.BCHEncoder` System object reference page. No known analytic formula describes the relationship among the codeword length, message length, and error-correction capability for BCH codes.

Narrow-Sense BCH Codes

The narrow-sense generator polynomial is $\text{LCM}[m_1(x), m_2(x), \dots, m_{2t}(x)]$, where:

- LCM represents the least common multiple,
- $m_i(x)$ represents the minimum polynomial corresponding to α^i , α is a root of the default primitive polynomial for the field $\text{GF}(n+1)$,
- and t represents the error-correcting capability of the code.

Reed-Solomon Codes

Reed-Solomon codes are useful for correcting errors that occur in bursts. In the simplest case, the length of codewords in a Reed-Solomon code is of the form $N = 2^M - 1$, where the 2^M is the number of symbols for the code. The error-correction capability of a Reed-Solomon code is $\text{floor}((N-K)/2)$, where K is the length of message words. The difference $N-K$ must be even.

It is sometimes convenient to use a shortened Reed-Solomon code in which N is less than $2^M - 1$. In this case, the encoder appends $2^M - 1 - N$ zero symbols to each message word and codeword. The error-correction capability of a shortened Reed-Solomon code is also $\text{floor}((N-K)/2)$. The Communications Toolbox Reed-Solomon blocks can implement shortened Reed-Solomon codes.

Effect of Nonbinary Symbols - One difference between Reed-Solomon codes and the other codes supported in this product is that Reed-Solomon codes process symbols in $GF(2^M)$ instead of $GF(2)$. M bits specify each symbol. The nonbinary nature of the Reed-Solomon code symbols causes the Reed-Solomon blocks to differ from other coding blocks in these ways:

- You can use the integer format, via the Integer-Input RS Encoder and Integer-Output RS Decoder blocks.
- The binary format expects the vector lengths to be an integer multiple of $M \cdot K$ (not K) for messages and the same integer $M \cdot N$ (not N) for codewords.

Error Information - The Reed-Solomon decoding blocks (Binary-Output RS Decoder and Integer-Output RS Decoder) return error-related information during the simulation. The second output signal indicates the number of errors that the block detected in the input codeword. A -1 in the second output indicates that the block detected more errors than it could correct using the coding scheme.

Shortening, Puncturing, and Erasures

Many standards utilize punctured codes, and digital receivers can easily output erasures. BCH and RS performance improves significantly in fading channels where the receiver generates erasures.

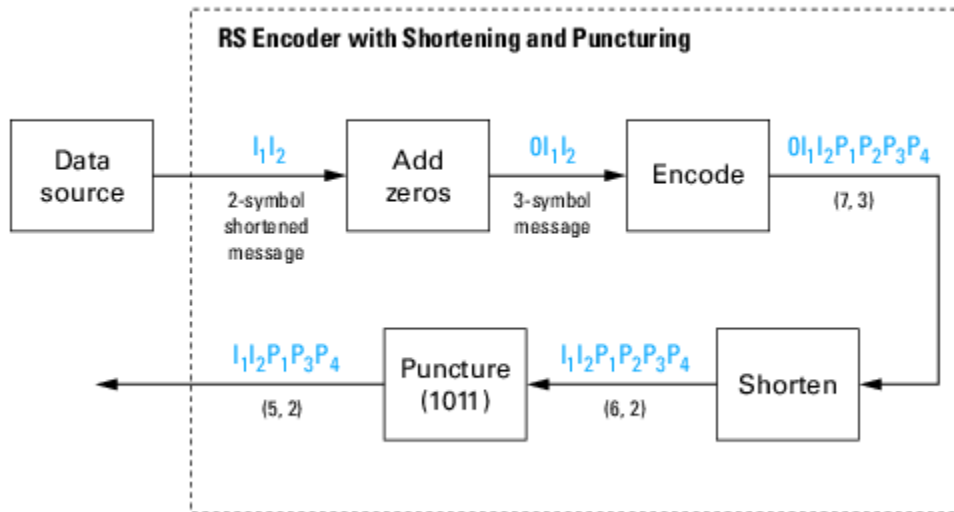
A *punctured codeword* has only parity symbols removed, and a *shortened codeword* has only information symbols removed. A codeword with erasures can have those erasures in either information symbols or parity symbols.

Reed Solomon Examples with Shortening, Puncturing, and Erasures

In this section, a representative example of Reed Solomon coding with shortening, puncturing, and erasures is built with increasing complexity of error correction.

Encoder Example with Shortening and Puncturing

The following figure shows a representative example of a (7,3) Reed Solomon encoder with shortening and puncturing.

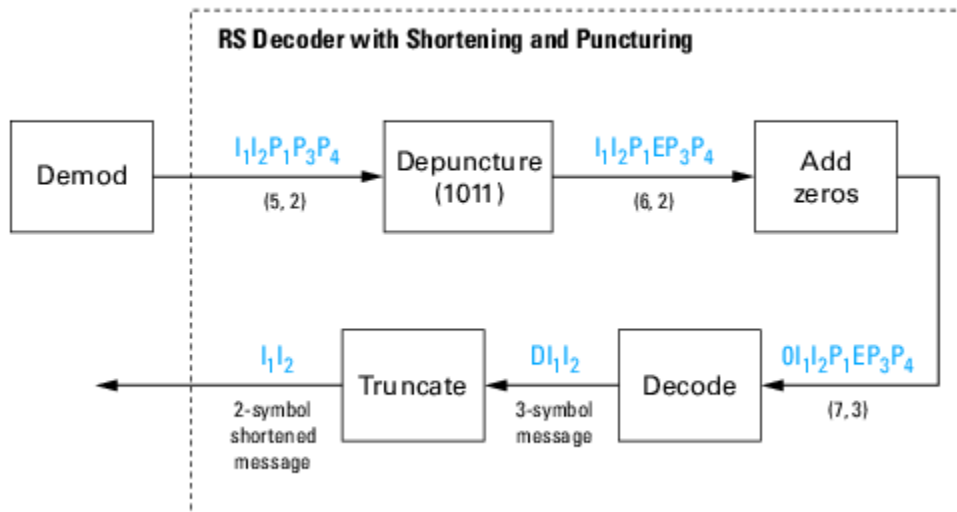


In this figure, the message source outputs two information symbols, designated by I_1I_2 . (For a BCH example, the symbols are binary bits.) Because the code is a shortened (7,3) code, a zero must be added ahead of the information symbols, yielding a three-symbol message of $0I_1I_2$. The modified message sequence is RS encoded, and the added information zero is then removed, which yields a result of $I_1I_2P_1P_2P_3P_4$. (In this example, the parity bits are at the end of the codeword.)

The puncturing operation is governed by the puncture vector, which, in this case, is 1011. Within the puncture vector, a 1 means that the symbol is kept, and a 0 means that the symbol is thrown away. In this example, the puncturing operation removes the second parity symbol, yielding a final vector of $I_1I_2P_1P_3P_4$.

Decoder Example with Shortening and Puncturing

The following figure shows how the RS decoder operates on a shortened and punctured codeword.

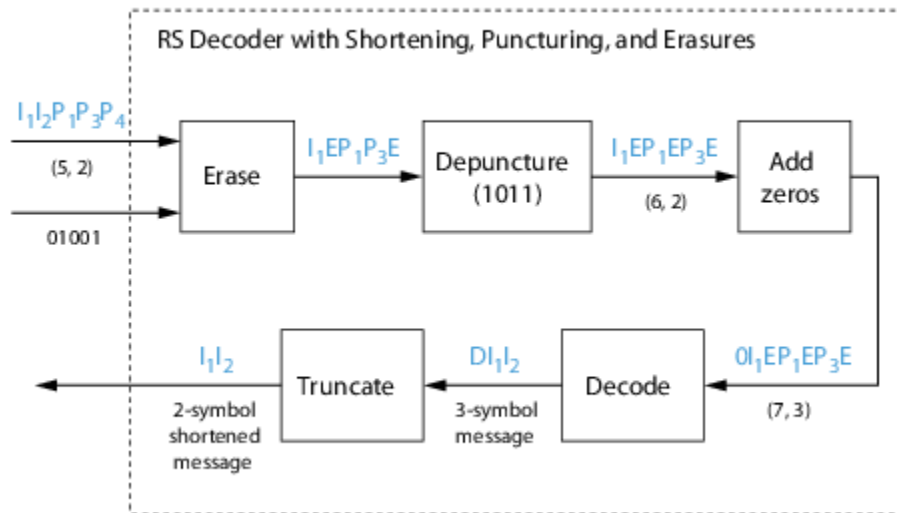


This case corresponds to the encoder operations shown in the figure of the RS encoder with shortening and puncturing. As shown in the preceding figure, the encoder receives a (5, 2) codeword, because it has been shortened from a (7, 3) codeword by one symbol, and one symbol has also been punctured.

As a first step, the decoder adds an erasure, designated by E, in the second parity position of the codeword. This corresponds to the puncture vector 1011. Adding a zero accounts for shortening, in the same way as shown in the preceding figure. The single erasure does not exceed the erasure-correcting capability of the code, which can correct four erasures. The decoding operation results in the three-symbol message DI_1I_2 . The first symbol is truncated, as in the preceding figure, yielding a final output of I_1I_2 .

Decoder Example with Shortening, Puncturing, and Erasures

The following figure shows the decoder operating on the punctured, shortened codeword, while also correcting erasures generated by the receiver.



In this figure, demodulator receives the $I_1I_2P_1P_3P_4$ vector that the encoder sent. The demodulator declares that two of the five received symbols are unreliable enough to be erased, such that symbols 2 and 5 are deemed to be erasures. The 01001 vector, provided by an external source, indicates these erasures. Within the erasures vector, a 1 means that the symbol is to be replaced with an erasure symbol, and a 0 means that the symbol is passed unaltered.

The decoder blocks receive the codeword and the erasure vector, and perform the erasures indicated by the vector 01001. Within the erasures vector, a 1 means that the symbol is to be replaced with an erasure symbol, and a 0 means that the symbol is passed unaltered. The resulting codeword vector is $I_1EP_1P_3E$, where E is an erasure symbol.

The codeword is then depunctured, according to the puncture vector used in the encoding operation (i.e., 1011). Thus, an erasure symbol is inserted between P_1 and P_3 , yielding a codeword vector of $I_1EP_1EP_3E$.

Just prior to decoding, the addition of zeros at the beginning of the information vector accounts for the shortening. The resulting vector is $0I_1EP_1EP_3E$, such that a (7,3) codeword is sent to the Berlekamp algorithm.

This codeword is decoded, yielding a three-symbol message of DI_1I_2 (where D refers to a dummy symbol). Finally, the removal of the D symbol from the message vector accounts for the shortening and yields the original I_1I_2 vector.

For additional information, see the “Reed-Solomon Coding with Erasures, Punctures, and Shortening” on page 11-20 example.

Reed-Solomon Code in Integer Format

To open an example model that uses a Reed-Solomon code in integer format, type `doc_rscoding` at the MATLAB command line. For more information about the model, see “Example: Reed-Solomon Code in Integer Format” on page 8-25

Find a Generator Polynomial

To find a generator polynomial for a cyclic, BCH, or Reed-Solomon code, use the `cyclpoly`, `bchgenpoly`, or `rsgenpoly` function, respectively. The commands

```
genpolyCyclic = cyclpoly(15,5) % 1+X^5+X^10
genpolyBCH = bchgenpoly(15,5) % x^10+x^8+x^5+x^4+x^2+x+1
genpolyRS = rsgenpoly(15,5)
```

find generator polynomials for block codes of different types. The output is below.

```
genpolyCyclic =
    1    0    0    0    0    1    0    0    0    0    1

genpolyBCH = GF(2) array.
Array elements =
    1    0    1    0    0    1    1    0    1    1    1

genpolyRS = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
Array elements =
    1    4    8    10    12    9    4    2    12    2    7
```

The formats of these outputs vary:

- `cyclpoly` represents a generator polynomial using an integer row vector that lists the polynomial's coefficients in order of *ascending* powers of the variable.

- `bchgenpoly` and `rsgenpoly` represent a generator polynomial using a Galois row vector that lists the polynomial's coefficients in order of *descending* powers of the variable.
- `rsgenpoly` uses coefficients in a Galois field other than the binary field GF(2). For more information on the meaning of these coefficients, see “How Integers Correspond to Galois Field Elements” on page 8-111 and “Polynomials over Galois Fields” on page 8-130.

Nonuniqueness of Generator Polynomials

Some pairs of message length and codeword length do not uniquely determine the generator polynomial. The syntaxes for functions in the example above also include options for retrieving generator polynomials that satisfy certain constraints that you specify. See the functions' reference pages for details about syntax options.

Algebraic Expression for Generator Polynomials

The generator polynomials produced by `bchgenpoly` and `rsgenpoly` have the form $(X - A^b)(X - A^{b+1})\dots(X - A^{b+2t-1})$, where A is a primitive element for an appropriate Galois field, and b and t are integers. See the functions' reference pages for more information about this expression.

Performing Other Block Code Tasks

This section describes functions that compute typical parameters associated with linear block codes, as well as functions that convert information from one format to another.

- **Error Correction Versus Error Detection for Linear Block Codes**

You can use a linear block code to detect $d_{\min} - 1$ errors or to correct $t = \left\lfloor \frac{1}{2}(d_{\min} - 1) \right\rfloor$ errors.

If you compromise the error correction capability of a code, you can detect more than t errors. For example, a code with $d_{\min} = 7$ can correct $t = 3$ errors or it can detect up to 4 errors and correct up to 2 errors.

- **Finding the Error-Correction Capability**

The `bchgenpoly` and `rsgenpoly` functions can return an optional second output argument that indicates the error-correction capability of a BCH or Reed-Solomon code. For example, the commands

```
[g,t] = bchgenpoly(31,16);  
t  
t =  
  
3
```

find that a [31, 16] BCH code can correct up to three errors in each codeword.

- **Finding Generator and Parity-Check Matrices**

To find a parity-check and generator matrix for a Hamming code with codeword length $2^m - 1$, use the `hammgen` function as below. `m` must be at least three.

```
[parmat,genmat] = hammgen(m); % Hamming
```

To find a parity-check and generator matrix for a cyclic code, use the `cyclgen` function. You must provide the codeword length and a valid generator polynomial. You can use the `cyclpoly` function to produce one possible generator polynomial after you provide the codeword length and message length. For example,

```
[parmat,genmat] = cyclgen(7,cyclpoly(7,4)); % Cyclic
```

- **Converting Between Parity-Check and Generator Matrices**

The `gen2par` function converts a generator matrix into a parity-check matrix, and vice versa. The reference page for `gen2par` contains examples to illustrate this.

Selected Bibliography for Block Coding

- [1] Berlekamp, Elwyn R., *Algebraic Coding Theory*, New York, McGraw-Hill, 1968.
- [2] Clark, George C. Jr., and J. Bibb Cain, *Error-Correction Coding for Digital Communications*, New York, Plenum Press, 1981.
- [3] Lin, Shu, and Daniel J. Costello, Jr., *Error Control Coding: Fundamentals and Applications*, Englewood Cliffs, NJ, Prentice-Hall, 1983.
- [4] Peterson, W. Wesley, and E. J. Weldon, Jr., *Error-Correcting Codes*, 2nd ed., Cambridge, MA, MIT Press, 1972.
- [5] van Lint, J. H., *Introduction to Coding Theory*, New York, Springer-Verlag, 1982.
- [6] Wicker, Stephen B., *Error Control Systems for Digital Communication and Storage*, Upper Saddle River, NJ, Prentice Hall, 1995.

[7] Gallager, Robert G., *Low-Density Parity-Check Codes*, Cambridge, MA, MIT Press, 1963.

[8] Ryan, William E., "An introduction to LDPC codes," *Coding and Signal Processing for Magnetic Recording Systems* (Vasic, B., ed.), CRC Press, 2004.

Convolutional Codes

- "Convolutional Code Features" on page 8-37
- "Polynomial Description of a Convolutional Code" on page 8-39
- "Trellis Description of a Convolutional Code" on page 8-42
- "Create and Decode Convolutional Codes" on page 8-46
- "Design a Rate-2/3 Feedforward Encoder Using MATLAB" on page 8-55
- "Design a Rate 2/3 Feedforward Encoder Using Simulink" on page 8-57
- "Puncture a Convolutional Code Using MATLAB" on page 8-60
- "Implement a Systematic Encoder with Feedback Using Simulink" on page 8-61
- "Soft-Decision Decoding" on page 8-62
- "Tailbiting Encoding Using Feedback Encoders" on page 8-70
- "Selected Bibliography for Convolutional Coding" on page 8-71

Convolutional Code Features

Convolutional coding is a special case of error-control coding. Unlike a block coder, a convolutional coder is not a memoryless device. Even though a convolutional coder accepts a fixed number of message symbols and produces a fixed number of code symbols, its computations depend not only on the current set of input symbols but on some of the previous input symbols.

Communications Toolbox provides convolutional coding capabilities as Simulink blocks, System objects, and MATLAB functions. This product supports feedforward and feedback convolutional codes that can be described by a trellis structure or a set of generator polynomials. It uses the Viterbi algorithm to implement hard-decision and soft-decision decoding.

The product also includes an *a posteriori* probability decoder, which can be used for soft output decoding of convolutional codes.

For background information about convolutional coding, see the works listed in Selected Bibliography for Convolutional Coding on page 8-71.

Block Parameters for Convolutional Coding

To process convolutional codes, use the Convolutional Encoder, Viterbi Decoder, and/or APP Decoder blocks in the Convolutional sublibrary. If a mask parameter is required in both the encoder and the decoder, use the same value in both blocks.

The blocks in the Convolutional sublibrary assume that you use one of two different representations of a convolutional encoder:

- If you design your encoder using a diagram with shift registers and modulo-2 adders, you can compute the code generator polynomial matrix and subsequently use the `poly2trellis` function (in Communications Toolbox) to generate the corresponding trellis structure mask parameter automatically. For an example, see “Design a Rate 2/3 Feedforward Encoder Using Simulink” on page 8-57.
- If you design your encoder using a trellis diagram, you can construct the trellis structure in MATLAB and use it as the mask parameter.

For more information about these representations, see Polynomial Description of a Convolutional Code on page 8-39 and Trellis Description of a Convolutional Code on page 8-42.

Using the Polynomial Description in Blocks

To use the polynomial description with the Convolutional Encoder, Viterbi Decoder, or APP Decoder blocks, use the utility function `poly2trellis` from Communications Toolbox. This function accepts a polynomial description and converts it into a trellis description. For example, the following command computes the trellis description of an encoder whose constraint length is 5 and whose generator polynomials are 35 and 31:

```
trellis = poly2trellis(5,[35 31]);
```

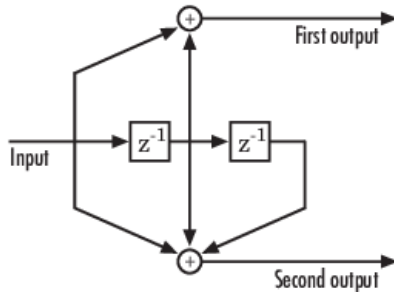
To use this encoder with one of the convolutional-coding blocks, simply place a `poly2trellis` command such as

```
poly2trellis(5,[35 31]);
```

in the **Trellis structure** parameter field.

Polynomial Description of a Convolutional Code

A polynomial description of a convolutional encoder describes the connections among shift registers and modulo 2 adders. For example, the figure below depicts a feedforward convolutional encoder that has one input, two outputs, and two shift registers.



A polynomial description of a convolutional encoder has either two or three components, depending on whether the encoder is a feedforward or feedback type:

- Constraint lengths on page 8-39
- Generator polynomials on page 8-39
- Feedback connection polynomials on page 8-40 (for feedback encoders only)

Constraint Lengths

The constraint lengths of the encoder form a vector whose length is the number of inputs in the encoder diagram. The elements of this vector indicate the number of bits stored in each shift register, *including* the current input bits.

In the figure above, the constraint length is three. It is a scalar because the encoder has one input stream, and its value is one plus the number of shift registers for that input.

Generator Polynomials

If the encoder diagram has k inputs and n outputs, the code generator matrix is a k -by- n matrix. The element in the i th row and j th column indicates how the i th input contributes to the j th output.

For *systematic* bits of a systematic feedback encoder, match the entry in the code generator matrix with the corresponding element of the feedback connection vector. See "Feedback Connection Polynomials" on page 8-40 below for details.

In other situations, you can determine the (i,j) entry in the matrix as follows:

- 1 Build a binary number representation by placing a 1 in each spot where a connection line from the shift register feeds into the adder, and a 0 elsewhere. The leftmost spot in the binary number represents the current input, while the rightmost spot represents the oldest input that still remains in the shift register.
- 2 Convert this binary representation into an octal representation by considering consecutive triplets of bits, starting from the rightmost bit. The rightmost bit in each triplet is the least significant. If the number of bits is not a multiple of three, place zero bits at the left end as necessary. (For example, interpret 1101010 as 001 101 010 and convert it to 152.)

For example, the binary numbers corresponding to the upper and lower adders in the figure above are 110 and 111, respectively. These binary numbers are equivalent to the octal numbers 6 and 7, respectively, so the generator polynomial matrix is [6 7].

Note You can perform the binary-to-octal conversion in MATLAB by using code like `str2num(dec2base(bin2dec('110'),8))`.

For a table of some good convolutional code generators, refer to [2] in the section “Selected Bibliography for Block Coding” on page 8-36, especially that book's appendices.

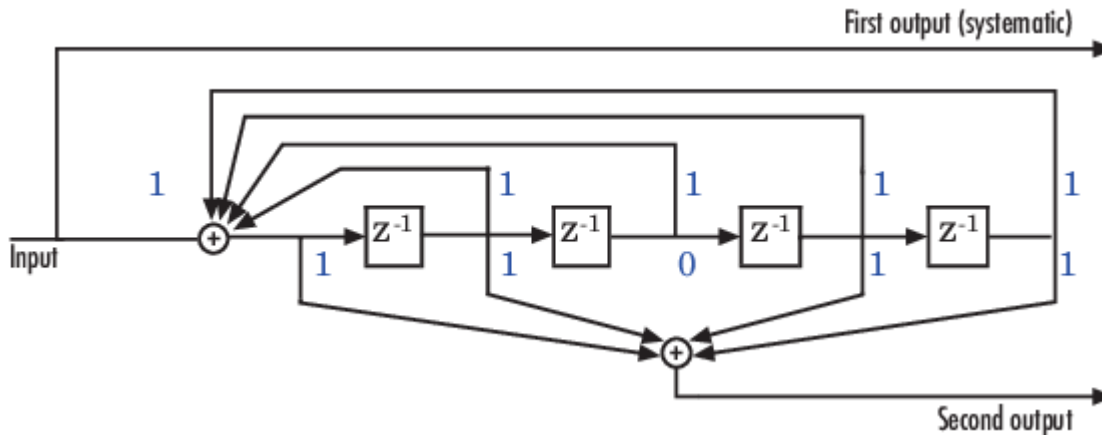
Feedback Connection Polynomials

If you are representing a feedback encoder, you need a vector of feedback connection polynomials. The length of this vector is the number of inputs in the encoder diagram. The elements of this vector indicate the feedback connection for each input, using an octal format. First build a binary number representation as in step 1 above. Then convert the binary representation into an octal representation as in step 2 above.

If the encoder has a feedback configuration and is also systematic, the code generator and feedback connection parameters corresponding to the systematic bits must have the same values.

Trellis Structure for 1/2 Feedback Convolutional Encoder

Create a trellis structure for the rate 1/2 systematic convolutional encoder with feedback that is shown. Use the trellis to encode and decode a random bit stream.



This encoder has a constraint length of 5, a generator polynomial matrix of $[37\ 33]$, and a feedback connection polynomial of 37.

The first generator polynomial matches the feedback connection polynomial because the first output corresponds to the systematic bits. The feedback polynomial is represented by the binary vector $[1\ 1\ 1\ 1\ 1]$, corresponding to the upper row of binary digits in the diagram. These digits indicate connections from the outputs of the registers to the adder. The initial 1 corresponds to the input bit. The octal representation of the binary number 11111 is 37.

The second generator polynomial is represented by the binary vector $[1\ 1\ 0\ 1\ 1]$, corresponding to the lower row of binary digits in the diagram. The octal number corresponding to the binary number 11011 is 33.

Use `poly2trellis` to convert the polynomial to a trellis structure. When used with a feedback polynomial, `poly2trellis` makes a feedback connection to the input of the trellis.

```
trellis = poly2trellis(5, [37 33], 37)
```

```
trellis = struct with fields:
    numInputSymbols: 2
    numOutputSymbols: 4
```

```
numStates: 16
nextStates: [16x2 double]
outputs: [16x2 double]
```

Generate random binary data, convolutionally encode the data, and decode the data using the Viterbi algorithm.

```
data = randi([0 1],70,1);
codedData = convenc(data,trellis);
decodedData = vitdec(codedData,trellis,34,'trunc','hard');
```

Verify the decoded data has no bit errors.

```
biterr(data,decodedData)
```

```
ans = 0
```

Using the Polynomial Description in MATLAB

To use the polynomial description with the functions `convenc` and `vitdec`, first convert it into a trellis description using the `poly2trellis` function. For example, the command below computes the trellis description of the encoder pictured in the section Polynomial Description of a Convolutional Code on page 8-39.

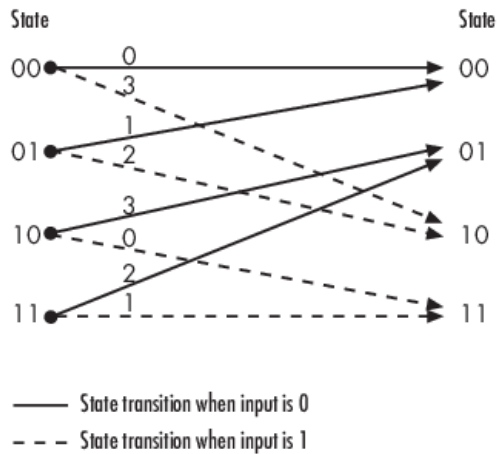
```
trellis = poly2trellis(3,[6 7]);
```

The MATLAB structure `trellis` is a suitable input argument for `convenc` and `vitdec`.

Trellis Description of a Convolutional Code

A trellis description of a convolutional encoder shows how each possible input to the encoder influences both the output and the state transitions of the encoder. This section describes trellises, and how to represent on page 8-43 trellises in MATLAB, and gives an example of a MATLAB trellis on page 8-45.

The figure below depicts a trellis for the convolutional encoder from the previous section. The encoder has four states (numbered in binary from 00 to 11), a one-bit input, and a two-bit output. (The ratio of input bits to output bits makes this encoder a rate-1/2 encoder.) Each solid arrow shows how the encoder changes its state if the current input is zero, and each dashed arrow shows how the encoder changes its state if the current input is one. The octal numbers above each arrow indicate the current output of the encoder.



As an example of interpreting this trellis diagram, if the encoder is in the 10 state and receives an input of zero, it outputs the code symbol 3 and changes to the 01 state. If it is in the 10 state and receives an input of one, it outputs the code symbol 0 and changes to the 11 state.

Note that any polynomial description of a convolutional encoder is equivalent to some trellis description, although some trellises have no corresponding polynomial descriptions.

Specifying a Trellis in MATLAB

To specify a trellis in MATLAB, use a specific form of a MATLAB structure called a trellis structure. A trellis structure must have five fields, as in the table below.

Fields of a Trellis Structure for a Rate k/n Code

Field in Trellis Structure	Dimensions	Meaning
numInputSymbols	Scalar	Number of input symbols to the encoder: 2^k
numOutputsymbols	Scalar	Number of output symbols from the encoder: 2^n
numStates	Scalar	Number of states in the encoder
nextStates	numStates-by- 2^k matrix	Next states for all combinations of current state and current input
outputs	numStates-by- 2^k matrix	Outputs (in octal) for all combinations of current state and current input

Note While your trellis structure can have any name, its fields must have the *exact* names as in the table. Field names are case sensitive.

In the `nextStates` matrix, each entry is an integer between 0 and `numStates-1`. The element in the i th row and j th column denotes the next state when the starting state is $i-1$ and the input bits have decimal representation $j-1$. To convert the input bits to a decimal value, use the first input bit as the most significant bit (MSB). For example, the second column of the `nextStates` matrix stores the next states when the current set of input values is $\{0, \dots, 0, 1\}$. To learn how to assign numbers to states, see the reference page for `istrellis`.

In the `outputs` matrix, the element in the i th row and j th column denotes the encoder's output when the starting state is $i-1$ and the input bits have decimal representation $j-1$. To convert to decimal value, use the first output bit as the MSB.

How to Create a MATLAB Trellis Structure

Once you know what information you want to put into each field, you can create a trellis structure in any of these ways:

- Define each of the five fields individually, using `structurename.fieldname` notation. For example, set the first field of a structure called `s` using the command below. Use additional commands to define the other fields.

```
s.numInputSymbols = 2;
```

The reference page for the `istrellis` function illustrates this approach.

- Collect all field names and their values in a single `struct` command. For example:

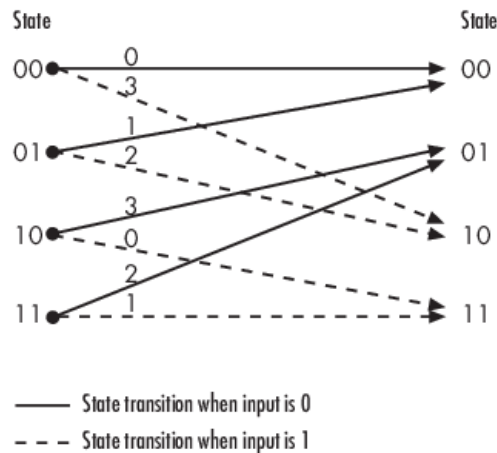
```
s = struct('numInputSymbols',2,'numOutputSymbols',2,...
          'numStates',2,'nextStates',[0 1;0 1],'outputs',[0 0;1 1]);
```

- Start with a polynomial description of the encoder and use the `poly2trellis` function to convert it to a valid trellis structure. For more information, see Polynomial Description of a Convolutional Code on page 8-39.

To check whether your structure is a valid trellis structure, use the `istrellis` function.

Example: A MATLAB Trellis Structure

Consider the trellis shown below.



To build a trellis structure that describes it, use the command below.

```
trellis = struct('numInputSymbols',2,'numOutputSymbols',4,...
                'numStates',4,'nextStates',[0 2;0 2;1 3;1 3],...
                'outputs',[0 3;1 2;3 0;2 1]);
```

The number of input symbols is 2 because the trellis diagram has two types of input path: the solid arrow and the dashed arrow. The number of output symbols is 4 because the numbers above the arrows can be either 0, 1, 2, or 3. The number of states is 4 because there are four bullets on the left side of the trellis diagram (equivalently, four on the right side). To compute the matrix of next states, create a matrix whose rows correspond to the four current states on the left side of the trellis, whose columns correspond to the inputs of 0 and 1, and whose elements give the next states at the end of the arrows on the right side of the trellis. To compute the matrix of outputs, create a matrix whose rows and columns are as in the next states matrix, but whose elements give the octal outputs shown above the arrows in the trellis.

Create and Decode Convolutional Codes

The functions for encoding and decoding convolutional codes are `convenc` and `vitdec`. This section discusses using these functions to create and decode convolutional codes.

Encoding

A simple way to use `convenc` to create a convolutional code is shown in the commands below.

Define a trellis.

```
t = poly2trellis([4 3],[4 5 17;7 4 2]);
```

Encode a vector of ones.

```
x = ones(100,1);  
code = convenc(x,t);
```

The first command converts a polynomial description of a feedforward convolutional encoder to the corresponding trellis description. The second command encodes 100 bits, or 50 two-bit symbols. Because the code rate in this example is $2/3$, the output vector `code` contains 150 bits (that is, 100 input bits times $3/2$).

To check whether your trellis corresponds to a catastrophic convolutional code, use the `iscatastrophic` function.

Hard-Decision Decoding

To decode using hard decisions, use the `vitdec` function with the flag `'hard'` and with *binary* input data. Because the output of `convenc` is binary, hard-decision decoding can use the output of `convenc` directly, without additional processing. This example extends the previous example and implements hard-decision decoding.

Define a trellis.

```
t = poly2trellis([4 3],[4 5 17;7 4 2]);
```

Encode a vector of ones.

```
code = convenc(ones(100,1),t);
```

Set the traceback length for decoding and decode using `vitdec`.

```
tb = 2;  
decoded = vitdec(code,t,tb,'trunc','hard');
```

Verify that the decoded data is a vector of 100 ones.

```
isequal(decoded,ones(100,1))
```

```
ans = logical  
     1
```

Soft-Decision Decoding

To decode using soft decisions, use the `vitdec` function with the flag `'soft'`. Specify the number, `nsdec`, of soft-decision bits and use input data consisting of integers between 0 and $2^{\text{nsdec}} - 1$.

An input of 0 represents the most confident 0, while an input of $2^{\text{nsdec}} - 1$ represents the most confident 1. Other values represent less confident decisions. For example, the table below lists interpretations of values for 3-bit soft decisions.

Input Values for 3-bit Soft Decisions

Input Value	Interpretation
0	Most confident 0
1	Second most confident 0
2	Third most confident 0
3	Least confident 0
4	Least confident 1
5	Third most confident 1
6	Second most confident 1
7	Most confident 1

Implement Soft-Decision Decoding Using MATLAB

The script below illustrates decoding with 3-bit soft decisions. First it creates a convolutional code with `convenc` and adds white Gaussian noise to the code with `awgn`. Then, to prepare for soft-decision decoding, the example uses `quantiz` to map the noisy data values to appropriate decision-value integers between 0 and 7. The second argument in `quantiz` is a partition vector that determines which data values map to 0, 1, 2, etc. The partition is chosen so that values near 0 map to 0, and values near 1 map to 7. (You can refine the partition to obtain better decoding performance if your application requires it.) Finally, the example decodes the code and computes the bit error rate. When comparing the decoded data with the original message, the example must take the decoding delay into account. The continuous operation mode of `vitdec` causes a delay equal to the traceback length, so `msg(1)` corresponds to `decoded(tblen+1)` rather than to `decoded(1)`.

```
s = RandStream.create('mt19937ar', 'seed', 94384);
prevStream = RandStream.setGlobalStream(s);
msg = randi([0 1], 4000, 1); % Random data
t = poly2trellis(7, [171 133]); % Define trellis.
% Create a ConvolutionalEncoder System object
hConvEnc = comm.ConvolutionalEncoder(t);
% Create an AWGNChannel System object.
hChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)', ...
    'SNR', 6);
% Create a ViterbiDecoder System object
hVitDec = comm.ViterbiDecoder(t, 'InputFormat', 'Soft', ...
    'SoftInputWordLength', 3, 'TracebackDepth', 48, ...
```

```

    'TerminationMethod', 'Continuous');
% Create a ErrorRate Calculator System object. Account for the receive
% delay caused by the traceback length of the viterbi decoder.
hErrorCalc = comm.ErrorRate('ReceiveDelay', 48);
ber = zeros(3,1); % Store BER values
code = step(hConvEnc,msg); % Encode the data.
hChan.SignalPower = (code'*code)/length(code);
ncode = step(hChan,code); % Add noise.

% Quantize to prepare for soft-decision decoding.
qcode = quantiz(ncode,[0.001,.1,.3,.5,.7,.9,.999]);

tblen = 48; delay = tblen; % Traceback length
decoded = step(hVitDec,qcode); % Decode.

% Compute bit error rate.
ber = step(hErrorCalc, msg, decoded);
ratio = ber(1)
number = ber(2)
RandStream.setGlobalStream(prevStream);

```

The output is below.

```
number =
```

```
5
```

```
ratio =
```

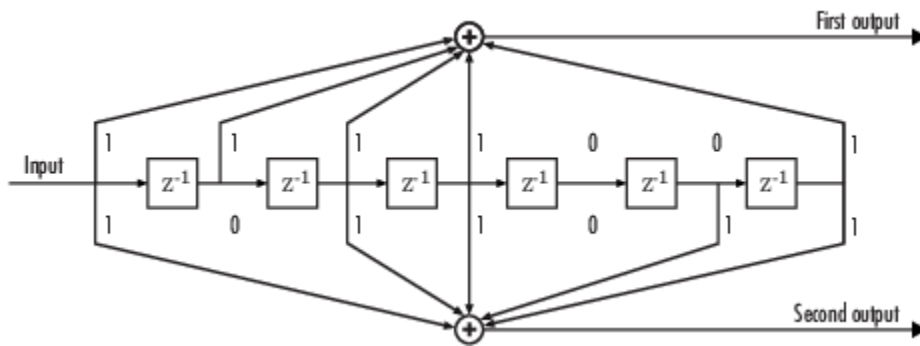
```
0.0013
```

Implement Soft-Decision Decoding Using Simulink

This example creates a rate 1/2 convolutional code. It uses a quantizer and the Viterbi Decoder block to perform soft-decision decoding. To open the model, enter `doc_softdecision` at the MATLAB command line. For a description of the model, see Overview of the Simulation on page 8-63.

Defining the Convolutional Code

The feedforward convolutional encoder in this example is depicted below.



The encoder's constraint length is a scalar since the encoder has one input. The value of the constraint length is the number of bits stored in the shift register, including the current input. There are six memory registers, and the current input is one bit. Thus the constraint length of the code is 7.

The code generator is a 1-by-2 matrix of octal numbers because the encoder has one input and two outputs. The first element in the matrix indicates which input values contribute to the first output, and the second element in the matrix indicates which input values contribute to the second output.

For example, the first output in the encoder diagram is the modulo-2 sum of the rightmost and the four leftmost elements in the diagram's array of input values. The seven-digit binary number 1111001 captures this information, and is equivalent to the octal number 171. The octal number 171 thus becomes the first entry of the code generator matrix. Here, each triplet of bits uses the leftmost bit as the most significant bit. The second output corresponds to the binary number 1011011, which is equivalent to the octal number 133. The code generator is therefore [171 133].

The **Trellis structure** parameter in the Convolutional Encoder block tells the block which code to use when processing data. In this case, the `poly2trellis` function, in Communications Toolbox, converts the constraint length and the pair of octal numbers into a valid trellis structure.

While the message data entering the Convolutional Encoder block is a scalar bit stream, the encoded data leaving the block is a stream of binary vectors of length 2.

Mapping the Received Data

The received data, that is, the output of the AWGN Channel block, consists of complex numbers that are close to -1 and 1. In order to reconstruct the original binary message,

the receiver part of the model must decode the convolutional code. The Viterbi Decoder block in this model expects its input data to be integers between 0 and 7. The demodulator, a custom subsystem in this model, transforms the received data into a format that the Viterbi Decoder block can interpret properly. More specifically, the demodulator subsystem

- Converts the received data signal to a real signal by removing its imaginary part. It is reasonable to assume that the imaginary part of the received data does not contain essential information, because the imaginary part of the transmitted data is zero (ignoring small roundoff errors) and because the channel noise is not very powerful.
- Normalizes the received data by dividing by the standard deviation of the noise estimate and then multiplying by -1.
- Quantizes the normalized data using three bits.

The combination of this mapping and the Viterbi Decoder block's decision mapping reverses the BPSK modulation that the BPSK Modulator Baseband block performs on the transmitting side of this model. To examine the demodulator subsystem in more detail, double-click the icon labeled Soft-Output BPSK Demodulator.

Decoding the Convolutional Code

After the received data is properly mapped to length-2 vectors of 3-bit decision values, the Viterbi Decoder block decodes it. The block uses a soft-decision algorithm with 2^3 different input values because the **Decision type** parameter is `Soft Decision` and the **Number of soft decision bits** parameter is 3.

Soft-Decision Interpretation of Data

When the **Decision type** parameter is set to `Soft Decision`, the Viterbi Decoder block requires input values between 0 and 2^b-1 , where b is the **Number of soft decision bits** parameter. The block interprets 0 as the most confident decision that the codeword bit is a 0 and interprets 2^b-1 as the most confident decision that the codeword bit is a 1. The values in between these extremes represent less confident decisions. The following table lists the interpretations of the eight possible input values for this example.

Decision Value	Interpretation
0	Most confident 0
1	Second most confident 0
2	Third most confident 0

Decision Value	Interpretation
3	Least confident 0
4	Least confident 1
5	Third most confident 1
6	Second most confident 1
7	Most confident 1

Traceback and Decoding Delay

The **Traceback depth** parameter in the Viterbi Decoder block represents the length of the decoding delay. Typical values for a traceback depth are about five or six times the constraint length, which would be 35 or 42 in this example. However, some hardware implementations offer options of 48 and 96. This example chooses 48 because that is closer to the targets (35 and 42) than 96 is.

Delay in Received Data

The **Receive delay** parameter of the Error Rate Calculation block is nonzero because a given message bit and its corresponding recovered bit are separated in time by a nonzero amount of simulation time. The **Receive delay** parameter tells the block which elements of its input signals to compare when checking for errors.

In this case, the Receive delay value is equal to the Traceback depth value (48).

Comparing Simulation Results with Theoretical Results

This section describes how to compare the bit error rate in this simulation with the bit error rate that would theoretically result from unquantized decoding. The process includes these steps

- **Computing Theoretical Bounds for the Bit Error Rate**

To calculate theoretical bounds for the bit error rate P_b of the convolutional code in this model, you can use this estimate based on unquantized-decision decoding:

$$P_b < \sum_{d=f}^{\infty} c_d P_d$$

In this estimate, c_d is the sum of bit errors for error events of distance d , and f is the free distance of the code. The quantity P_d is the pairwise error probability, given by

$$P_d = \frac{1}{2} \operatorname{erfc} \left(\sqrt{dR \frac{E_b}{N_0}} \right)$$

where R is the code rate of $1/2$, and erfc is the MATLAB complementary error function, defined by

$$\operatorname{erfc}(x) = \frac{2}{\sqrt{\pi}} \int_x^{\infty} e^{-t^2} dt$$

Values for the coefficients c_d and the free distance f are in published articles such as Frenger, P., P. Orten, and T. Ottosson, "Convolution Codes with Optimum Distance Spectrum," *IEEE Communications Letters*, vol. 3, pp. 317-319, November 1999. [3]. The free distance for this code is $f = 10$.

The following commands calculate the values of P_b for E_b/N_0 values in the range from 1 to 4, in increments of 0.5:

```

EbNoVec = [1:0.5:4.0];
R = 1/2;
% Errs is the vector of sums of bit errors for
% error events at distance d, for d from 10 to 29.
Errs = [36 0 211 0 1404 0 11633 0 77433 0 502690 0, ...
        3322763 0 21292910 0 134365911 0 843425871 0];
% P is the matrix of pairwise error probabilities, for
% Eb/No values in EbNoVec and d from 10 to 29.
P = zeros(20,7); % Initialize.
for d = 10:29
    P(d-9,:) = (1/2)*erfc(sqrt(d*R*10.^(EbNoVec/10)));
end
% Bounds is the vector of upper bounds for the bit error
% rate, for Eb/No values in EbNoVec.
Bounds = Errs*P;

```

- **Simulating Multiple Times to Collect Bit Error Rates**

You can efficiently vary the simulation parameters by using the `sim` function to run the simulation from the MATLAB command line. For example, the following code calculates the bit error rate at bit energy-to-noise ratios ranging from 1 dB to 4 dB, in increments of 0.5 dB. It collects all bit error rates from these simulations in the matrix `BERVec`. It also plots the bit error rates in a figure window along with the theoretical bounds computed in the preceding code fragment.

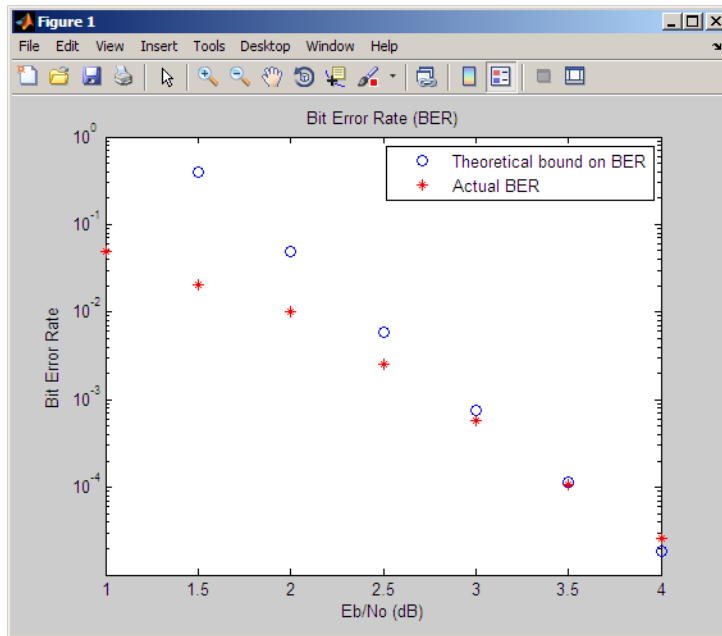
Note First open the model by clicking here in the MATLAB Help browser. Then execute these commands, which might take a few minutes.

```
% Plot theoretical bounds and set up figure.
figure;
semilogy(EbNoVec,Bounds,'bo',1,NaN,'r*');
xlabel('Eb/No (dB)'); ylabel('Bit Error Rate');
title('Bit Error Rate (BER)');
legend('Theoretical bound on BER','Actual BER');
axis([1 4 1e-5 1]);
hold on;

BERVec = [];
% Make the noise level variable.
set_param('doc_softdecision/AWGN Channel',...
    'EsNodB','EbNodB+10*log10(1/2)');
% Simulate multiple times.
for n = 1:length(EbNoVec)
    EbNodB = EbNoVec(n);
    sim('doc_softdecision',5000000);
    BERVec(n,:) = BER_Data;
    semilogy(EbNoVec(n),BERVec(n,1),'r*'); % Plot point.
    drawnow;
end
hold off;
```

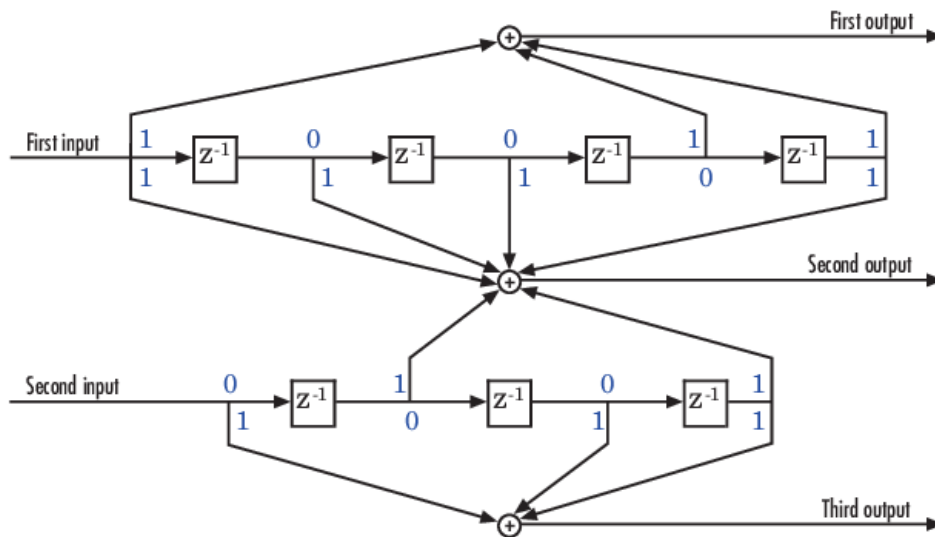
Note The estimate for P_b assumes that the decoder uses unquantized data, that is, an infinitely fine quantization. By contrast, the simulation in this example uses 8-level (3-bit) quantization. Because of this quantization, the simulated bit error rate is not quite as low as the bound when the signal-to-noise ratio is high.

The plot of bit error rate against signal-to-noise ratio follows. The locations of your actual BER points might vary because the simulation involves random numbers.



Design a Rate-2/3 Feedforward Encoder Using MATLAB

The example below uses the rate 2/3 feedforward encoder depicted in this schematic. The accompanying description explains how to determine the trellis structure parameter from a schematic of the encoder and then how to perform coding using this encoder.



Determining Coding Parameters

The `convenc` and `vitdec` functions can implement this code if their parameters have the appropriate values.

The encoder's constraint length is a vector of length 2 because the encoder has two inputs. The elements of this vector indicate the number of bits stored in each shift register, including the current input bits. Counting memory spaces in each shift register in the diagram and adding one for the current inputs leads to a constraint length of [5 4].

To determine the code generator parameter as a 2-by-3 matrix of octal numbers, use the element in the i th row and j th column to indicate how the i th input contributes to the j th output. For example, to compute the element in the second row and third column, the leftmost and two rightmost elements in the second shift register of the diagram feed into the sum that forms the third output. Capture this information as the binary number 1011, which is equivalent to the octal number 13. The full value of the code generator matrix is [23 35 0; 0 5 13].

To use the constraint length and code generator parameters in the `convenc` and `vitdec` functions, use the `poly2trellis` function to convert those parameters into a trellis structure. The command to do this is below.

```
trel = poly2trellis([5 4],[23 35 0;0 5 13]); % Define trellis.
```

Using the Encoder

Below is a script that uses this encoder.

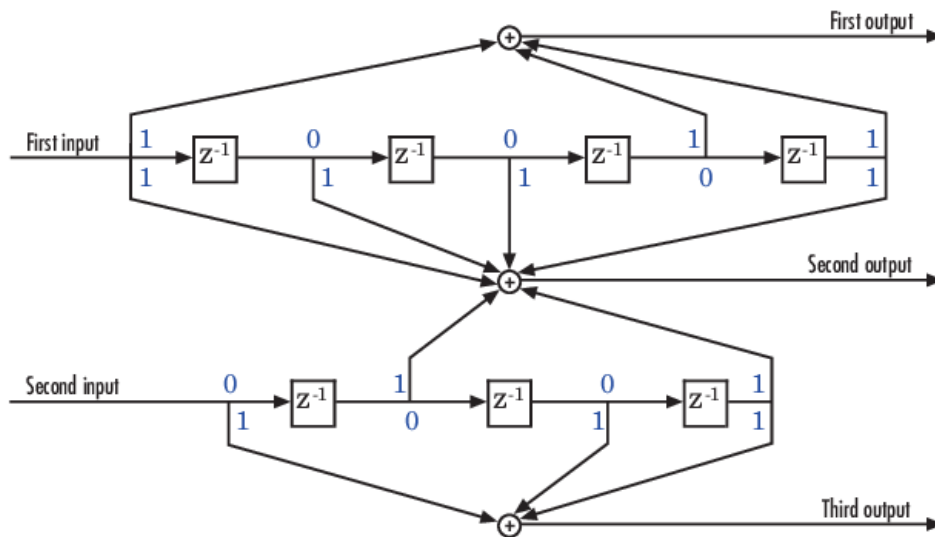
```
len = 1000;

msg = randi([0 1],2*len,1); % Random binary message of 2-bit symbols
trellis = poly2trellis([5 4],[23 35 0;0 5 13]); % Trellis
% Create a ConvolutionalEncoder System object
hConvEnc = comm.ConvolutionalEncoder(trellis);
% Create a ViterbiDecoder System object
hVitDec = comm.ViterbiDecoder(trellis, 'InputFormat', 'hard', ...
    'TracebackDepth', 34, 'TerminationMethod', 'Continuous');
% Create a ErrorRate Calculator System object. Since each symbol represents
% two bits, the receive delay for this object is twice the traceback length
% of the viterbi decoder.
hErrorCalc = comm.ErrorRate('ReceiveDelay', 68);
ber = zeros(3,1); % Store BER values
code = step(hConvEnc,msg); % Encode the message.
ncode = rem(code + randerr(3*len,1,[0 1;.96 .04]),2); % Add noise.
decoded = step(hVitDec, ncode); % Decode.
ber = step(hErrorCalc, msg, decoded);
```

convenc accepts a vector containing 2-bit symbols and produces a vector containing 3-bit symbols, while vitdec does the opposite. Also notice that biterr ignores the first 68 elements of decoded. That is, the decoding delay is 68, which is the number of bits per symbol (2) of the recovered message times the traceback depth value (34) in the vitdec function. The first 68 elements of decoded are 0s, while subsequent elements represent the decoded messages.

Design a Rate 2/3 Feedforward Encoder Using Simulink

This example uses the rate 2/3 feedforward convolutional encoder depicted in the following figure. The description explains how to determine the coding blocks' parameters from a schematic of a rate 2/3 feedforward encoder. This example also illustrates the use of the Error Rate Calculation block with a receive delay.



How to Determine Coding Parameters

The Convolutional Encoder and Viterbi Decoder blocks can implement this code if their parameters have the appropriate values.

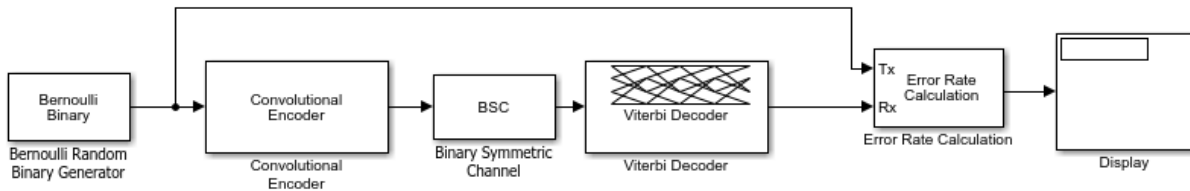
The encoder's constraint length is a vector of length 2 since the encoder has two inputs. The elements of this vector indicate the number of bits stored in each shift register, including the current input bits. Counting memory spaces in each shift register in the diagram and adding one for the current inputs leads to a constraint length of [5 4].

To determine the code generator parameter as a 2-by-3 matrix of octal numbers, use the element in the *i*th row and *j*th column to indicate how the *i*th input contributes to the *j*th output. For example, to compute the element in the second row and third column, notice that the leftmost and two rightmost elements in the second shift register of the diagram feed into the sum that forms the third output. Capture this information as the binary number 1011, which is equivalent to the octal number 13. The full value of the code generator matrix is [27 33 0; 0 5 13].

To use the constraint length and code generator parameters in the Convolutional Encoder and Viterbi Decoder blocks, use the `poly2trellis` function to convert those parameters into a trellis structure.

How to Simulate the Encoder

The following model simulates this encoder.



To open the completed model, enter `doc_convcoding` at the MATLAB command line. To build the model, gather and configure these blocks:

- Bernoulli Binary Generator, in the Comm Sources library
 - Set **Probability of a zero** to `.5`.
 - Set **Initial seed** to any positive integer scalar, preferably the output of the `randn` function.
 - Set **Sample time** to `.5`.
 - Check the **Frame-based outputs** check box.
 - Set **Samples per frame** to `2`.
- Convolutional Encoder
 - Set **Trellis structure** to `poly2trellis([5 4],[23 35 0; 0 5 13])`.
- Binary Symmetric Channel, in the Channels library
 - Set **Error probability** to `0.02`.
 - Set **Initial seed** to any positive integer scalar, preferably the output of the `randn` function.
 - Clear the **Output error vector** check box.
- Viterbi Decoder
 - Set **Trellis structure** to `poly2trellis([5 4],[23 35 0; 0 5 13])`.
 - Set **Decision type** to `Hard decision`.
- Error Rate Calculation, in the Comm Sinks library

- Set **Receive delay** to 68.
- Set **Output data** to Port.
- Check the **Stop simulation** check box.
- Set **Target number of errors** to 100.
- Display, in the Simulink Sinks library
 - Drag the bottom edge of the icon to make the display big enough for three entries.

Connect the blocks as in the figure. In the model window, select **Simulation > Model Configuration parameters** to open the Configuration Parameters dialog box. Set **Stop time** to inf.

Notes on the Model

The matrix size annotations appear on the connecting lines only if you click the **Display** menu and select **Signals & Ports > Signal Dimensions**. The encoder accepts a 2-by-1 column vector and produces a 3-by-1 column vector, while the decoder does the opposite. The **Samples per frame** parameter in the Bernoulli Binary Generator block is 2 because the block must generate a message word of length 2.

The **Receive delay** parameter in the Error Rate Calculation block is 68, which is the vector length (2) of the recovered message times the **Traceback depth** value (34) in the Viterbi Decoder block. If you examine the transmitted and received signals as matrices in the MATLAB workspace, you see that the first 34 rows of the recovered message consist of zeros, while subsequent rows are the decoded messages. Thus the delay in the received signal is 34 vectors of length 2, or 68 samples.

Running the model produces display output consisting of three numbers: the error rate, the total number of errors, and the total number of comparisons that the Error Rate Calculation block makes during the simulation. (The first two numbers vary depending on your **Initial seed** values in the Bernoulli Binary Generator and Binary Symmetric Channel blocks.) The simulation stops after 100 errors occur, because **Target number of errors** is set to 100 in the Error Rate Calculation block. The error rate is much less than 0.02, the **Error probability** in the Binary Symmetric Channel block.

Puncture a Convolutional Code Using MATLAB

This example processes a punctured convolutional code. It begins by generating 30,000 random bits and encoding them using a rate-3/4 convolutional encoder with a puncture pattern of [1 1 1 0 0 1]. The resulting vector contains 40,000 bits, which are mapped to values of -1 and 1 for transmission. The punctured code, `punctcode`, passes through an

additive white Gaussian noise channel. Then `vitdec` decodes the noisy vector using the 'unquant' decision type.

Finally, the example computes the bit error rate and the number of bit errors.

```
len = 30000; msg = randi([0 1], len, 1); % Random data
t = poly2trellis(7, [133 171]); % Define trellis.
% Create a ConvolutionalEncoder System object
hConvEnc = comm.ConvolutionalEncoder(t, ...
    'PuncturePatternSource', 'Property', ...
    'PuncturePattern', [1;1;1;0;0;1]);
% Create an AWGNChannel System object.
hChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)',...
    'SNR', 3);
% Create a ViterbiDecoder System object
hVitDec = comm.ViterbiDecoder(t, 'InputFormat', 'Unquantized', ...
    'TracebackDepth', 96, 'TerminationMethod', 'Truncated', ...
    'PuncturePatternSource', 'Property', ...
    'PuncturePattern', [1;1;1;0;0;1]);
% Create a ErrorRate Calculator System object.
hErrorCalc = comm.ErrorRate;
berP = zeros(3,1); berPE = berP; % Store BER values
punctcode = step(hConvEnc,msg); % Length is (2*len)*2/3.
tcode = 1-2*punctcode; % Map "0" bit to 1 and "1" bit to -1
hChan.SignalPower = (tcode'*tcode)/length(tcode);
ncode = step(hChan,tcode); % Add noise.

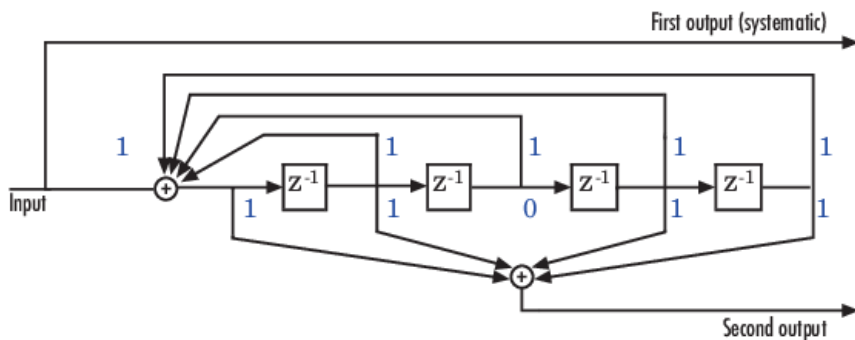
% Decode the punctured code
decoded = step(hVitDec,ncode); % Decode.
berP = step(hErrorCalc, msg, decoded);% Bit error rate
% Erase the least reliable 100 symbols, then decode
release(hVitDec); reset(hErrorCalc)
hVitDec.ErasuresInputPort = true;
[dummy idx] = sort(abs(ncode));
erasures = zeros(size(ncode)); erasures(idx(1:100)) = 1;
decoded = step(hVitDec,ncode, erasures); % Decode.
berPE = step(hErrorCalc, msg, decoded);% Bit error rate

fprintf('Number of errors with puncturing: %d\n', berP(2))
fprintf('Number of errors with puncturing and erasures: %d\n', berPE(2))
```

Implement a Systematic Encoder with Feedback Using Simulink

This section explains how to use the Convolutional Encoder block to implement a systematic encoder with feedback. A code is *systematic* if the actual message words

appear as part of the codewords. The following diagram shows an example of a systematic encoder.



To implement this encoder, set the **Trellis structure** parameter in the Convolutional Encoder block to `poly2trellis(5, [37 33], 37)`. This setting corresponds to

- Constraint length: 5
- Generator polynomial pair: [37 33]
- Feedback polynomial: 37

The feedback polynomial is represented by the binary vector [1 1 1 1 1], corresponding to the upper row of binary digits. These digits indicate connections from the outputs of the registers to the adder. The initial 1 corresponds to the input bit. The octal representation of the binary number 11111 is 37.

To implement a systematic code, set the first generator polynomial to be the same as the feedback polynomial in the **Trellis structure** parameter of the Convolutional Encoder block. In this example, both polynomials have the octal representation 37.

The second generator polynomial is represented by the binary vector [1 1 0 1 1], corresponding to the lower row of binary digits. The octal number corresponding to the binary number 11011 is 33.

For more information on setting the mask parameters for the Convolutional Encoder block, see Polynomial Description of a Convolutional Code on page 8-39.

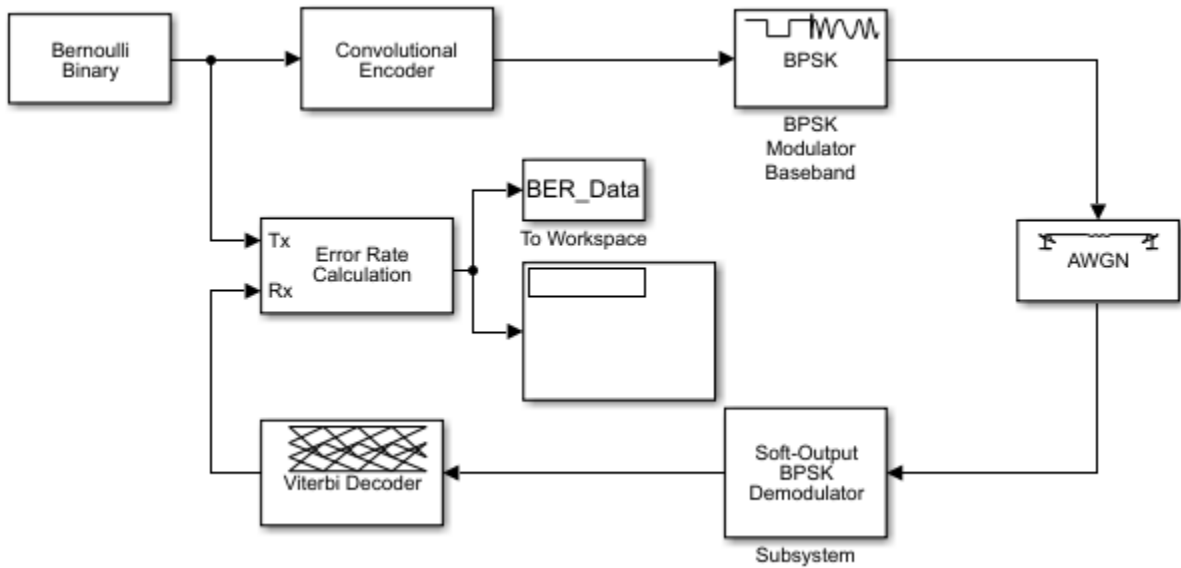
Soft-Decision Decoding

This example creates a rate 1/2 convolutional code. It uses a quantizer and the Viterbi Decoder block to perform soft-decision decoding. This description covers these topics:

- “Overview of the Simulation” on page 8-63
- “Defining the Convolutional Code” on page 8-64
- “Mapping the Received Data” on page 8-65
- “Decoding the Convolutional Code” on page 8-66
- “Delay in Received Data” on page 8-67
- “Comparing Simulation Results with Theoretical Results” on page 8-67

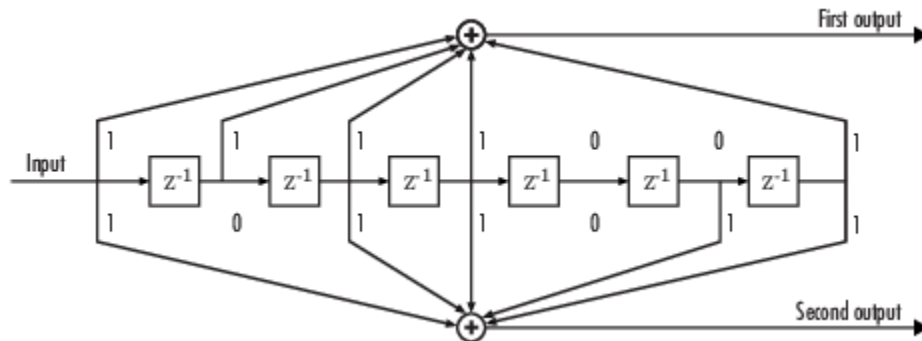
Overview of the Simulation

The model is in the following figure. To open the model, enter `doc_softdecision` at the MATLAB command line. The simulation creates a random binary message signal, encodes the message into a convolutional code, modulates the code using the binary phase shift keying (BPSK) technique, and adds white Gaussian noise to the modulated data in order to simulate a noisy channel. Then, the simulation prepares the received data for the decoding block and decodes. Finally, the simulation compares the decoded information with the original message signal in order to compute the bit error rate. The Convolutional encoder is configured as a rate 1/2 encoder. For every 2 bits, the encoder adds another 2 redundant bits. To accommodate this, and add the correct amount of noise, the **Eb/No (dB)** parameter of the AWGN block is in effect halved by subtracting $10 \cdot \log_{10}(2)$. The simulation ends after processing 100 bit errors or 10^7 message bits, whichever comes first.



Defining the Convolutional Code

The feedforward convolutional encoder in this example is depicted below.



The encoder's constraint length is a scalar since the encoder has one input. The value of the constraint length is the number of bits stored in the shift register, including the current input. There are six memory registers, and the current input is one bit. Thus the constraint length of the code is 7.

The code generator is a 1-by-2 matrix of octal numbers because the encoder has one input and two outputs. The first element in the matrix indicates which input values contribute to the first output, and the second element in the matrix indicates which input values contribute to the second output.

For example, the first output in the encoder diagram is the modulo-2 sum of the rightmost and the four leftmost elements in the diagram's array of input values. The seven-digit binary number 1111001 captures this information, and is equivalent to the octal number 171. The octal number 171 thus becomes the first entry of the code generator matrix. Here, each triplet of bits uses the leftmost bit as the most significant bit. The second output corresponds to the binary number 1011011, which is equivalent to the octal number 133. The code generator is therefore [171 133].

The **Trellis structure** parameter in the Convolutional Encoder block tells the block which code to use when processing data. In this case, the `poly2trellis` function, in Communications Toolbox, converts the constraint length and the pair of octal numbers into a valid trellis structure.

While the message data entering the Convolutional Encoder block is a scalar bit stream, the encoded data leaving the block is a stream of binary vectors of length 2.

Mapping the Received Data

The received data, that is, the output of the AWGN Channel block, consists of complex numbers that are close to -1 and 1. In order to reconstruct the original binary message, the receiver part of the model must decode the convolutional code. The Viterbi Decoder block in this model expects its input data to be integers between 0 and 7. The demodulator, a custom subsystem in this model, transforms the received data into a format that the Viterbi Decoder block can interpret properly. More specifically, the demodulator subsystem

- Converts the received data signal to a real signal by removing its imaginary part. It is reasonable to assume that the imaginary part of the received data does not contain essential information, because the imaginary part of the transmitted data is zero (ignoring small roundoff errors) and because the channel noise is not very powerful.
- Normalizes the received data by dividing by the standard deviation of the noise estimate and then multiplying by -1.
- Quantizes the normalized data using three bits.

The combination of this mapping and the Viterbi Decoder block's decision mapping reverses the BPSK modulation that the BPSK Modulator Baseband block performs on the

transmitting side of this model. To examine the demodulator subsystem in more detail, double-click the icon labeled Soft-Output BPSK Demodulator.

Decoding the Convolutional Code

After the received data is properly mapped to length-2 vectors of 3-bit decision values, the Viterbi Decoder block decodes it. The block uses a soft-decision algorithm with 2^3 different input values because the **Decision type** parameter is `Soft Decision` and the **Number of soft decision bits** parameter is 3.

Soft-Decision Interpretation of Data

When the **Decision type** parameter is set to `Soft Decision`, the Viterbi Decoder block requires input values between 0 and 2^b-1 , where b is the **Number of soft decision bits** parameter. The block interprets 0 as the most confident decision that the codeword bit is a 0 and interprets 2^b-1 as the most confident decision that the codeword bit is a 1. The values in between these extremes represent less confident decisions. The following table lists the interpretations of the eight possible input values for this example.

Decision Value	Interpretation
0	Most confident 0
1	Second most confident 0
2	Third most confident 0
3	Least confident 0
4	Least confident 1
5	Third most confident 1
6	Second most confident 1
7	Most confident 1

Traceback and Decoding Delay

The **Traceback depth** parameter in the Viterbi Decoder block represents the length of the decoding delay. Typical values for a traceback depth are about five or six times the constraint length, which would be 35 or 42 in this example. However, some hardware implementations offer options of 48 and 96. This example chooses 48 because that is closer to the targets (35 and 42) than 96 is.

Delay in Received Data

The **Receive delay** parameter of the Error Rate Calculation block is nonzero because a given message bit and its corresponding recovered bit are separated in time by a nonzero amount of simulation time. The **Receive delay** parameter tells the block which elements of its input signals to compare when checking for errors.

In this case, the Receive delay value is equal to the Traceback depth value (48).

Comparing Simulation Results with Theoretical Results

This section describes how to compare the bit error rate in this simulation with the bit error rate that would theoretically result from unquantized decoding. The process includes a few steps, described in these sections:

Computing Theoretical Bounds for the Bit Error Rate

To calculate theoretical bounds for the bit error rate P_b of the convolutional code in this model, you can use this estimate based on unquantized-decision decoding:

$$P_b < \sum_{d=f}^{\infty} c_d P_d$$

In this estimate, c_d is the sum of bit errors for error events of distance d , and f is the free distance of the code. The quantity P_d is the pairwise error probability, given by

$$P_d = \frac{1}{2} \operatorname{erfc} \left(\sqrt{dR \frac{E_b}{N_0}} \right)$$

where R is the code rate of 1/2, and erfc is the MATLAB complementary error function, defined by

$$\operatorname{erfc}(x) = \frac{2}{\sqrt{\pi}} \int_x^{\infty} e^{-t^2} dt$$

Values for the coefficients c_d and the free distance f are in published articles such as Frenger, P., P. Orten, and Ottosson, "Convolutional Codes with Optimum Distance Spectrum," *IEEE Communications* vol. 3, pp. 317-319, November 1999. The free distance for this code is $f = 10$.

The following commands calculate the values of P_b for E_b/N_0 values in the range from 1 to 4, in increments of 0.5:

```
EbNoVec = [1:0.5:4.0];
R = 1/2;
% Errs is the vector of sums of bit errors for
% error events at distance d, for d from 10 to 29.
Errs = [36 0 211 0 1404 0 11633 0 77433 0 502690 0,...
        3322763 0 21292910 0 134365911 0 843425871 0];
% P is the matrix of pairwise error probabilities, for
% Eb/No values in EbNoVec and d from 10 to 29.
P = zeros(20,7); % Initialize.
for d = 10:29
    P(d-9,:) = (1/2)*erfc(sqrt(d*R*10.^(EbNoVec/10)));
end
% Bounds is the vector of upper bounds for the bit error
% rate, for Eb/No values in EbNoVec.
Bounds = Errs*P;
```

Simulating Multiple Times to Collect Bit Error Rates

You can efficiently vary the simulation parameters by using the `sim` function to run the simulation from the MATLAB command line. For example, the following code calculates the bit error rate at bit energy-to-noise ratios ranging from 1 dB to 4 dB, in increments of 0.5 dB. It collects all bit error rates from these simulations in the matrix `BERVec`. It also plots the bit error rates in a figure window along with the theoretical bounds computed in the preceding code fragment.

Note First open the model by clicking [here](#) in the MATLAB Help browser. Then execute these commands, which might take a few minutes.

```
% Plot theoretical bounds and set up figure.
figure;
semilogy(EbNoVec, Bounds, 'bo', 1, NaN, 'r*');
xlabel('Eb/No (dB)'); ylabel('Bit Error Rate');
title('Bit Error Rate (BER)');
legend('Theoretical bound on BER', 'Actual BER');
axis([1 4 1e-5 1]);
hold on;

BERVec = [];
% Make the noise level variable.
set_param('doc_softdecision/AWGN Channel',...
    'EsNodB', 'EbNodB+10*log10(1/2)');
% Simulate multiple times.
```

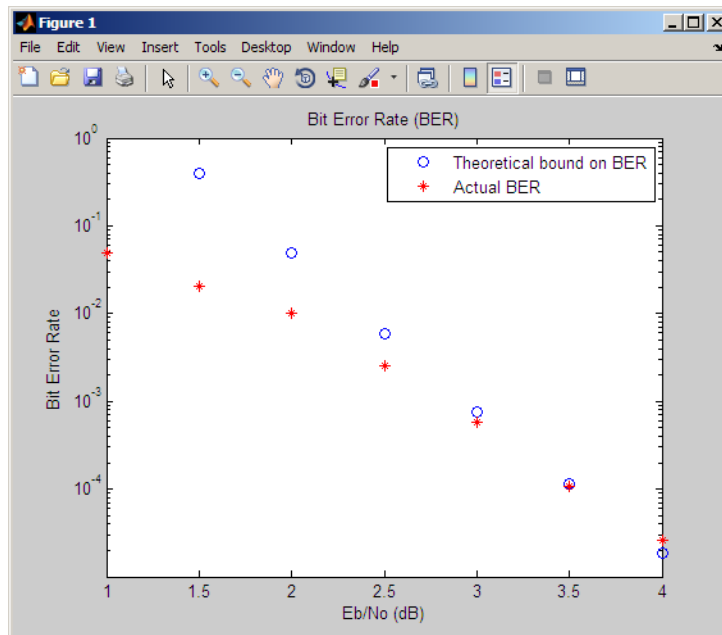
```

for n = 1:length(EbNoVec)
    EbNodB = EbNoVec(n);
    sim('doc_softdecision',5000000);
    BERVec(n,:) = BER_Data;
    semilogy(EbNoVec(n),BERVec(n,1),'r*'); % Plot point.
    drawnow;
end
hold off;

```

Note The estimate for P_b assumes that the decoder uses unquantized data, that is, an infinitely fine quantization. By contrast, the simulation in this example uses 8-level (3-bit) quantization. Because of this quantization, the simulated bit error rate is not quite as low as the bound when the signal-to-noise ratio is high.

The plot of bit error rate against signal-to-noise ratio follows. The locations of your actual BER points might vary because the simulation involves random numbers.



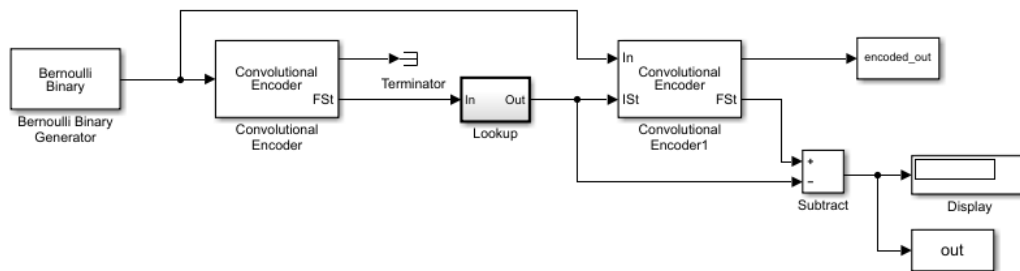
Tailbiting Encoding Using Feedback Encoders

This example demonstrates Tailbiting encoding using feedback encoders. For feedback encoders, the ending state depends on the entire block of data. To accomplish tailbiting, you must calculate for a given information vector (of N bits), the initial state, that leads to the same ending state after the block of data is encoded.

This is achieved in two steps:

- The first step is to determine the zero-state response for a given block of data. The encoder starts in the all-zeros state. The whole block of data is input and the output bits are ignored. After N bits, the encoder is in a state $X_N^{[zs]}$. From this state, we calculate the corresponding initial state X_0 and initialize the encoder with X_0 .
- The second step is the actual encoding. The encoder starts with the initial state X_0 , the data block is input and a valid codeword is output which conforms to the same state boundary condition.

Refer to [8] for a theoretical calculation of the initial state X_0 from $X_N^{[zs]}$ using state-space formulation. This is a one-time calculation which depends on the block length and in practice could be implemented as a look-up table. Here we determine this mapping table by simulating all possible entries for a chosen trellis and block length.



To open the model, type `doc_mtailbiting_wfeedback` at the MATLAB command line.

```
function mapStValues = getMapping(blkLen, trellis)
% The function returns the mapping value for the given block
length and trellis to be used for determining the initial
state from the zero-state response.
```

```
% All possible combinations of the mappings
mapStValuesTab = perms(0:trellis.numStates-1);
```

```

% Loop over all the combinations of the mapping entries:
for i = 1:length(mapStValuesTab)
    mapStValues = mapStValuesTab(i,:);

% Model parameterized for the Block length
sim('mtailbiting_wfeedback');

% Check the boundary condition for each run
% if ending and starting states match, choose that mapping set
if unique(out)==0
    return
end
end

```

Selecting the returned `mapStValues` for the **Table data** parameter of the `Direct Lookup Table (n-D)` block in the Lookup subsystem will perform tailbiting encoding for the chosen block length and trellis.

Selected Bibliography for Convolutional Coding

- [1] Clark, George C. Jr., and J. Bibb Cain, *Error-Correction Coding for Digital Communications*, New York, Plenum Press, 1981.
- [2] Gitlin, Richard D., Jeremiah F. Hayes, and Stephen B. Weinstein, *Data Communications Principles*, New York, Plenum Press, 1992.
- [3] Frenger, P., P. Orten, and T. Ottosson, "Convolution Codes with Optimum Distance Spectrum," *IEEE Communications Letters*, vol. 3, pp. 317-319, November 1999.

Linear Block Codes

- "Represent Words for Linear Block Codes" on page 8-71
- "Configure Parameters for Linear Block Codes" on page 8-75
- "Create and Decode Linear Block Codes" on page 8-79

Represent Words for Linear Block Codes

The cyclic, Hamming, and generic linear block code functionality in this product offers you multiple ways to organize bits in messages or codewords. These topics explain the available formats:

- “Use MATLAB to Create Messages and Codewords in Binary Vector Format” on page 8-72
- “Use MATLAB to Create Messages and Codewords in Binary Matrix Format” on page 8-73
- “Use MATLAB to Create Messages and Codewords in Decimal Vector Format” on page 8-74

To learn how to represent words for BCH or Reed-Solomon codes, see “Represent Words for BCH Codes” on page 8-89 or “Represent Words for Reed-Solomon Codes” on page 8-96.

Use MATLAB to Create Messages and Codewords in Binary Vector Format

Your messages and codewords can take the form of vectors containing 0s and 1s. For example, messages and codes might look like `msg` and `code` in the lines below.

```
n = 6; k = 4; % Set codeword length and message length
% for a [6,4] code.
msg = [1 0 0 1 1 0 1 0 1 0 1 1]'; % Message is a binary column.
code = encode(msg,n,k,'cyclic'); % Code will be a binary column.
msg'
code'
```

The output is below.

```
ans =
```

```
Columns 1 through 5
```

```
1      0      0      1      1
```

```
Columns 6 through 10
```

```
0      1      0      1      0
```

```
Columns 11 through 12
```

```
1      1
```

```
ans =
```

```
Columns 1 through 5
```

```

    1      1      1      0      0
Columns 6 through 10
    1      0      0      1      0
Columns 11 through 15
    1      0      0      1      1
Columns 16 through 18
    0      1      1

```

In this example, `msg` consists of 12 entries, which are interpreted as three 4-digit (because $k = 4$) messages. The resulting vector `code` comprises three 6-digit (because $n = 6$) codewords, which are concatenated to form a vector of length 18. The parity bits are at the beginning of each codeword.

Use MATLAB to Create Messages and Codewords in Binary Matrix Format

You can organize coding information so as to emphasize the grouping of digits into messages and codewords. If you use this approach, each message or codeword occupies a row in a binary matrix. The example below illustrates this approach by listing each 4-bit message on a distinct row in `msg` and each 6-bit codeword on a distinct row in `code`.

```

n = 6; k = 4; % Set codeword length and message length.
msg = [1 0 0 1; 1 0 1 0; 1 0 1 1]; % Message is a binary matrix.
code = encode(msg,n,k,'cyclic'); % Code will be a binary matrix.
msg
code

```

The output is below.

```

msg =
    1    0    0    1
    1    0    1    0
    1    0    1    1

code =
    1    1    1    0    0    1

```

0	0	1	0	1	0
0	1	1	0	1	1

Note In the binary matrix format, the message matrix must have k columns. The corresponding code matrix has n columns. The parity bits are at the beginning of each row.

Use MATLAB to Create Messages and Codewords in Decimal Vector Format

Your messages and codewords can take the form of vectors containing integers. Each element of the vector gives the decimal representation of the bits in one message or one codeword.

Note If 2^n or 2^k is very large, you should use the default binary format instead of the decimal format. This is because the function uses a binary format internally, while the roundoff error associated with converting many bits to large decimal numbers and back might be substantial.

Note When you use the decimal vector format, `encode` expects the *leftmost* bit to be the least significant bit.

The syntax for the `encode` command must mention the decimal format explicitly, as in the example below. Notice that `/decimal` is appended to the fourth argument in the `encode` command.

```
n = 6; k = 4; % Set codeword length and message length.
msg = [9;5;13]; % Message is a decimal column vector.
% Code will be a decimal vector.
code = encode(msg,n,k,'cyclic/decimal')
```

The output is below.

```
code =
    39
    20
    54
```

Note The three examples above used cyclic coding. The formats for messages and codes are similar for Hamming and generic linear block codes.

Configure Parameters for Linear Block Codes

This subsection describes the items that you might need in order to process $[n,k]$ cyclic, Hamming, and generic linear block codes. The table below lists the items and the coding techniques for which they are most relevant.

Parameters Used in Block Coding Techniques

Parameter	Block Coding Technique
“Generator Matrix” on page 8-75	Generic linear block
“Parity-Check Matrix” on page 8-75	Generic linear block
“Generator Polynomial” on page 8-77	Cyclic
“Decoding Table” on page 8-77	Generic linear block, Hamming

Generator Matrix

The process of encoding a message into an $[n,k]$ linear block code is determined by a k -by- n generator matrix G . Specifically, the 1-by- k message vector v is encoded into the 1-by- n codeword vector vG . If G has the form $[I_k \ P]$ or $[P \ I_k]$, where P is some k -by- $(n-k)$ matrix and I_k is the k -by- k identity matrix, G is said to be in *standard form*. (Some authors, e.g., Clark and Cain [2], use the first standard form, while others, e.g., Lin and Costello [3], use the second.) Most functions in this toolbox assume that a generator matrix is in standard form when you use it as an input argument.

Some examples of generator matrices are in the next section, “Parity-Check Matrix” on page 8-75.

Parity-Check Matrix

Decoding an $[n,k]$ linear block code requires an $(n-k)$ -by- n parity-check matrix H . It satisfies $GH^{\text{tr}} = 0 \pmod{2}$, where H^{tr} denotes the matrix transpose of H , G is the code's generator matrix, and this zero matrix is k -by- $(n-k)$. If $G = [I_k \ P]$ then $H = [-P^{\text{tr}} \ I_{n-k}]$. Most functions in this product assume that a parity-check matrix is in standard form when you use it as an input argument.

The table below summarizes the standard forms of the generator and parity-check matrices for an $[n,k]$ binary linear block code.

Type of Matrix	Standard Form	Dimensions
Generator	$[I_k \ P]$ or $[P \ I_k]$	k-by-n
Parity-check	$[-P' \ I_{n-k}]$ or $[I_{n-k} \ -P']$	(n-k)-by-n

I_k is the identity matrix of size k and the ' symbol indicates matrix transpose. (For *binary* codes, the minus signs in the parity-check form listed above are irrelevant; that is, $-1 = 1$ in the binary field.)

Examples

In the command below, `parmat` is a parity-check matrix and `genmat` is a generator matrix for a Hamming code in which $[n,k] = [2^3-1, n-3] = [7,4]$. `genmat` has the standard form $[P \ I_k]$.

```
[parmat,genmat] = hamngen(3)
parmat =

     1     0     0     1     0     1     1
     0     1     0     1     1     1     0
     0     0     1     0     1     1     1

genmat =

     1     1     0     1     0     0     0
     0     1     1     0     1     0     0
     1     1     1     0     0     1     0
     1     0     1     0     0     0     1
```

The next example finds parity-check and generator matrices for a $[7,3]$ cyclic code. The `cyclpoly` function is mentioned below in "Generator Polynomial" on page 8-77.

```
genpoly = cyclpoly(7,3);
[parmat,genmat] = cyclgen(7,genpoly)
parmat =

     1     0     0     0     1     1     0
     0     1     0     0     0     1     1
     0     0     1     0     1     1     1
     0     0     0     1     1     0     1

genmat =

     1     0     1     1     1     0     0
```

$$\begin{array}{ccccccc} 1 & 1 & 1 & 0 & 0 & 1 & 0 \\ 0 & 1 & 1 & 1 & 0 & 0 & 1 \end{array}$$

The example below converts a generator matrix for a [5,3] linear block code into the corresponding parity-check matrix.

```
genmat = [1 0 0 1 0; 0 1 0 1 1; 0 0 1 0 1];
parmat = gen2par(genmat)
```

```
parmat =
```

$$\begin{array}{ccccc} 1 & 1 & 0 & 1 & 0 \\ 0 & 1 & 1 & 0 & 1 \end{array}$$

The same function `gen2par` can also convert a parity-check matrix into a generator matrix.

Generator Polynomial

Cyclic codes have algebraic properties that allow a polynomial to determine the coding process completely. This so-called *generator polynomial* is a degree-(n-k) divisor of the polynomial x^n-1 . Van Lint [5] explains how a generator polynomial determines a cyclic code.

The `cyclpoly` function produces generator polynomials for cyclic codes. `cyclpoly` represents a generator polynomial using a row vector that lists the polynomial's coefficients in order of *ascending* powers of the variable. For example, the command

```
genpoly = cyclpoly(7,3)
```

```
genpoly =
```

$$1 \quad 0 \quad 1 \quad 1 \quad 1$$

finds that one valid generator polynomial for a [7,3] cyclic code is $1 + x^2 + x^3 + x^4$.

Decoding Table

A decoding table tells a decoder how to correct errors that might have corrupted the code during transmission. Hamming codes can correct any single-symbol error in any codeword. Other codes can correct, or partially correct, errors that corrupt more than one symbol in a given codeword.

This toolbox represents a decoding table as a matrix with n columns and $2^{(n-k)}$ rows. Each row gives a correction vector for one received codeword vector. A Hamming

decoding table has $n+1$ rows. The `syndtable` function generates a decoding table for a given parity-check matrix.

Use a Decoding Table in MATLAB

The script below shows how to use a Hamming decoding table to correct an error in a received message. The `hammgen` function produces the parity-check matrix, while the `syndtable` function produces the decoding table. The transpose of the parity-check matrix is multiplied on the left by the received codeword, yielding the *syndrome*. The decoding table helps determine the correction vector. The corrected codeword is the sum (modulo 2) of the correction vector and the received codeword.

```
% Use a [7,4] Hamming code.
m = 3; n = 2^m-1; k = n-m;
parmat = hammgen(m); % Produce parity-check matrix.
trt = syndtable(parmat); % Produce decoding table.
recd = [1 0 0 1 1 1 1] % Suppose this is the received vector.
syndrome = rem(recd * parmat',2);
syndrome_de = bi2de(syndrome,'left-msb'); % Convert to decimal.
disp(['Syndrome = ',num2str(syndrome_de),...
      ' (decimal), ',num2str(syndrome),' (binary)'])
corrvect = trt(1+syndrome_de,:) % Correction vector
% Now compute the corrected codeword.
correctedcode = rem(corrvect+recd,2)
```

The output is below.

```
recd =
      1      0      0      1      1      1      1

Syndrome = 3 (decimal), 0 1 1 (binary)

corrvect =
      0      0      0      0      1      0      0

correctedcode =
      1      0      0      1      0      1      1
```

Create and Decode Linear Block Codes

The functions for encoding and decoding cyclic, Hamming, and generic linear block codes are `encode` and `decode`. This section discusses how to use these functions to create and decode generic linear block on page 8-79 codes, cyclic on page 8-80 codes, and Hamming on page 8-81 codes.

Generic Linear Block Codes

Encoding a message using a generic linear block code requires a generator matrix. If you have defined variables `msg`, `n`, `k`, and `genmat`, either of the commands

```
code = encode(msg,n,k,'linear',genmat);
code = encode(msg,n,k,'linear/decimal',genmat);
```

encodes the information in `msg` using the $[n,k]$ code that the generator matrix `genmat` determines. The `/decimal` option, suitable when 2^n and 2^k are not very large, indicates that `msg` contains nonnegative decimal integers rather than their binary representations. See “Represent Words for Linear Block Codes” on page 8-71 or the reference page for `encode` for a description of the formats of `msg` and `code`.

Decoding the code requires the generator matrix and possibly a decoding table. If you have defined variables `code`, `n`, `k`, `genmat`, and possibly also `trt`, then the commands

```
newmsg = decode(code,n,k,'linear',genmat);
newmsg = decode(code,n,k,'linear/decimal',genmat);
newmsg = decode(code,n,k,'linear',genmat,trt);
newmsg = decode(code,n,k,'linear/decimal',genmat,trt);
```

decode the information in `code`, using the $[n,k]$ code that the generator matrix `genmat` determines. `decode` also corrects errors according to instructions in the decoding table that `trt` represents.

Example: Generic Linear Block Coding

The example below encodes a message, artificially adds some noise, decodes the noisy code, and keeps track of errors that the decoder detects along the way. Because the decoding table contains only zeros, the decoder does not correct any errors.

```
n = 4; k = 2;
genmat = [[1 1; 1 0], eye(2)]; % Generator matrix
msg = [0 1; 0 0; 1 0]; % Three messages, two bits each
% Create three codewords, four bits each.
code = encode(msg,n,k,'linear',genmat);
```

```
noisycode = rem(code + randerr(3,4,[0 1;.7 .3]),2); % Add noise.
trt = zeros(2^(n-k),n); % No correction of errors
% Decode, keeping track of all detected errors.
[newmsg,err] = decode(noisycode,n,k,'linear',genmat,trt);
err_words = find(err~=0) % Find out which words had errors.
```

The output indicates that errors occurred in the first and second words. Your results might vary because this example uses random numbers as errors.

```
err_words =
     1
     2
```

Cyclic Codes

A cyclic code is a linear block code with the property that cyclic shifts of a codeword (expressed as a series of bits) are also codewords. An alternative characterization of cyclic codes is based on its generator polynomial, as mentioned in “Generator Polynomial” on page 8-77 and discussed in [5].

Encoding a message using a cyclic code requires a generator polynomial. If you have defined variables `msg`, `n`, `k`, and `genpoly`, then either of the commands

```
code = encode(msg,n,k,'cyclic',genpoly);
code = encode(msg,n,k,'cyclic/decimal',genpoly);
```

encodes the information in `msg` using the $[n,k]$ code determined by the generator polynomial `genpoly`. `genpoly` is an optional argument for `encode`. The default generator polynomial is `cyclpoly(n,k)`. The `/decimal` option, suitable when 2^n and 2^k are not very large, indicates that `msg` contains nonnegative decimal integers rather than their binary representations. See “Represent Words for Linear Block Codes” on page 8-71 or the reference page for `encode` for a description of the formats of `msg` and `code`.

Decoding the code requires the generator polynomial and possibly a decoding table. If you have defined variables `code`, `n`, `k`, `genpoly`, and `trt`, then the commands

```
newmsg = decode(code,n,k,'cyclic',genpoly);
newmsg = decode(code,n,k,'cyclic/decimal',genpoly);
newmsg = decode(code,n,k,'cyclic',genpoly,trt);
newmsg = decode(code,n,k,'cyclic/decimal',genpoly,trt);
```

decode the information in `code`, using the $[n,k]$ code that the generator matrix `genmat` determines. `decode` also corrects errors according to instructions in the decoding table

that `trt` represents. `genpoly` is an optional argument in the first two syntaxes above. The default generator polynomial is `cyclpoly(n,k)`.

Example

You can modify the example in “Generic Linear Block Codes” on page 8-79 so that it uses the cyclic coding technique, instead of the linear block code with the generator matrix `genmat`. Make the changes listed below:

- Replace the second line by


```
genpoly = [1 0 1]; % generator poly is 1 + x^2
```
- In the fifth and ninth lines (`encode` and `decode` commands), replace `genmat` by `genpoly` and replace `'linear'` by `'cyclic'`.

Another example of encoding and decoding a cyclic code is on the reference page for `encode`.

Hamming Codes

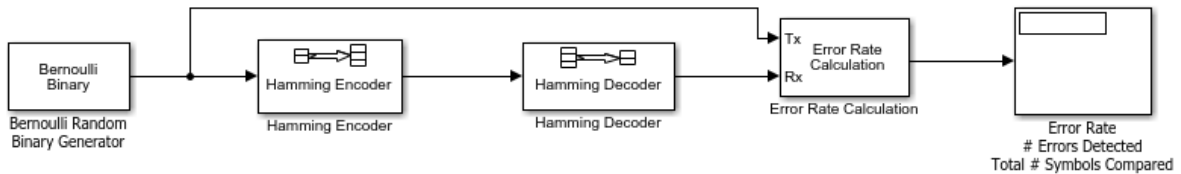
The reference pages for `encode` and `decode` contain examples of encoding and decoding Hamming codes. Also, the section “Decoding Table” on page 8-77 illustrates error correction in a Hamming code.

Hamming Codes

- “Create a Hamming Code in Binary Format Using Simulink” on page 8-81
- “Reduce the Error Rate Using a Hamming Code” on page 8-82

Create a Hamming Code in Binary Format Using Simulink

This example shows very simply how to use an encoder and decoder. It illustrates the appropriate vector lengths of the code and message signals for the coding blocks. Because the Error Rate Calculation block accepts only scalars or frame-based column vectors as the transmitted and received signals, this example uses frame-based column vectors throughout. (It thus avoids having to change signal attributes using a block such as Convert 1-D to 2-D.)



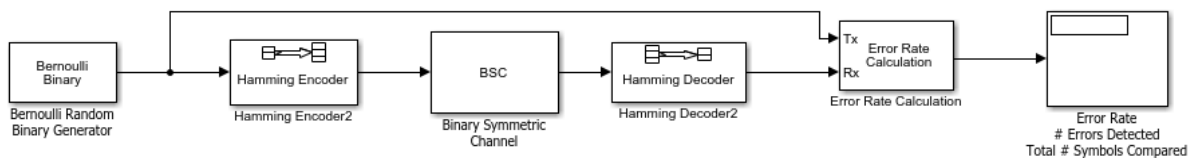
Open this model by entering `doc_hamming` at the MATLAB command line. To build the model, gather and configure these blocks:

- Bernoulli Binary Generator, in the Comm Sources library
 - Set **Probability of a zero** to .5.
 - Set **Initial seed** to any positive integer scalar, preferably the output of the `randn` function.
 - Check the **Frame-based outputs** check box.
 - Set **Samples per frame** to 4.
- Hamming Encoder, with default parameter values
- Hamming Decoder, with default parameter values
- Error Rate Calculation, in the Comm Sinks library, with default parameter values

Connect the blocks as in the preceding figure. Click the **Display** menu and select **Signals & Ports > Signal Dimensions**. After updating the diagram if necessary (**Simulation > Update Diagram**), the connector lines show relevant signal attributes. The connector lines are double lines to indicate frame-based signals, and the annotations next to the lines show that the signals are column vectors of appropriate sizes.

Reduce the Error Rate Using a Hamming Code

This section describes how to reduce the error rate by adding an error-correcting code. This figure shows model that uses a Hamming code.

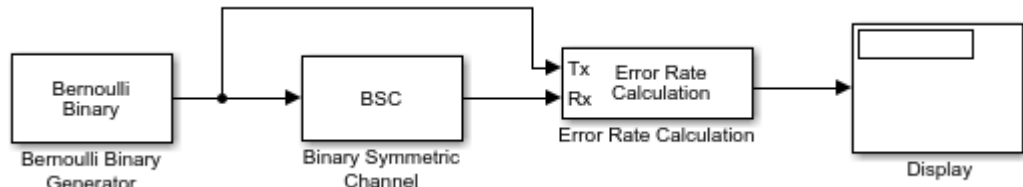


To open a complete version of the model, type `doc_hamming` at the MATLAB prompt.

Building the Hamming Code Model

You can build the Hamming code model by following these steps:

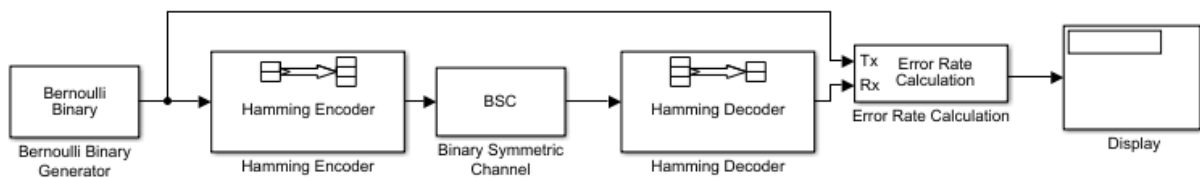
- 1 Type `doc_channel` at the MATLAB prompt to open the channel noise model.



Then save the model as `my_hamming` in the folder where you keep your work files.

- 2 From the Simulink Library Browser drag the Hamming Encoder and Hamming Decoder blocks from the Error Detection and Correction/Block sublibrary into the model window.
- 3 Click the right border of the model and drag it to the right to widen the model window.
- 4 Move the Binary Symmetric Channel, Error Rate Calculation, and Display blocks to the right by clicking and dragging.
- 5 Create enough space between the Bernoulli Binary Generator and Binary Symmetric Channel blocks to fit the Hamming Encoder between them.
- 6 Click and drag the Hamming Encoder block on top of the line between the Bernoulli Binary Generator block and the Binary Symmetric Channel block, to the right of the branch point, as shown in the following figure. Then release the mouse button. The Hamming Encoder block should automatically connect to the line from the Bernoulli Binary Generator block to the Binary Symmetric Channel block.
- 7 Move blocks again to create enough space between the Binary Symmetric Channel and the Error Rate Calculation blocks to fit the Hamming Decoder between them.
- 8 Click and drag the Hamming Decoder block on top of the line between the Binary Symmetric Channel block and the Error Rate Calculation block.

The model should now resemble this figure.



Using the Hamming Encoder and Decoder Blocks

The Hamming Encoder block encodes the data before it is sent through the channel. The default code is the [7,4] Hamming code, which encodes message words of length 4 into codewords of length 7. As a result, the block converts frames of size 4 into frames of size 7. The code can correct one error in each transmitted codeword.

For an $[n,k]$ code, the input to the Hamming Encoder block must consist of vectors of size k . In this example, $k = 4$.

The Hamming Decoder block decodes the data after it is sent through the channel. If at most one error is created in a codeword by the channel, the block decodes the word correctly. However, if more than one error occurs, the Hamming Decoder block might decode incorrectly.

To learn more about block coding features, see Block Codes on page 8-20.

Setting Parameters in the Hamming Code Model

Double-click the Bernoulli Binary Generator block and make the following changes to the parameter settings in the block's dialog box, as shown in the following figure:

- 1 Set **Samples per frame** to 4. This converts the output of the block into frames of size 4, in order to meet the input requirement of the Hamming Encoder Block. See “Sample-Based and Frame-Based Processing” on page 2-4 for more information about frames.

Note Many blocks, such as the Hamming Encoder block, require their input to be a vector of a specific size. If you connect a source block, such as the Bernoulli Binary Generator block, to one of these blocks, set **Samples per frame** to the required value. For this model the **Samples per frame** parameter of the Bernoulli Binary Generator block must be a multiple of the **Message Length K** parameter of the Hamming Encoder block.

Labeling the Display Block

You can change the label that appears below a block to make it more informative. For example, to change the label below the Display block to 'Error Rate Display', first select the label with the mouse. This causes a box to appear around the text. Enter the changes to the text in the box.

Running the Hamming Code Model

To run the model, select **Simulation > Start**. The model terminates after 100 errors occur. The error rate, displayed in the top window of the Display block, is approximately .001. You get slightly different results if you change the **Initial seed** parameters in the model or run a simulation for a different length of time.

You expect an error rate of approximately .001 for the following reason: The probability of two or more errors occurring in a codeword of length 7 is

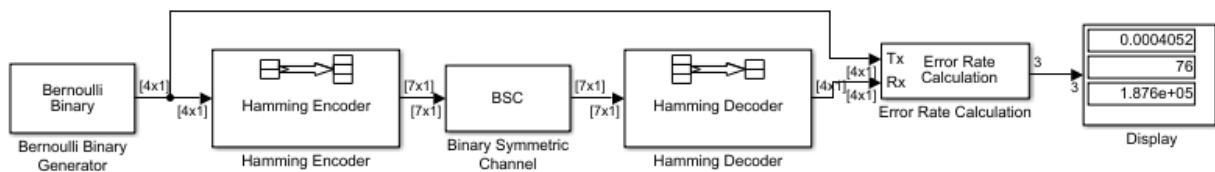
$$1 - (0.99)^7 - 7(0.99)^6(0.01) = 0.002$$

If the codewords with two or more errors are decoded randomly, you expect about half the bits in the decoded message words to be incorrect. This indicates that .001 is a reasonable value for the bit error rate.

To obtain a lower error rate for the same probability of error, try using a Hamming code with larger parameters. To do this, change the parameters **Codeword length** and **Message length** in the Hamming Encoder and Hamming Decoder block dialog boxes. You also have to make the appropriate changes to the parameters of the Bernoulli Binary Generator block and the Binary Symmetric Channel block.

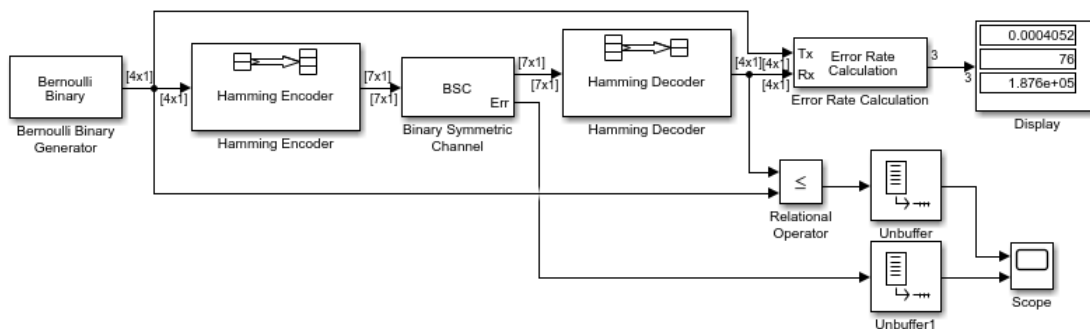
Displaying Frame Sizes

You can display the sizes of data frames in different parts of the model by clicking the **Display** menu and selecting **Signals & Ports > Signal Dimensions**. The line leading out of the Bernoulli Binary Generator block is labeled [4x1], indicating that its output consists of column vectors of size 4. Because the Hamming Encoder block uses a [7,4] code, it converts frames of size 4 into frames of size 7, so its output is labeled [7x1].



Adding a Scope to the Model

To display the channel errors produced by the Binary Symmetric Channel block, add a Scope block to the model. This is a good way to see whether your model is functioning correctly. The example shown in the following figure shows where to insert the Scope block into the model.



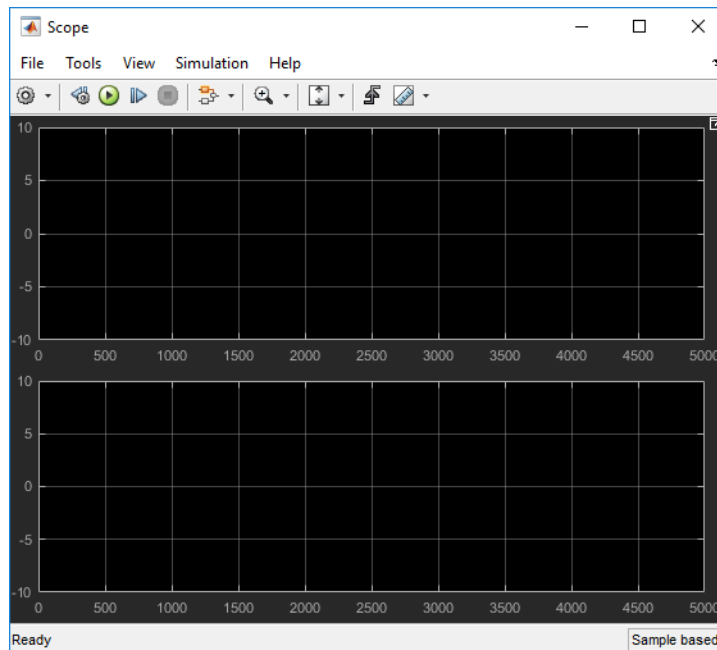
To build this model from the one shown in the figure “Reduce the Error Rate Using a Hamming Code” on page 8-82, follow these steps:

- 1 Drag the following blocks from the Simulink Library Browser into the model window:
 - Relational Operator block, from the Simulink Logic and Bit Operations library
 - Scope block, from the Simulink Sinks library
 - Two copies of the Unbuffer block, from the Buffers sublibrary of the Signal Management library in DSP System Toolbox
- 2 Double-click the Binary Symmetric Channel block to open its dialog box, and select **Output error vector**. This creates a second output port for the block, which carries the error vector.
- 3 Double-click the Scope block, under **View > Configuration Properties ...**, set **Number of input ports** to 2. Select **Layout** and highlight two blocks vertically. Click **OK**.
- 4 Connect the blocks as shown in the preceding figure.

Setting Parameters in the Expanded Model

Make the following changes to the parameters for the blocks you added to the model.

- **Error Rate Calculation Block** - Double-click the Error Rate Calculation block and clear the box next to **Stop simulation** in the block's dialog box.
- **Scope Block** - The Scope block displays the channel errors and uncorrected errors. To configure the block,
 - 1 Double-click the Scope block, select '**View > Configuration Properties ...**'.
 - 2 Select the **Time** tab and set **Time span** to 5000.
 - 3 Select the **Logging** tab and set **Limit data points to last** to 30000.
 - 4 Click **OK**.
 - 5 The scope should now appear as shown.

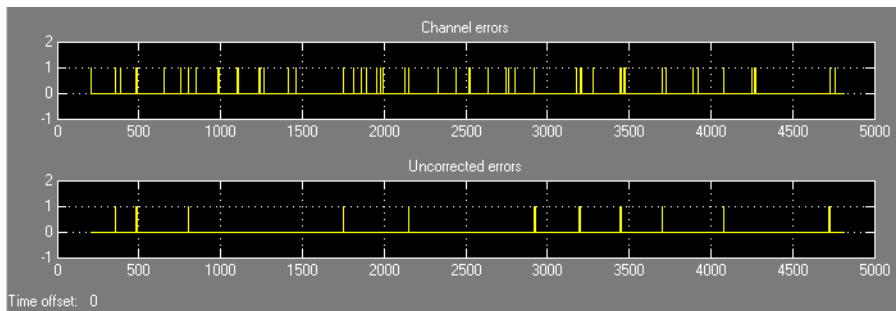


- 6 To configure the axes, follow these steps:
 - a Right-click the vertical axis at the left side of the upper scope.
 - b In the context menu, select **Axes properties**.
 - c Set **Y-limits (Minimum)** to -1.
 - d Set **Y-limits (Maximum)** to 2, and click **OK**.

- e Repeat the same steps for the vertical axis of the lower scope.
 - f Widen the scope window until it is roughly three times as wide as it is high. You can do this by clicking the right border of the window and dragging the border to the right, while pressing the left-mouse button.
- **Relational Operator** - Set **Relational Operator** to $\sim =$ in the block's dialog box. The Relational Operator block compares the transmitted signal, coming from the Bernoulli Random Generator block, with the received signal, coming from the Hamming Decoder block. The block outputs a 0 when the two signals agree and a 1 when they disagree.

Observing Channel Errors with the Scope

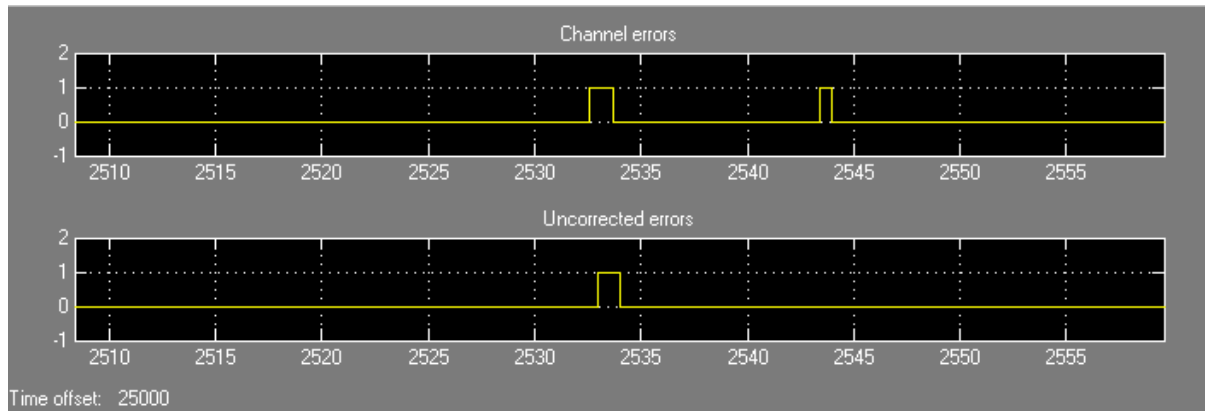
When you run the model, the scope displays the error data. At the end of each 5000 time steps, the scope appears as shown in this figure. The scope then clears the displayed data and displays the next 5000 data points.



The upper scope shows the channel errors generated by the Binary Symmetric Channel block. The lower scope shows errors that are not corrected by channel coding.

Click the **Stop** button on the toolbar at the top of the model window to stop the scope.

You can see individual errors by zooming in on the scope. First click the middle magnifying glass button at the top left of the Scope window. Then click one of the lines in the lower scope. This zooms in horizontally on the line. Continue clicking the lines in the lower scope until the horizontal scale is fine enough to detect individual errors. A typical example of what you might see is shown in the figure below.



The wider rectangular pulse in the middle of the upper scope represents two 1s. These two errors, which occur in a single codeword, are not corrected. This accounts for the uncorrected errors in the lower scope. The narrower rectangular pulse to the right of the upper scope represents a single error, which is corrected.

When you are done observing the errors, select **Simulation > Stop**.

“Export Data to MATLAB” on page 1-3 explains how to send the error data to the MATLAB workspace for more detailed analysis.

BCH Codes

- “Represent Words for BCH Codes” on page 8-89
- “Parameters for BCH Codes” on page 8-90
- “Create and Decode BCH Codes” on page 8-90
- “Algorithms for BCH and RS Errors-only Decoding” on page 8-93

Represent Words for BCH Codes

A message for an $[n,k]$ BCH code must be a k -column binary Galois field array. The code that corresponds to that message is an n -column binary Galois field array. Each row of these Galois field arrays represents one word.

The example below illustrates how to represent words for a $[15, 11]$ BCH code.

```
msg = [1 0 0 1 0; 1 0 1 1 1]; % Messages in a Galois array
obj = comm.BCHEncoder;
```

```
c1 = step(obj, msg(1,:));  
c2 = step(obj, msg(2,:));  
cbch = [c1 c2].'
```

The output is

Columns 1 through 5

1	0	0	1	0
1	0	1	1	1

Columns 6 through 10

0	0	1	1	1
0	0	0	0	1

Columns 11 through 15

1	0	1	0	1
0	1	0	0	1

Parameters for BCH Codes

BCH codes use special values of n and k :

- n , the codeword length, is an integer of the form $2^m - 1$ for some integer $m > 2$.
- k , the message length, is a positive integer less than n . However, only some positive integers less than n are valid choices for k . See the BCH Encoder block reference page for a list of some valid values of k corresponding to values of n up to 511.

Create and Decode BCH Codes

The BCH Encoder and BCH Decoder System objects create and decode BCH codes, using the data described in “Represent Words for BCH Codes” on page 8-89 and “Parameters for BCH Codes” on page 8-90.

The topics are

- “Example: BCH Coding Syntaxes” on page 8-90
- “Detect and Correct Errors in a BCH Code Using MATLAB” on page 8-91

Example: BCH Coding Syntaxes

The example below illustrates how to encode and decode data using a [15, 5] BCH code.


```

n = 15; k = 5; % Codeword length and message length
msg = randi([0 1],4*k,1); % Four random binary messages

% Simplest syntax for encoding
enc = comm.BCHEncoder(n,k);
dec = comm.BCHDecoder(n,k);
c1 = step(enc,msg); % BCH encoding
d1 = step(dec,c1); % BCH decoding

% Check that the decoding worked correctly.
chk = isequal(d1,msg)

% The following code shows how to perform the encoding and decoding
% operations if one chooses to prepend the parity symbols.

% Steps for converting encoded data with appended parity symbols
% to encoded data with prepended parity symbols
c11 = reshape(c1, n, []);
c12 = circshift(c11,n-k);
c1_prepend = c12(:); % BCH encoded data with prepended parity symbols

% Steps for converting encoded data with prepended parity symbols
% to encoded data with appended parity symbols prior to decoding
c21 = reshape(c1_prepend, n, []);
c22 = circshift(c21,k);
c1_append = c22(:); % BCH encoded data with appended parity symbols

% Check that the prepend-to-append conversion worked correctly.
d1_append = step(dec,c1_append);
chk = isequal(msg,d1_append)

```

The output is below.

```

chk =
     1

```

Detect and Correct Errors in a BCH Code Using MATLAB

The following example illustrates the decoding results for a corrupted code. The example encodes some data, introduces errors in each codeword, and attempts to decode the noisy code using the BCH Decoder System object.

```

n = 15; k = 5; % Codeword length and message length
[gp,t] = bchgenpoly(n,k); % t is error-correction capability.

```

```
nw = 4; % Number of words to process
msgw = randi([0 1], nw*k, 1); % Random k-symbol messages
enc = comm.BCHEncoder(n,k,gp);
dec = comm.BCHDecoder(n,k,gp);
c = step(enc, msgw); % Encode the data.
noise = randerr(nw,n,t); % t errors per codeword
noisy = noise';
noisy = noisy(:);
cnoisy = mod(c + noisy,2); % Add noise to the code.
[dc, nerrs] = step(dec, cnoisy); % Decode cnoisy.

% Check that the decoding worked correctly.
chk2 = isequal(dc,msgw)
nerrs % Find out how many errors have been corrected.
```

Notice that the array of noise values contains binary values, and that the addition operation `c + noise` takes place in the Galois field $GF(2)$ because `c` is a Galois field array in $GF(2)$.

The output from the example is below. The nonzero value of `ans` indicates that the decoder was able to correct the corrupted codewords and recover the original message. The values in the vector `nerrs` indicate that the decoder corrected `t` errors in each codeword.

```
chk2 =
     1

nerrs =
     3
     3
     3
     3
```

Excessive Noise in BCH Codewords

In the previous example, the `BCH Decoder System` object corrected all the errors. However, each BCH code has a finite error-correction capability. To learn more about how the `BCH Decoder System` object behaves when the noise is excessive, see the analogous discussion for Reed-Solomon codes in “Excessive Noise in Reed-Solomon Codewords” on page 8-101.

Algorithms for BCH and RS Errors-only Decoding

Overview

The errors-only decoding algorithm used for BCH and RS codes can be described by the following steps (sections 5.3.2, 5.4, and 5.6 in [2]).

- 1 Calculate the first $2t$ terms of the infinite degree syndrome polynomial, $S(z)$.
- 2 If those $2t$ terms of $S(z)$ are all equal to 0, then the code has no errors, no correction needs to be performed, and the decoding algorithm ends.
- 3 If one or more terms of $S(z)$ are nonzero, calculate the error locator polynomial, $\Lambda(z)$, via the Berlekamp algorithm.
- 4 Calculate the error evaluator polynomial, $\Omega(z)$, via

$$\Lambda(z)S(z) = \Omega(z) \bmod z^{2t}$$

- 5 Correct an error in the codeword according to

$$e_{i_m} = \frac{\Omega(\alpha^{-i_m})}{\Lambda'(\alpha^{-i_m})}$$

where e_{i_m} is the error magnitude in the i_m th position in the codeword, m is a value less than the error-correcting capability of the code, $\Omega(z)$ is the error magnitude polynomial, $\Lambda'(z)$ is the formal derivative [5] of the error locator polynomial, $\Lambda(z)$, and α is the primitive element of the Galois field of the code.

Further description of several of the steps is given in the following sections.

Syndrome Calculation

For narrow-sense codes, the $2t$ terms of $S(z)$ are calculated by evaluating the received codeword at successive powers of α (the field's primitive element) from 0 to $2t-1$. In other words, if we assume one-based indexing of codewords $C(z)$ and the syndrome polynomial $S(z)$, and that codewords are of the form $[c_1 \ c_1 \ \dots \ c_N]$, then each term S_i of $S(z)$ is given as

$$S_i = \sum_{j=1}^N c_j \alpha^{N-1-i}$$

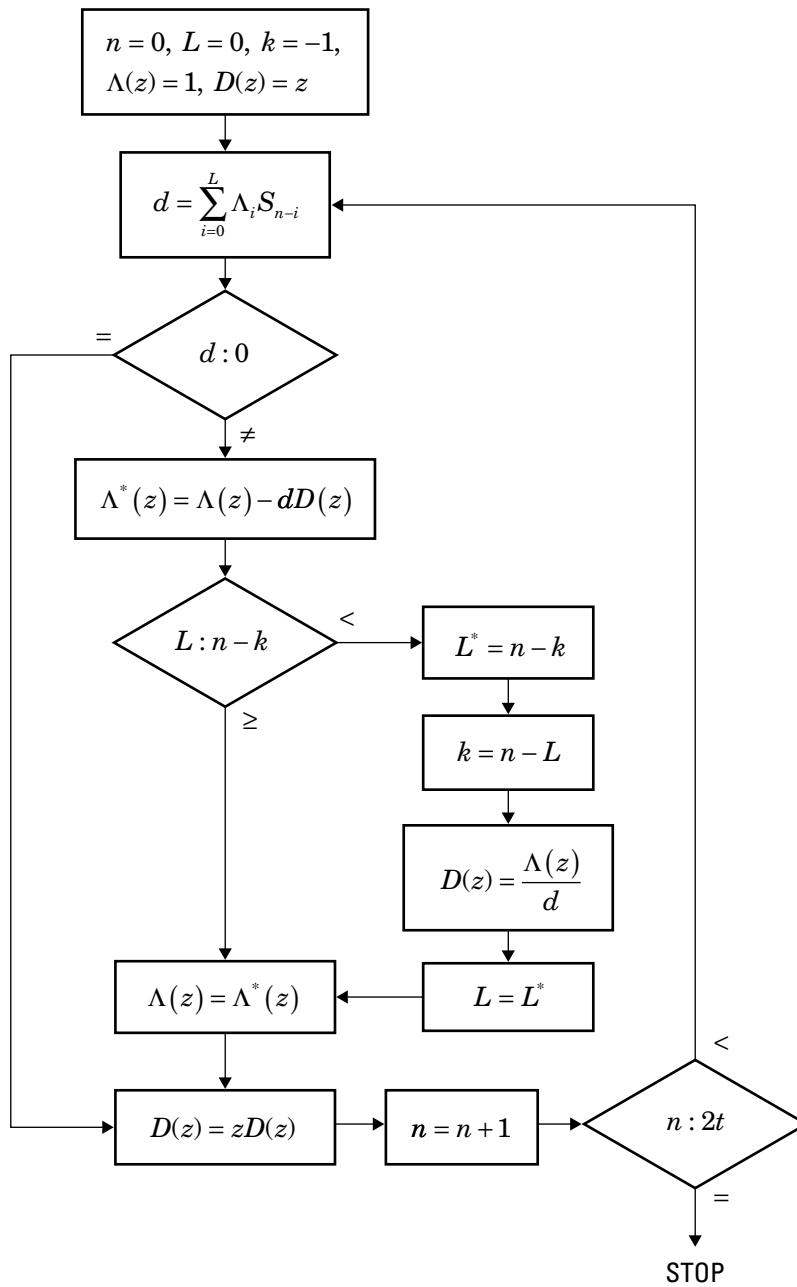
Error Locator Polynomial Calculation

The error locator polynomial, $\Lambda(z)$, is found using the Berlekamp algorithm. A complete description of this algorithm is found in [2], but we summarize the algorithm as follows.

We define the following variables.

Variable	Description
n	Iterator variable
k	Iterator variable
L	Length of the feedback register used to generate the first $2t$ terms of $S(z)$
$D(z)$	Correction polynomial
d	Discrepancy

The following diagram shows the iterative procedure (i.e., the Berlekamp algorithm) used to find $\Lambda(z)$.



Error Evaluator Polynomial Calculation

The error evaluator polynomial, $\Omega(z)$, is simply the convolution of $\Lambda(z)$ and $S(z)$.

Reed-Solomon Codes

- “Represent Words for Reed-Solomon Codes” on page 8-96
- “Parameters for Reed-Solomon Codes” on page 8-97
- “Create and Decode Reed-Solomon Codes” on page 8-98
- “Find a Generator Polynomial” on page 8-102
- “Reed Solomon Examples with Shortening, Puncturing, and Erasures” on page 8-104

Represent Words for Reed-Solomon Codes

This toolbox supports Reed-Solomon codes that use m-bit symbols instead of bits. A message for an $[n,k]$ Reed-Solomon code must be a k-column Galois field array in the field $GF(2^m)$. Each array entry must be an integer between 0 and 2^m-1 . The code corresponding to that message is an n-column Galois field array in $GF(2^m)$. The codeword length n must be between 3 and 2^m-1 .

Note For information about Galois field arrays and how to create them, see “Representing Elements of Galois Fields” on page 8-109 or the reference page for the `gf` function.

The example below illustrates how to represent words for a $[7,3]$ Reed-Solomon code.

```
n = 7; k = 3; % Codeword length and message length
m = 3; % Number of bits in each symbol
msg = [1 6 4; 0 4 3]; % Message is a Galois array.
obj = comm.RSEncoder(n, k);
c1 = step(obj, msg(1,:));
c2 = step(obj, msg(2,:));
c = [c1 c2].'
```

The output is

C =

1	6	4	4	3	6	3
0	4	3	3	7	4	7

Parameters for Reed-Solomon Codes

This section describes several integers related to Reed-Solomon codes and discusses how to find generator polynomials on page 8-97.

Allowable Values of Integer Parameters

The table below summarizes the meanings and allowable values of some positive integer quantities related to Reed-Solomon codes as supported in this toolbox. The quantities n and k are input parameters for Reed-Solomon functions in this toolbox.

Symbol	Meaning	Value or Range
m	Number of bits per symbol	Integer between 3 and 16
n	Number of symbols per codeword	Integer between 3 and 2^m-1
k	Number of symbols per message	Positive integer less than n , such that $n-k$ is even
t	Error-correction capability of the code	$(n-k)/2$

Generator Polynomial

The `rsgenpoly` function produces generator polynomials for Reed-Solomon codes. `rsgenpoly` represents a generator polynomial using a Galois row vector that lists the polynomial's coefficients in order of *descending* powers of the variable. If each symbol has m bits, the Galois row vector is in the field $GF(2^m)$. For example, the command

```
r = rsgenpoly(15,13)
```

```
r = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
```

```
Array elements =
```

```
1    6    8
```

finds that one generator polynomial for a [15,13] Reed-Solomon code is $X^2 + (A^2 + A)X + (A^3)$, where A is a root of the default primitive polynomial for $GF(16)$.

Algebraic Expression for Generator Polynomials

The generator polynomials that `rsgenpoly` produces have the form $(X - A^b)(X - A^{b+1})\dots(X - A^{b+2t-1})$, where b is an integer, A is a root of the primitive polynomial for the Galois field,

and t is $(n-k)/2$. The default value of b is 1. The output from `rsgenpoly` is the result of multiplying the factors and collecting like powers of X . The example below checks this formula for the case of a $[15,13]$ Reed-Solomon code, using $b = 1$.

```
n = 15;
a = gf(2, log2(n+1)); % Root of primitive polynomial
f1 = [1 a]; f2 = [1 a^2]; % Factors that form generator polynomial
f = conv(f1, f2) % Generator polynomial, same as r above.
```

Create and Decode Reed-Solomon Codes

The `RS Encoder` and `RS Decoder System` objects create and decode Reed-Solomon codes, using the data described in “Represent Words for Reed-Solomon Codes” on page 8-96 and “Parameters for Reed-Solomon Codes” on page 8-97.

This section illustrates how to use the `RS Encoder` and `RS Decoder System` objects. The topics are

- “Reed-Solomon Coding Syntaxes in MATLAB” on page 8-98
- “Detect and Correct Errors in a Reed-Solomon Code Using MATLAB” on page 8-100
- “Excessive Noise in Reed-Solomon Codewords” on page 8-101
- “Create Shortened Reed-Solomon Codes” on page 8-101

Reed-Solomon Coding Syntaxes in MATLAB

The example below illustrates multiple ways to encode and decode data using a $[15,13]$ Reed-Solomon code. The example shows that you can

- Vary the generator polynomial for the code, using `rsgenpoly` to produce a different generator polynomial.
- Vary the primitive polynomial for the Galois field that contains the symbols, using an input argument in `gf`.
- Vary the position of the parity symbols within the codewords, choosing either the end (default) or beginning.

This example also shows that corresponding syntaxes of the `RS Encoder` and `RS Decoder System` objects use the same input arguments, except for the first input argument.

```
m = 4; % Number of bits in each symbol
n = 2^m-1; k = 13; % Codeword length and message length
```



```

msg = randi([0 m-1],4*k,1); % Four random integer messages

% Simplest syntax for encoding
hEnc = comm.RSEncoder(n,k);
hDec = comm.RSDecoder(n,k);
c1 = step(hEnc, msg);
d1 = step(hDec, c1);

% Vary the generator polynomial for the code.
release(hEnc), release(hDec)
hEnc.GeneratorPolynomialSource = 'Property';
hDec.GeneratorPolynomialSource = 'Property';
hEnc.GeneratorPolynomial       = rsgenpoly(n,k,19,2);
hDec.GeneratorPolynomial       = rsgenpoly(n,k,19,2);
c2 = step(hEnc, msg);
d2 = step(hDec, c2);

% Vary the primitive polynomial for GF(16).
release(hEnc), release(hDec)
hEnc.PrimitivePolynomialSource = 'Property';
hDec.PrimitivePolynomialSource = 'Property';
hEnc.GeneratorPolynomialSource = 'Auto';
hDec.GeneratorPolynomialSource = 'Auto';
hEnc.PrimitivePolynomial       = [1 1 0 0 1];
hDec.PrimitivePolynomial       = [1 1 0 0 1];
c3 = step(hEnc, msg);
d3 = step(hDec, c3);

% Check that the decoding worked correctly.
chk = isequal(d1,msg) & isequal(d2,msg) & isequal(d3,msg)

% The following code shows how to perform the encoding and decoding
% operations if one chooses to prepend the parity symbols.

% Steps for converting encoded data with appended parity symbols
% to encoded data with prepended parity symbols
c31 = reshape(c3, n, []);
c32 = circshift(c31,n-k);
c3_prepend = c32(:); % RS encoded data with prepended parity symbols

% Steps for converting encoded data with prepended parity symbols
% to encoded data with appended parity symbols prior to decoding
c34 = reshape(c3_prepend, n, []);

```

```
c35 = circshift(c34,k);
c3_append = c35(:); % RS encoded data with appended parity symbols

% Check that the prepend-to-append conversion worked correctly.
d3_append = step(hDec,c3_append);
chk = isequal(msg,d3_append)
```

The output is

```
chk =
```

```
1
```

Detect and Correct Errors in a Reed-Solomon Code Using MATLAB

The example below illustrates the decoding results for a corrupted code. The example encodes some data, introduces errors in each codeword, and attempts to decode the noisy code using the RS Decoder System object.

```
m = 3; % Number of bits per symbol
n = 2^m-1; k = 3; % Codeword length and message length
t = (n-k)/2; % Error-correction capability of the code
nw = 4; % Number of words to process
msgw = randi([0 n],nw*k,1); % Random k-symbol messages
hEnc = comm.RSEncoder(n,k);
hDec = comm.RSDecoder(n,k);
c = step(hEnc, msgw); % Encode the data.
noise = (1+randi([0 n-1],nw,n)).*randerr(nw,n,t); % t errors per codeword
noisy = noise';
noisy = noisy(:);
cnoisy = gf(c,m) + noisy; % Add noise to the code under gf(m) arithmetic.
[dc nerrs] = step(hDec, cnoisy.x); % Decode the noisy code.
% Check that the decoding worked correctly.
isequal(dc,msgw)
nerrs % Find out how many errors hDec corrected.
```

The array of noise values contains integers between 1 and 2^m , and the addition operation $c + \text{noise}$ takes place in the Galois field $GF(2^m)$ because c is a Galois field array in $GF(2^m)$.

The output from the example is below. The nonzero value of `ans` indicates that the decoder was able to correct the corrupted codewords and recover the original message. The values in the vector `nerrs` indicates that the decoder corrected `t` errors in each codeword.

```
ans =
     1
nerrs =
     2
     2
     2
     2
```

Excessive Noise in Reed-Solomon Codewords

In the previous example, RS Encoder System object corrected all of the errors. However, each Reed-Solomon code has a finite error-correction capability. If the noise is so great that the corrupted codeword is too far in Hamming distance from the correct codeword, that means either

- The corrupted codeword is close to a valid codeword *other than* the correct codeword. The decoder returns the message that corresponds to the other codeword.
- The corrupted codeword is not close enough to any codeword for successful decoding. This situation is called a *decoding failure*. The decoder removes the symbols in parity positions from the corrupted codeword and returns the remaining symbols.

In both cases, the decoder returns the wrong message. However, you can tell when a decoding failure occurs because RS Decoder System object also returns a value of -1 in its second output.

To examine cases in which codewords are too noisy for successful decoding, change the previous example so that the definition of noise is

```
noise = (1+randi([0 n-1],nw,n)).*randerr(nw,n,t+1); % t+1 errors/row
```

Create Shortened Reed-Solomon Codes

Every Reed-Solomon encoder uses a codeword length that equals $2^m - 1$ for an integer m . A shortened Reed-Solomon code is one in which the codeword length is not $2^m - 1$. A shortened $[n, k]$ Reed-Solomon code implicitly uses an $[n_1, k_1]$ encoder, where

- $n_1 = 2^m - 1$, where m is the number of bits per symbol
- $k_1 = k + (n_1 - n)$

The RS Encoder System object supports shortened codes using the same syntaxes it uses for nonshortened codes. You do not need to indicate explicitly that you want to use a shortened code.

```
hEnc = comm.RSEncoder(7,5);  
ordinarycode = step(hEnc,[1 1 1 1 1]');  
hEnc = comm.RSEncoder(5,3);  
shortenedcode = step(hEnc,[1 1 1 ]');
```

How the RS Encoder System Object Creates a Shortened Code

When creating a shortened code, the RS Encoder System object performs these steps:

- Pads each message by prepending zeros
- Encodes each padded message using a Reed-Solomon encoder having an allowable codeword length and the desired error-correction capability
- Removes the extra zeros from the nonparity symbols of each codeword

The following example illustrates this process.

```
n = 12; k = 8; % Lengths for the shortened code  
m = ceil(log2(n+1)); % Number of bits per symbol  
msg = randi([0 2^m-1],3*k,1); % Random array of 3 k-symbol words  
hEnc = comm.RSEncoder(n,k);  
code = step(hEnc, msg); % Create a shortened code.  
  
% Do the shortening manually, just to show how it works.  
n_pad = 2^m-1; % Codeword length in the actual encoder  
k_pad = k+(n_pad-n); % Messageword length in the actual encoder  
hEnc = comm.RSEncoder(n_pad,k_pad);  
mw = reshape(msg,k,[]); % Each column vector represents a messageword  
msg_pad = [zeros(n_pad-n,3); mw]; % Prepend zeros to each word.  
msg_pad = msg_pad(:);  
code_pad = step(hEnc,msg_pad); % Encode padded words.  
cw = reshape(code_pad,2^m-1,[]); % Each column vector represents a codeword  
code_eqv = cw(n_pad-n+1:n_pad,:); % Remove extra zeros.  
code_eqv = code_eqv(:);  
ck = isequal(code_eqv,code); % Returns true (1).
```

Find a Generator Polynomial

To find a generator polynomial for a cyclic, BCH, or Reed-Solomon code, use the `cyclpoly`, `bchgenpoly`, or `rsgenpoly` function, respectively. The commands

```

genpolyCyclic = cyclpoly(15,5) % 1+X^5+X^10
genpolyBCH = bchgenpoly(15,5) % x^10+x^8+x^5+x^4+x^2+x+1
genpolyRS = rsgenpoly(15,5)

```

find generator polynomials for block codes of different types. The output is below.

```

genpolyCyclic =
    1    0    0    0    0    1    0    0    0    0    1

genpolyBCH = GF(2) array.
Array elements =
    1    0    1    0    0    1    1    0    1    1    1

genpolyRS = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
Array elements =
    1    4    8    10    12    9    4    2    12    2    7

```

The formats of these outputs vary:

- `cyclpoly` represents a generator polynomial using an integer row vector that lists the polynomial's coefficients in order of *ascending* powers of the variable.
- `bchgenpoly` and `rsgenpoly` represent a generator polynomial using a Galois row vector that lists the polynomial's coefficients in order of *descending* powers of the variable.
- `rsgenpoly` uses coefficients in a Galois field other than the binary field GF(2). For more information on the meaning of these coefficients, see “How Integers Correspond to Galois Field Elements” on page 8-111 and “Polynomials over Galois Fields” on page 8-130.

Nonuniqueness of Generator Polynomials

Some pairs of message length and codeword length do not uniquely determine the generator polynomial. The syntaxes for functions in the example above also include options for retrieving generator polynomials that satisfy certain constraints that you specify. See the functions' reference pages for details about syntax options.

Algebraic Expression for Generator Polynomials

The generator polynomials produced by `bchgenpoly` and `rsgenpoly` have the form $(X - A^b)(X - A^{b+1})\dots(X - A^{b+2t-1})$, where A is a primitive element for an appropriate Galois field,

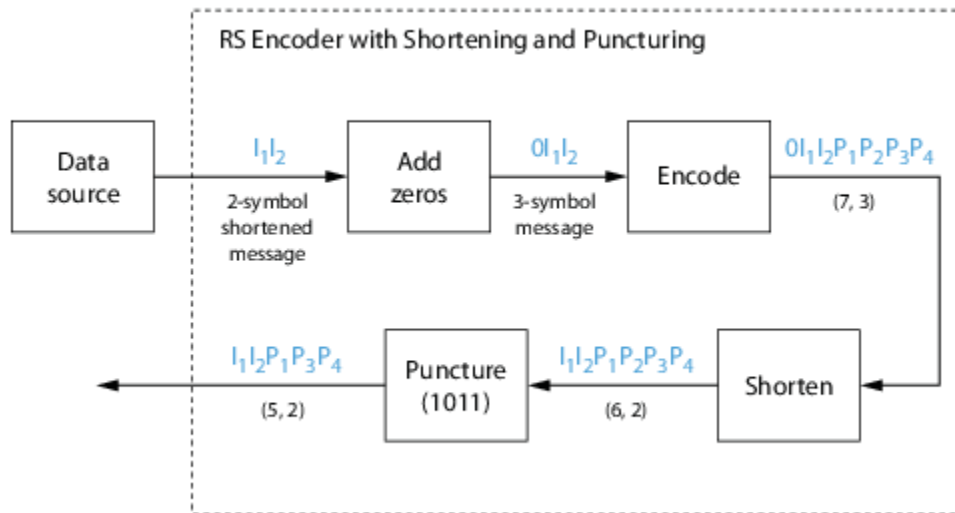
and b and t are integers. See the functions' reference pages for more information about this expression.

Reed Solomon Examples with Shortening, Puncturing, and Erasures

In this section, a representative example of Reed Solomon coding with shortening, puncturing, and erasures is built with increasing complexity of error correction.

Encoder Example with Shortening and Puncturing

The following figure shows a representative example of a (7,3) Reed Solomon encoder with shortening and puncturing.



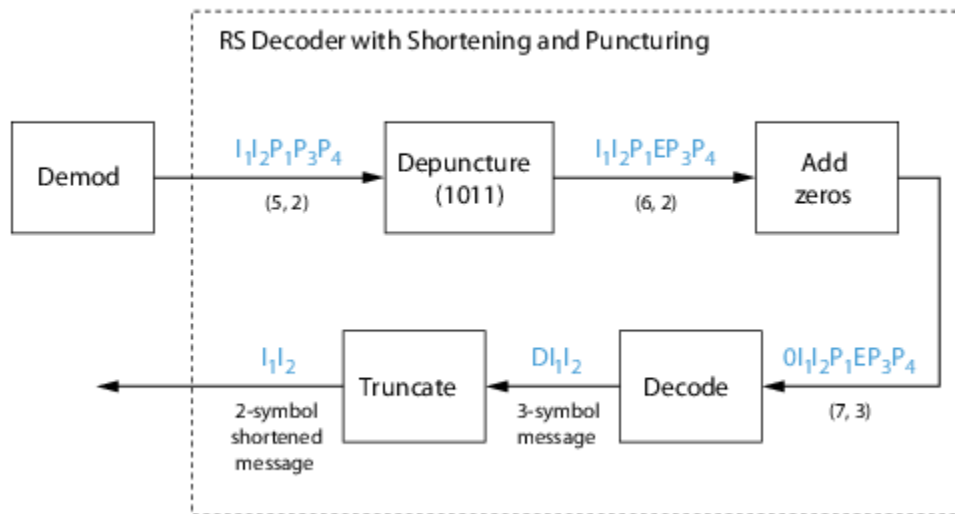
In this figure, the message source outputs two information symbols, designated by I_1I_2 . (For a BCH example, the symbols are simply binary bits.) Because the code is a shortened (7,3) code, a zero must be added ahead of the information symbols, yielding a three-symbol message of $0I_1I_2$. The modified message sequence is then RS encoded, and the added information zero is subsequently removed, which yields a result of $I_1I_2P_1P_2P_3P_4$. (In this example, the parity bits are at the end of the codeword.)

The puncturing operation is governed by the puncture vector, which, in this case, is 1011. Within the puncture vector, a 1 means that the symbol is kept, and a 0 means that the

symbol is thrown away. In this example, the puncturing operation removes the second parity symbol, yielding a final vector of $I_1I_2P_1P_3P_4$.

Decoder Example with Shortening and Puncturing

The following figure shows how the RS encoder operates on a shortened and punctured codeword.

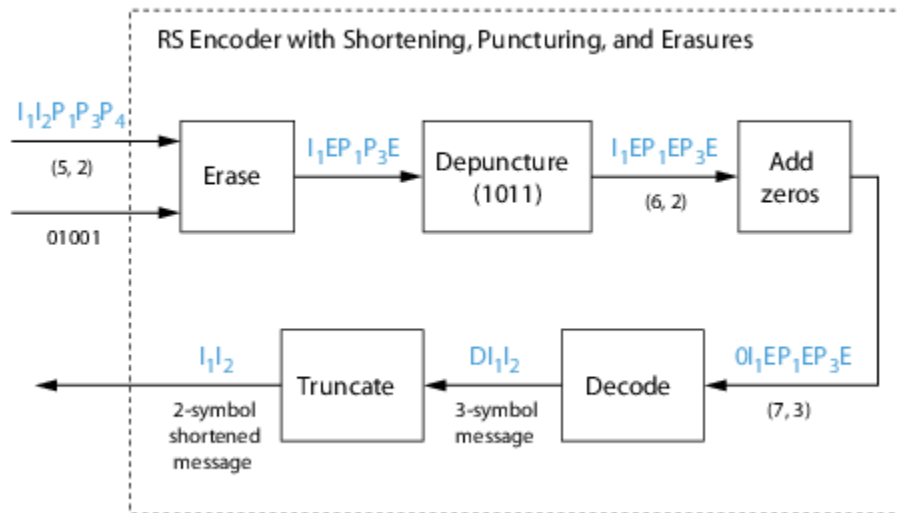


This case corresponds to the encoder operations shown in the figure of the RS encoder with shortening and puncturing. As shown in the preceding figure, the encoder receives a (5, 2) codeword, because it has been shortened from a (7, 3) codeword by one symbol, and one symbol has also been punctured.

As a first step, the decoder adds an erasure, designated by E, in the second parity position of the codeword. This corresponds to the puncture vector 1011. Adding a zero accounts for shortening, in the same way as shown in the preceding figure. The single erasure does not exceed the erasure-correcting capability of the code, which can correct four erasures. The decoding operation results in the three-symbol message DI_1I_2 . The first symbol is truncated, as in the preceding figure, yielding a final output of I_1I_2 .

Encoder Example with Shortening, Puncturing, and Erasures

The following figure shows the decoder operating on the punctured, shortened codeword, while also correcting erasures generated by the receiver.



In this figure, demodulator receives the $I_1I_2P_1P_3P_4$ vector that the encoder sent. The demodulator declares that two of the five received symbols are unreliable enough to be erased, such that symbols 2 and 5 are deemed to be erasures. The 01001 vector, provided by an external source, indicates these erasures. Within the erasures vector, a 1 means that the symbol is to be replaced with an erasure symbol, and a 0 means that the symbol is passed unaltered.

The decoder blocks receive the codeword and the erasure vector, and perform the erasures indicated by the vector 01001. Within the erasures vector, a 1 means that the symbol is to be replaced with an erasure symbol, and a 0 means that the symbol is passed unaltered. The resulting codeword vector is $I_1EP_1P_3E$, where E is an erasure symbol.

The codeword is then depunctured, according to the puncture vector used in the encoding operation (i.e., 1011). Thus, an erasure symbol is inserted between P_1 and P_3 , yielding a codeword vector of $I_1EP_1EP_3E$.

Just prior to decoding, the addition of zeros at the beginning of the information vector accounts for the shortening. The resulting vector is $0I_1EP_1EP_3E$, such that a (7,3) codeword is sent to the Berlekamp algorithm.

This codeword is decoded, yielding a three-symbol message of DI_1I_2 (where D refers to a dummy symbol). Finally, the removal of the D symbol from the message vector accounts for the shortening and yields the original I_1I_2 vector.

For additional information, see the “Reed-Solomon Coding with Erasures, Punctures, and Shortening” on page 11-20 example.

LDPC Codes

Low-Density Parity-Check (LDPC) codes are linear error control codes with:

- Sparse parity-check matrices
- Long block lengths that can attain performance near the Shannon limit (see LDPC Encoder and LDPC Decoder)

Communications Toolbox performs LDPC Coding using Simulink blocks and MATLAB objects.

The decoding process is done iteratively. If the number of iterations is too small, the algorithm may not converge. You may need to experiment with the number of iterations to find an appropriate value for your model. For details on the decoding algorithm, see Decoding Algorithm.

Unlike some other codecs, you cannot connect an LDPC decoder directly to the output of an LDPC encoder, because the decoder requires log-likelihood ratios (LLR). Thus, you may use a demodulator to compute the LLRs.



Also, unlike other decoders, it is possible (although rare) that the output of the LDPC decoder does not satisfy all parity checks.

Galois Field Computations

A *Galois field* is an algebraic field that has a finite number of members. Galois fields having 2^m members are used in error-control coding and are denoted $GF(2^m)$. This chapter describes how to work with fields that have 2^m members, where m is an integer between 1 and 16. The sections in this chapter are as follows.

- “Galois Field Terminology” on page 8-108
- “Representing Elements of Galois Fields” on page 8-109
- “Arithmetic in Galois Fields” on page 8-115
- “Logical Operations in Galois Fields” on page 8-121
- “Matrix Manipulation in Galois Fields” on page 8-123
- “Linear Algebra in Galois Fields” on page 8-124
- “Signal Processing Operations in Galois Fields” on page 8-127
- “Polynomials over Galois Fields” on page 8-130
- “Manipulating Galois Variables” on page 8-134
- “Speed and Nondefault Primitive Polynomials” on page 8-137
- “Selected Bibliography for Galois Fields” on page 8-138

If you need to use Galois fields having an odd number of elements, see “Galois Fields of Odd Characteristic” on page 8-138.

For more details about specific functions that process arrays of Galois field elements, see the online reference pages in the documentation for MATLAB or for Communications Toolbox software.

Note Please note that the Galois field objects do not support the copy method.

MATLAB functions whose generalization to Galois fields is straightforward to describe do not have reference pages in this manual because the entries would be identical to those in the MATLAB documentation.

Galois Field Terminology

The discussion of Galois fields in this document uses a few terms that are not used consistently in the literature. The definitions adopted here appear in van Lint [4]:

- A *primitive element* of $\text{GF}(2^m)$ is a cyclic generator of the group of nonzero elements of $\text{GF}(2^m)$. This means that every nonzero element of the field can be expressed as the primitive element raised to some integer power.
- A *primitive polynomial* for $\text{GF}(2^m)$ is the minimal polynomial of some primitive element of $\text{GF}(2^m)$. It is the binary-coefficient polynomial of smallest nonzero degree having a certain primitive element as a root in $\text{GF}(2^m)$. As a consequence, a primitive polynomial has degree m and is irreducible.

The definitions imply that a primitive element is a root of a corresponding primitive polynomial.

Representing Elements of Galois Fields

- “Section Overview” on page 8-109
- “Creating a Galois field array” on page 8-109
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Section Overview

This section describes how to create a *Galois field array*, which is a MATLAB expression that represents the elements of a Galois field. This section also describes how MATLAB technical computing software interprets the numbers that you use in the representation, and includes several examples.

Creating a Galois field array

To begin working with data from a Galois field $GF(2^m)$, you must set the context by associating the data with crucial information about the field. The `gf` function performs this association and creates a Galois field array in MATLAB. This function accepts as inputs

- The Galois field data, x , which is a MATLAB array whose elements are integers between 0 and $2^m - 1$.
- (*Optional*) An integer, m , that indicates x is in the field $GF(2^m)$. Valid values of m are between 1 and 16. The default is 1, which means that the field is $GF(2)$.
- (*Optional*) A positive integer that indicates which primitive polynomial for $GF(2^m)$ you are using in the representations in x . If you omit this input argument, `gf` uses a default primitive polynomial for $GF(2^m)$. For information about this argument, see “Specifying the Primitive Polynomial” on page 8-113.

The output of the `gf` function is a variable that MATLAB recognizes as a Galois field array, rather than an array of integers. As a result, when you manipulate the variable, MATLAB works within the Galois field you have specified. For example, if you apply the `log`

function to a Galois field array, MATLAB computes the logarithm in the Galois field and *not* in the field of real or complex numbers.

When MATLAB Implicitly Creates a Galois field array

Some operations on Galois field arrays require multiple arguments. If you specify one argument that is a Galois field array and another that is an ordinary MATLAB array, MATLAB interprets both as Galois field arrays in the same field. It implicitly invokes the `gf` function on the ordinary MATLAB array. This implicit invocation simplifies your syntax because you can omit some references to the `gf` function. For an example of the simplification, see “Example: Addition and Subtraction” on page 8-116.

Example: Creating Galois Field Variables

The code below creates a row vector whose entries are in the field $GF(4)$, and then adds the row to itself.

```
x = 0:3; % A row vector containing integers
m = 2; % Work in the field GF(2^2), or, GF(4).
a = gf(x,m) % Create a Galois array in GF(2^m).

b = a + a % Add a to itself, creating b.
```

The output is

```
a = GF(2^2) array. Primitive polynomial = D^2+D+1 (7 decimal)
```

```
Array elements =
```

```
    0    1    2    3
```

```
b = GF(2^2) array. Primitive polynomial = D^2+D+1 (7 decimal)
```

```
Array elements =
```

```
    0    0    0    0
```

The output shows the values of the Galois field arrays named `a` and `b`. Each output section indicates

- The field containing the variable, namely, $GF(2^2) = GF(4)$.
- The primitive polynomial for the field. In this case, it is the toolbox's default primitive polynomial for $GF(4)$.

- The array of Galois field values that the variable contains. In particular, the array elements in a are exactly the elements of the vector x , and the array elements in b are four instances of the zero element in $GF(4)$.

The command that creates b shows how, having defined the variable a as a Galois field array, you can add a to itself by using the ordinary $+$ operator. MATLAB performs the vectorized addition operation in the field $GF(4)$. The output shows that

- Compared to a , b is in the same field and uses the same primitive polynomial. It is not necessary to indicate the field when defining the sum, b , because MATLAB remembers that information from the definition of the addends, a .
- The array elements of b are zeros because the sum of any value with itself, in a Galois field of characteristic two, is zero. This result differs from the sum $x + x$, which represents an addition operation in the infinite field of integers.

Example: Representing Elements of $GF(8)$

To illustrate what the array elements in a Galois field array mean, the table below lists the elements of the field $GF(8)$ as integers and as polynomials in a primitive element, A . The table should help you interpret a Galois field array like

```
gf8 = gf([0:7],3); % Galois vector in GF(2^3)
```

Integer Representation	Binary Representation	Element of $GF(8)$
0	000	0
1	001	1
2	010	A
3	011	$A + 1$
4	100	A^2
5	101	$A^2 + 1$
6	110	$A^2 + A$
7	111	$A^2 + A + 1$

How Integers Correspond to Galois Field Elements

Building on the $GF(8)$ example above on page 8-111, this section explains the interpretation of array elements in a Galois field array in greater generality. The field $GF(2^m)$ has 2^m distinct elements, which this toolbox labels as $0, 1, 2, \dots, 2^m - 1$. These

integer labels correspond to elements of the Galois field via a polynomial expression involving a primitive element of the field. More specifically, each integer between 0 and $2^m - 1$ has a binary representation in m bits. Using the bits in the binary representation as coefficients in a polynomial, where the least significant bit is the constant term, leads to a binary polynomial whose order is at most $m - 1$. Evaluating the binary polynomial at a primitive element of $GF(2^m)$ leads to an element of the field.

Conversely, any element of $GF(2^m)$ can be expressed as a binary polynomial of order at most $m - 1$, evaluated at a primitive element of the field. The m -tuple of coefficients of the polynomial corresponds to the binary representation of an integer between 0 and 2^m .

Below is a symbolic illustration of the correspondence of an integer X , representable in binary form, with a Galois field element. Each b_k is either zero or one, while A is a primitive element.

$$\begin{aligned} X &= b_{m-1} \cdot 2^{m-1} + \dots + b_2 \cdot 4 + b_1 \cdot 2 + b_0 \\ &\leftrightarrow b_{m-1} \cdot A^{m-1} + \dots + b_2 \cdot A^2 + b_1 \cdot A + b_0 \end{aligned}$$

Example: Representing a Primitive Element

The code below defines a variable `alph` that represents a primitive element of the field $GF(2^4)$.

```
m = 4; % Or choose any positive integer value of m.  
alph = gf(2,m) % Primitive element in GF(2^m)
```

The output is

```
alph = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
```

```
Array elements =
```

```
2
```

The Galois field array `alph` represents a primitive element because of the correspondence among

- The integer 2, specified in the `gf` syntax
- The binary representation of 2, which is 10 (or 0010 using four bits)
- The polynomial $A + 0$, where A is a primitive element in this field (or $0A^3 + 0A^2 + A + 0$ using the four lowest powers of A)

Primitive Polynomials and Element Representations

This section builds on the discussion in “Creating a Galois field array” on page 8-109 by describing how to specify your own primitive polynomial when you create a Galois field array. The topics are

If you perform many computations using a nondefault primitive polynomial, see “Speed and Nondefault Primitive Polynomials” on page 8-137.

Specifying the Primitive Polynomial

The discussion in “How Integers Correspond to Galois Field Elements” on page 8-111 refers to a primitive element, which is a root of a primitive polynomial of the field. When you use the `gf` function to create a Galois field array, the function interprets the integers in the array with respect to a specific default primitive polynomial for that field, unless you explicitly provide a different primitive polynomial. A list of the default primitive polynomials is on the reference page for the `gf` function.

To specify your own primitive polynomial when creating a Galois field array, use a syntax like

```
c = gf(5,4,25) % 25 indicates the primitive polynomial for GF(16).
```

instead of

```
c1= gf(5,4); % Use default primitive polynomial for GF(16).
```

The extra input argument, 25 in this case, specifies the primitive polynomial for the field $GF(2^m)$ in a way similar to the representation described in “How Integers Correspond to Galois Field Elements” on page 8-111. In this case, the integer 25 corresponds to a binary representation of 11001, which in turn corresponds to the polynomial $D^4 + D^3 + 1$.

Note When you specify the primitive polynomial, the input argument must have a binary representation using exactly $m+1$ bits, not including unnecessary leading zeros. In other words, a primitive polynomial for $GF(2^m)$ always has order m .

When you use an input argument to specify the primitive polynomial, the output reflects your choice by showing the integer value as well as the polynomial representation.

```
d = gf([1 2 3],4,25)
```

```
d = GF(2^4) array. Primitive polynomial = D^4+D^3+1 (25 decimal)
```

Array elements =

1 2 3

Note After you have defined a Galois field array, you cannot change the primitive polynomial with respect to which MATLAB interprets the array elements.

Finding Primitive Polynomials

You can use the `primpoly` function to find primitive polynomials for $GF(2^m)$ and the `isprimitive` function to determine whether a polynomial is primitive for $GF(2^m)$. The code below illustrates.

```
m = 4;
defaultprimpoly = primpoly(m) % Default primitive poly for GF(16)
allprimpolys = primpoly(m, 'all') % All primitive polys for GF(16)
i1 = isprimitive(25) % Can 25 be the prim_poly input in gf(...)?
i2 = isprimitive(21) % Can 21 be the prim_poly input in gf(...)?
```

The output is below.

```
Primitive polynomial(s) =
```

```
D^4+D^1+1
```

```
defaultprimpoly =
```

```
19
```

```
Primitive polynomial(s) =
```

```
D^4+D^1+1
```

```
D^4+D^3+1
```

```
allprimpolys =
```

```
19
```

```
25
```

```
i1 =
```

```
1
```



```
i2 =
```

```
0
```

Effect of Nondefault Primitive Polynomials on Numerical Results

Most fields offer multiple choices for the primitive polynomial that helps define the representation of members of the field. When you use the `gf` function, changing the primitive polynomial changes the interpretation of the array elements and, in turn, changes the results of some subsequent operations on the Galois field array. For example, exponentiation of a primitive element makes it easy to see how the primitive polynomial affects the representations of field elements.

```
a11 = gf(2,3); % Use default primitive polynomial of 11.
a13 = gf(2,3,13); % Use D^3+D^2+1 as the primitive polynomial.
z = a13.^3 + a13.^2 + 1 % 0 because a13 satisfies the equation
nz = a11.^3 + a11.^2 + 1 % Nonzero. a11 does not satisfy equation.
```

The output below shows that when the primitive polynomial has integer representation 13, the Galois field array satisfies a certain equation. By contrast, when the primitive polynomial has integer representation 11, the Galois field array fails to satisfy the equation.

```
z = GF(2^3) array. Primitive polynomial = D^3+D^2+1 (13 decimal)
```

```
Array elements =
```

```
0
```

```
nz = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)
```

```
Array elements =
```

```
6
```

The output when you try this example might also include a warning about lookup tables. This is normal if you did not use the `gftable` function to optimize computations involving a nondefault primitive polynomial of 13.

Arithmetic in Galois Fields

- “Section Overview” on page 8-116

- “Example: Addition and Subtraction” on page 8-116
- “Example: Multiplication” on page 8-118
- “Example: Division” on page 8-119
- “Example: Exponentiation” on page 8-119
- “Example: Elementwise Logarithm” on page 8-120

Section Overview

You can perform arithmetic operations on Galois field arrays by using familiar MATLAB operators, listed in the table below. Whenever you operate on a pair of Galois field arrays, both arrays must be in the same Galois field.

Operation	Operator
Addition	+
Subtraction	-
Elementwise multiplication	.*
Matrix multiplication	*
Elementwise left division	./
Elementwise right division	.\
Matrix left division	/
Matrix right division	\
Elementwise exponentiation	.^
Elementwise logarithm	log ()
Exponentiation of a square Galois matrix by a scalar integer	^

For multiplication and division of polynomials over a Galois field, see “Addition and Subtraction of Polynomials” on page 8-131.

Example: Addition and Subtraction

The code below adds two Galois field arrays to create an addition table for GF(8). Addition uses the ordinary + operator. The code below also shows how to index into the array `addtb` to find the result of adding 1 to the elements of GF(8).

```
m = 3;
e = repmat([0:2^m-1],2^m,1);
```

```
f = gf(e,m); % Create a Galois array.
addtb = f + f' % Add f to its own matrix transpose.

addone = addtb(2,:); % Assign 2nd row to the Galois vector addone.
```

The output is below.

```
addtb = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)
```

Array elements =

0	1	2	3	4	5	6	7
1	0	3	2	5	4	7	6
2	3	0	1	6	7	4	5
3	2	1	0	7	6	5	4
4	5	6	7	0	1	2	3
5	4	7	6	1	0	3	2
6	7	4	5	2	3	0	1
7	6	5	4	3	2	1	0

As an example of reading this addition table, the (7,4) entry in the `addtb` array shows that `gf(6,3)` plus `gf(3,3)` equals `gf(5,3)`. Equivalently, the element A^2+A plus the element $A+1$ equals the element A^2+1 . The equivalence arises from the binary representation of 6 as 110, 3 as 011, and 5 as 101.

The subtraction table, which you can obtain by replacing `+` by `-`, is the same as `addtb`. This is because subtraction and addition are identical operations in a field of *characteristic two*. In fact, the zeros along the main diagonal of `addtb` illustrate this fact for `GF(8)`.

Simplifying the Syntax

The code below illustrates scalar expansion and the implicit creation of a Galois field array from an ordinary MATLAB array. The Galois field arrays `h` and `h1` are identical, but the creation of `h` uses a simpler syntax.

```
g = gf(ones(2,3),4); % Create a Galois array explicitly.
h = g + 5; % Add gf(5,4) to each element of g.
h1 = g + gf(5*ones(2,3),4) % Same as h.
```

The output is below.

```
h1 = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
```

Array elements =

```
    4    4    4
    4    4    4
```

Notice that $1+5$ is reported as 4 in the Galois field. This is true because the 5 represents the polynomial expression A^2+1 , and $1+(A^2+1)$ in $GF(16)$ is A^2 . Furthermore, the integer that represents the polynomial expression A^2 is 4.

Example: Multiplication

The example below multiplies individual elements in a Galois field array using the `.*` operator. It then performs matrix multiplication using the `*` operator. The elementwise multiplication produces an array whose size matches that of the inputs. By contrast, the matrix multiplication produces a Galois scalar because it is the matrix product of a row vector with a column vector.

```
m = 5;
row1 = gf([1:2:9],m); row2 = gf([2:2:10],m);
col = row2'; % Transpose to create a column array.
ep = row1 .* row2; % Elementwise product.
mp = row1 * col; % Matrix product.
```

Multiplication Table for GF(8)

As another example, the code below multiplies two Galois vectors using matrix multiplication. The result is a multiplication table for $GF(8)$.

```
m = 3;
els = gf([0:2^m-1]',m);
multb = els * els' % Multiply els by its own matrix transpose.
```

The output is below.

```
multb = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)
```

Array elements =

```
    0    0    0    0    0    0    0    0
    0    1    2    3    4    5    6    7
    0    2    4    6    3    1    7    5
    0    3    6    5    7    4    1    2
    0    4    3    7    6    2    5    1
    0    5    1    4    2    7    3    6
```

0	6	7	1	5	3	2	4
0	7	5	2	1	6	4	3

Example: Division

The examples below illustrate the four division operators in a Galois field by computing multiplicative inverses of individual elements and of an array. You can also compute inverses using `inv` or using exponentiation by `-1`.

Elementwise Division

This example divides 1 by each of the individual elements in a Galois field array using the `./` and `.\` operators. These two operators differ only in their sequence of input arguments. Each quotient vector lists the multiplicative inverses of the nonzero elements of the field. In this example, MATLAB expands the scalar 1 to the size of `nz` before computing; alternatively, you can use as arguments two arrays of the same size.

```
m = 5;
nz = gf([1:2^m-1],m); % Nonzero elements of the field
inv1 = 1 ./ nz; % Divide 1 by each element.
inv2 = nz .\ 1; % Obtain same result using .\ operator.
```

Matrix Division

This example divides the identity array by the square Galois field array `mat` using the `/` and `\` operators. Each quotient matrix is the multiplicative inverse of `mat`. Notice how the transpose operator (`'`) appears in the equivalent operation using `\`. For square matrices, the sequence of transpose operations is unnecessary, but for nonsquare matrices, it is necessary.

```
m = 5;
mat = gf([1 2 3; 4 5 6; 7 8 9],m);
minv1 = eye(3) / mat; % Compute matrix inverse.
minv2 = (mat' \ eye(3)')'; % Obtain same result using \ operator.
```

Example: Exponentiation

The examples below illustrate how to compute integer powers of a Galois field array. To perform matrix exponentiation on a Galois field array, you must use a square Galois field array as the base and an ordinary (not Galois) integer scalar as the exponent.

Elementwise Exponentiation

This example computes powers of a primitive element, `A`, of a Galois field. It then uses these separately computed powers to evaluate the default primitive polynomial at `A`. The

answer of zero shows that A is a root of the primitive polynomial. The `.` operator exponentiates each array element independently.

```
m = 3;
av = gf(2*ones(1,m+1),m); % Row containing primitive element
expa = av .^ [0:m]; % Raise element to different powers.
evp = expa(4)+expa(2)+expa(1) % Evaluate D^3 + D + 1.
```

The output is below.

```
evp = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)
```

Array elements =

```
0
```

Matrix Exponentiation

This example computes the inverse of a square matrix by raising the matrix to the power -1. It also raises the square matrix to the powers 2 and -2.

```
m = 5;
mat = gf([1 2 3; 4 5 6; 7 8 9],m);
minvs = mat ^ (-1); % Matrix inverse
matsq = mat^2; % Same as mat * mat
matinvssq = mat^(-2); % Same as minvs * minvs
```

Example: Elementwise Logarithm

The code below computes the logarithm of the elements of a Galois field array. The output indicates how to express each *nonzero* element of GF(8) as a power of the primitive element. The logarithm of the zero element of the field is undefined.

```
gf8_nonzero = gf([1:7],3); % Vector of nonzero elements of GF(8)
expformat = log(gf8_nonzero) % Logarithm of each element
```

The output is

```
expformat =
```

```
0    1    3    2    6    4    5
```

As an example of how to interpret the output, consider the last entry in each vector in this example. You can infer that the element `gf(7,3)` in GF(8) can be expressed as either

- A^5 , using the last element of `expformat`
- A^2+A+1 , using the binary representation of 7 as 111. See “Example: Representing Elements of GF(8)” on page 8-111 for more details.

Logical Operations in Galois Fields

- “Section Overview” on page 8-121
- “Testing for Equality” on page 8-121
- “Testing for Nonzero Values” on page 8-122

Section Overview

You can apply logical tests to Galois field arrays and obtain a logical array. Some important types of tests are testing for the equality of two Galois field arrays and testing for nonzero values on page 8-122 in a Galois field array.

Testing for Equality

To compare corresponding elements of two Galois field arrays that have the same size, use the operators `==` and `~=`. The result is a logical array, each element of which indicates the truth or falsity of the corresponding elementwise comparison. If you use the same operators to compare a scalar with a Galois field array, MATLAB technical computing software compares the scalar with each element of the array, producing a logical array of the same size.

```
m = 5; r1 = gf([1:3],m); r2 = 1 ./ r1;
lg1 = (r1 .* r2 == [1 1 1]) % Does each element equal one?
lg2 = (r1 .* r2 == 1) % Same as above, using scalar expansion
lg3 = (r1 ~= r2) % Does each element differ from its inverse?
```

The output is below.

```
lg1 =
```

```
    1    1    1
```

```
lg2 =
```

```
    1    1    1
```

```
lg3 =
```

0 1 1

Comparison of `isequal` and `==`

To compare entire arrays and obtain a logical *scalar* result rather than a logical array, use the built-in `isequal` function. However, `isequal` uses strict rules for its comparison, and returns a value of 0 (false) if you compare

- A Galois field array with an ordinary MATLAB array, even if the values of the underlying array elements match
- A scalar with a nonscalar array, even if all elements in the array match the scalar

The example below illustrates this difference between `==` and `isequal`.

```
m = 5; r1 = gf([1:3],m); r2 = 1 ./ r1;
lg4 = isequal(r1 .* r2, [1 1 1]); % False
lg5 = isequal(r1 .* r2, gf(1,m)); % False
lg6 = isequal(r1 .* r2, gf([1 1 1],m)); % True
```

Testing for Nonzero Values

To test for nonzero values in a Galois vector, or in the columns of a Galois field array that has more than one row, use the `any` or `all` function. These two functions behave just like the ordinary MATLAB functions `any` and `all`, except that they consider only the underlying array elements while ignoring information about which Galois field the elements are in. Examples are below.

```
m = 3; randels = gf(randi([0 2^m-1],6,1),m);
if all(randels) % If all elements are invertible
    invels = randels .\ 1; % Compute inverses of elements.
else
    disp('At least one element was not invertible.');
```

```
end
alph = gf(2,4);
poly = 1 + alph + alph^3;
if any(poly) % If poly contains a nonzero value
    disp('alph is not a root of 1 + D + D^3.');
```

```
end
code = [0:4 4 0; 3:7 4 5]
if all(code,2) % Is each row entirely nonzero?
    disp('Both codewords are entirely nonzero.');
```

```
else
```



```

    disp('At least one codeword contains a zero.');
```

end

Matrix Manipulation in Galois Fields

- “Basic Manipulations of Galois Field Arrays” on page 8-123
- “Basic Information About Galois Field Arrays” on page 8-124

Basic Manipulations of Galois Field Arrays

Basic array operations on Galois field arrays are in the table below. The functionality of these operations is analogous to the MATLAB operations having the same syntax.

Operation	Syntax
Index into array, possibly using colon operator instead of a vector of explicit indices	<code>a(vector)</code> or <code>a(vector,vector1)</code> , where <code>vector</code> and/or <code>vector1</code> can be <code>":"</code> instead of a vector
Transpose array	<code>a'</code>
Concatenate matrices	<code>[a,b]</code> or <code>[a;b]</code>
Create array having specified diagonal elements	<code>diag(vector)</code> or <code>diag(vector,k)</code>
Extract diagonal elements	<code>diag(a)</code> or <code>diag(a,k)</code>
Extract lower triangular part	<code>tril(a)</code> or <code>tril(a,k)</code>
Extract upper triangular part	<code>triu(a)</code> or <code>triu(a,k)</code>
Change shape of array	<code>reshape(a,k1,k2)</code>

The code below uses some of these syntaxes.

```

m = 4; a = gf([0:15],m);
a(1:2) = [13 13]; % Replace some elements of the vector a.
b = reshape(a,2,8); % Create 2-by-8 matrix.
c = [b([1 1 2],1:3); a(4:6)]; % Create 4-by-3 matrix.
d = [c, a(1:4)']; % Create 4-by-4 matrix.
dvec = diag(d); % Extract main diagonal of d.
dmat = diag(a(5:9)); % Create 5-by-5 diagonal matrix
dtril = tril(d); % Extract upper and lower triangular
dtriu = triu(d); % parts of d.
```

Basic Information About Galois Field Arrays

You can determine the length of a Galois vector or the size of any Galois field array using the `length` and `size` functions. The functionality for Galois field arrays is analogous to that of the MATLAB operations on ordinary arrays, except that the output arguments from `size` and `length` are always integers, not Galois field arrays. The code below illustrates the use of these functions.

```
m = 4; e = gf([0:5],m); f = reshape(e,2,3);
lne = length(e); % Vector length of e
szf = size(f); % Size of f, returned as a two-element row
[nr,nc] = size(f); % Size of f, returned as two scalars
nc2 = size(f,2); % Another way to compute number of columns
```

Positions of Nonzero Elements

Another type of information you might want to determine from a Galois field array are the positions of nonzero elements. For an ordinary MATLAB array, you might use the `find` function. However, for a Galois field array, you should use `find` in conjunction with the `~=` operator, as illustrated.

```
x = [0 1 2 1 0 2]; m = 2; g = gf(x,m);
nzx = find(x); % Find nonzero values in the ordinary array x.
nzg = find(g~=0); % Find nonzero values in the Galois array g.
```

Linear Algebra in Galois Fields

- “Inverting Matrices and Computing Determinants” on page 8-124
- “Computing Ranks” on page 8-125
- “Factoring Square Matrices” on page 8-126
- “Solving Linear Equations” on page 8-126

Inverting Matrices and Computing Determinants

To invert a square Galois field array, use the `inv` function. Related is the `det` function, which computes the determinant of a Galois field array. Both `inv` and `det` behave like their ordinary MATLAB counterparts, except that they perform computations in the Galois field instead of in the field of complex numbers.

Note A Galois field array is singular if and only if its determinant is exactly zero. It is not necessary to consider roundoff errors, as in the case of real and complex arrays.

The code below illustrates matrix inversion and determinant computation.

```
m = 4;
randommatrix = gf(randi([0 2^m-1],4,4),m);
gfid = gf(eye(4),m);
if det(randommatrix) ~= 0
    invmatrix = inv(randommatrix);
    check1 = invmatrix * randommatrix;
    check2 = randommatrix * invmatrix;
    if (isequal(check1,gfid) & isequal(check2,gfid))
        disp('inv found the correct matrix inverse.');
```

The output from this example is either of these two messages, depending on whether the randomly generated matrix is nonsingular or singular.

```
inv found the correct matrix inverse.
The matrix is not invertible.
```

Computing Ranks

To compute the rank of a Galois field array, use the rank function. It behaves like the ordinary MATLAB rank function when given exactly one input argument. The example below illustrates how to find the rank of square and nonsquare Galois field arrays.

```
m = 3;
asquare = gf([4 7 6; 4 6 5; 0 6 1],m);
r1 = rank(asquare);
anonsquare = gf([4 7 6 3; 4 6 5 1; 0 6 1 1],m);
r2 = rank(anonsquare);
[r1 r2]
```

The output is

```
ans =
     2     3
```

The values of r1 and r2 indicate that asquare has less than full rank but that anonsquare has full rank.

Factoring Square Matrices

To express a square Galois field array (or a permutation of it) as the product of a lower triangular Galois field array and an upper triangular Galois field array, use the `lu` function. This function accepts one input argument and produces exactly two or three output arguments. It behaves like the ordinary MATLAB `lu` function when given the same syntax. The example below illustrates how to factor using `lu`.

```
tofactor = gf([6 5 7 6; 5 6 2 5; 0 1 7 7; 1 0 5 1],3);
[L,U]=lu(tofactor); % lu with two output arguments
c1 = isequal(L*U, tofactor) % True
tofactor2 = gf([1 2 3 4;1 2 3 0;2 5 2 1; 0 5 0 0],3);
[L2,U2,P] = lu(tofactor2); % lu with three output arguments
c2 = isequal(L2*U2, P*tofactor2) % True
```

Solving Linear Equations

To find a particular solution of a linear equation in a Galois field, use the `\` or `/` operator on Galois field arrays. The table below indicates the equation that each operator addresses, assuming that `A` and `B` are previously defined Galois field arrays.

Operator	Linear Equation	Syntax	Equivalent Syntax Using <code>\</code>
Backslash (<code>\</code>)	$A * x = B$	<code>x = A \ B</code>	Not applicable
Slash (<code>/</code>)	$x * A = B$	<code>x = B / A</code>	<code>x = (A'\B')'</code>

The results of the syntax in the table depend on characteristics of the Galois field array `A`:

- If `A` is square and nonsingular, the output `x` is the unique solution to the linear equation.
- If `A` is square and singular, the syntax in the table produces an error.
- If `A` is not square, MATLAB attempts to find a particular solution. If `A'*A` or `A*A'` is a singular array, or if `A` is a matrix, where the rows outnumber the columns, that represents an overdetermined system, the attempt might fail.

Note An error message does not necessarily indicate that the linear equation has no solution. You might be able to find a solution by rephrasing the problem. For example, `gf([1 2; 0 0],3) \ gf([1; 0],3)` produces an error but the mathematically equivalent `gf([1 2],3) \ gf([1],3)` does not. The first syntax fails because `gf([1 2; 0 0],3)` is a singular square matrix.

Example: Solving Linear Equations

The examples below illustrate how to find particular solutions of linear equations over a Galois field.

```

m = 4;
A = gf(magic(3),m); % Square nonsingular matrix
Awide=[A, 2*A(:,3)]; % 3-by-4 matrix with redundancy on the right
Atall = Awide'; % 4-by-3 matrix with redundancy at the bottom
B = gf([0:2]',m);
C = [B; 2*B(3)];
D = [B; B(3)+1];
thesolution = A \ B; % Solution of A * x = B
thesolution2 = B' / A; % Solution of x * A = B'
ck1 = all(A * thesolution == B) % Check validity of solutions.
ck2 = all(thesolution2 * A == B')
% Awide * x = B has infinitely many solutions. Find one.
onesolution = Awide \ B;
ck3 = all(Awide * onesolution == B) % Check validity of solution.
% Atall * x = C has a solution.
asolution = Atall \ C;
ck4 = all(Atall * asolution == C) % Check validity of solution.
% Atall * x = D has no solution.
notasolution = Atall \ D;
ck5 = all(Atall * notasolution == D) % It is not a valid solution.

```

The output from this example indicates that the validity checks are all true (1), except for ck5, which is false (0).

Signal Processing Operations in Galois Fields

- “Section Overview” on page 8-127
- “Filtering” on page 8-128
- “Convolution” on page 8-129
- “Discrete Fourier Transform” on page 8-129

Section Overview

You can perform some signal-processing operations on Galois field arrays, such as filtering on page 8-128, convolution on page 8-129, and the discrete Fourier transform on page 8-129.

This section describes how to perform these operations.

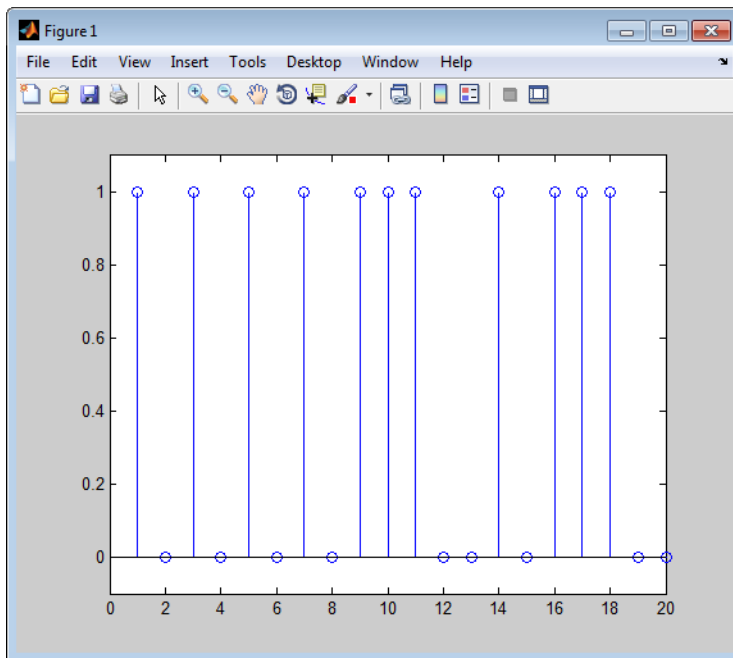
Other information about the corresponding operations for ordinary real vectors is in the Signal Processing Toolbox™ documentation.

Filtering

To filter a Galois vector, use the `filter` function. It behaves like the ordinary MATLAB `filter` function when given exactly three input arguments.

The code and diagram below give the impulse response of a particular filter over GF(2).

```
m = 1; % Work in GF(2).
b = gf([1 0 0 1 0 1 0 1],m); % Numerator
a = gf([1 0 1 1],m); % Denominator
x = gf([1,zeros(1,19)],m);
y = filter(b,a,x); % Filter x.
figure; stem(y,x); % Create stem plot.
axis([0 20 -.1 1.1])
```



Convolution

Communications Toolbox software offers two equivalent ways to convolve a pair of Galois vectors:

- Use the `conv` function, as described in “Multiplication and Division of Polynomials” on page 8-131. This works because convolving two vectors is equivalent to multiplying the two polynomials whose coefficients are the entries of the vectors.
- Use the `convmtx` function to compute the convolution matrix of one of the vectors, and then multiply that matrix by the other vector. This works because convolving two vectors is equivalent to filtering one of the vectors by the other. The equivalence permits the representation of a digital filter as a convolution matrix, which you can then multiply by any Galois vector of appropriate length.

Tip If you need to convolve large Galois vectors, multiplying by the convolution matrix might be faster than using `conv`.

Example

Computes the convolution matrix for a vector `b` in GF(4). Represent the numerator coefficients for a digital filter, and then illustrate the two equivalent ways to convolve `b` with `x` over the Galois field.

```
m = 2; b = gf([1 2 3]',m);
n = 3; x = gf(randi([0 2^m-1],n,1),m);
C = convmtx(b,n); % Compute convolution matrix.
v1 = conv(b,x); % Use conv to convolve b with x
v2 = C*x; % Use C to convolve b with x.
```

Discrete Fourier Transform

The discrete Fourier transform is an important tool in digital signal processing. This toolbox offers these tools to help you process discrete Fourier transforms:

- `fft`, which transforms a Galois vector
- `ifft`, which inverts the discrete Fourier transform on a Galois vector
- `dftmtx`, which returns a Galois field array that you can use to perform or invert the discrete Fourier transform on a Galois vector

In all cases, the vector being transformed must be a Galois vector of length 2^m-1 in the field $GF(2^m)$. The following example illustrates the use of these functions. You can check, using the `isequal` function, that `y` equals `y1`, `z` equals `z1`, and `z` equals `x`.

```
m = 4;
x = gf(randi([0 2^m-1],2^m-1,1),m); % A vector to transform
alph = gf(2,m);
dm = dftmtx(alph);
idm = dftmtx(1/alph);
y = dm*x; % Transform x using the result of dftmtx.
y1 = fft(x); % Transform x using fft.
z = idm*y; % Recover x using the result of dftmtx(1/alph).
z1 = ifft(y1); % Recover x using ifft.
```

Tip If you have many vectors that you want to transform (in the same field), it might be faster to use `dftmtx` once and matrix multiplication many times, instead of using `fft` many times.

Polynomials over Galois Fields

- “Section Overview” on page 8-130
- “Addition and Subtraction of Polynomials” on page 8-131
- “Multiplication and Division of Polynomials” on page 8-131
- “Evaluating Polynomials” on page 8-131
- “Roots of Polynomials” on page 8-132
- “Roots of Binary Polynomials” on page 8-133
- “Minimal Polynomials” on page 8-134

Section Overview

You can use Galois vectors to represent polynomials in an indeterminate quantity `x`, with coefficients in a Galois field. Form the representation by listing the coefficients of the polynomial in a vector in order of descending powers of `x`. For example, the vector

```
gf([2 1 0 3],4)
```

represents the polynomial $Ax^3 + 1x^2 + 0x + (A+1)$, where

- `A` is a primitive element in the field $GF(2^4)$.

- x is the indeterminate quantity in the polynomial.

You can then use such a Galois vector to perform arithmetic with, evaluate on page 8-131, and find roots on page 8-132 of polynomials. You can also find minimal polynomials on page 8-134 of elements of a Galois field.

Addition and Subtraction of Polynomials

To add and subtract polynomials, use `+` and `-` on equal-length Galois vectors that represent the polynomials. If one polynomial has lower degree than the other, you must pad the shorter vector with zeros at the beginning so the two vectors have the same length. The example below shows how to add a degree-one and a degree-two polynomial.

```
lin = gf([4 2],3); % A^2 x + A, which is linear in x
linpadded = gf([0 4 2],3); % The same polynomial, zero-padded
quadr = gf([1 4 2],3); % x^2 + A^2 x + A, which is quadratic in x
% Can't do lin + quadr because they have different vector lengths.
sumpoly = [0, lin] + quadr; % Sum of the two polynomials
sumpoly2 = linpadded + quadr; % The same sum
```

Multiplication and Division of Polynomials

To multiply and divide polynomials, use `conv` and `deconv` on Galois vectors that represent the polynomials. Multiplication and division of polynomials is equivalent to convolution and deconvolution of vectors. The `deconv` function returns the quotient of the two polynomials as well as the remainder polynomial. Examples are below.

```
m = 4;
apoly = gf([4 5 3],m); % A^2 x^2 + (A^2 + 1) x + (A + 1)
bpoly = gf([1 1],m); % x + 1
xpoly = gf([1 0],m); % x
% Product is A^2 x^3 + x^2 + (A^2 + A) x + (A + 1).
cpoly = conv(apoly,bpoly);
[a2,remd] = deconv(cpoly,bpoly); % a2==apoly. remd is zero.
[otherpol,remd2] = deconv(cpoly,xpoly); % remd is nonzero.
```

The multiplication and division operators in “Arithmetic in Galois Fields” on page 8-115 multiply elements or matrices, not polynomials.

Evaluating Polynomials

To evaluate a polynomial at an element of a Galois field, use `polyval`. It behaves like the ordinary MATLAB `polyval` function when given exactly two input arguments. The example below evaluates a polynomial at several elements in a field and checks the results using `.^` and `.*` in the field.

```
m = 4;
apoly = gf([4 5 3],m); % A^2 x^2 + (A^2 + 1) x + (A + 1)
x0 = gf([0 1 2],m); % Points at which to evaluate the polynomial
y = polyval(apoly,x0)

a = gf(2,m); % Primitive element of the field, corresponding to A.
y2 = a.^2.*x0.^2 + (a.^2+1).*x0 + (a+1) % Check the result.
```

The output is below.

```
y = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
```

```
Array elements =
```

```
3    2   10
```

```
y2 = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
```

```
Array elements =
```

```
3    2   10
```

The first element of y evaluates the polynomial at 0 and, therefore, returns the polynomial's constant term of 3.

Roots of Polynomials

To find the roots of a polynomial in a Galois field, use the `roots` function on a Galois vector that represents the polynomial. This function finds roots that are in the same field that the Galois vector is in. The number of times an entry appears in the output vector from `roots` is exactly its multiplicity as a root of the polynomial.

Note If the Galois vector is in $GF(2^m)$, the polynomial it represents might have additional roots in some extension field $GF((2^m)^k)$. However, `roots` does not find those additional roots or indicate their existence.

The examples below find roots of cubic polynomials in $GF(8)$.

```
p = 3; m = 2;
field = gftuple([-1:p^m-2]',m,p); % List of all elements of GF(9)
% Use default primitive polynomial here.
polynomial = [1 0 1 1]; % 1 + x^2 + x^3
```

```

rts =gfroots(polynomial,m,p) % Find roots in exponential format
% Check that each one is actually a root.
for ii = 1:3
    root = rts(ii);
    rootsquared = gfmul(root,root,field);
    rootcubed = gfmul(root,rootsquared,field);
    answer(ii)= gfadd(gfadd(0,rootsquared,field),rootcubed,field);
    % Recall that 1 is really alpha to the zero power.
    % If answer = -Inf, then the variable root represents
    % a root of the polynomial.
end
answer

```

Roots of Binary Polynomials

In the special case of a polynomial having binary coefficients, it is also easy to find roots that exist in an extension field. This is because the elements 0 and 1 have the same unambiguous representation in all fields of characteristic two. To find roots of a binary polynomial in an extension field, apply the `roots` function to a Galois vector in the extension field whose array elements are the binary coefficients of the polynomial.

The example below seeks the roots of a binary polynomial in various fields.

```

gf2poly = gf([1 1 1],1); % x^2 + x + 1 in GF(2)
noroots = roots(gf2poly); % No roots in the ground field, GF(2)
gf4poly = gf([1 1 1],2); % x^2 + x + 1 in GF(4)
roots4 = roots(gf4poly); % The roots are A and A+1, in GF(4).
gf16poly = gf([1 1 1],4); % x^2 + x + 1 in GF(16)
roots16 = roots(gf16poly); % Roots in GF(16)
checkanswer4 = polyval(gf4poly,roots4); % Zero vector
checkanswer16 = polyval(gf16poly,roots16); % Zero vector

```

The roots of the polynomial do not exist in $GF(2)$, so `noroots` is an empty array. However, the roots of the polynomial exist in $GF(4)$ as well as in $GF(16)$, so `roots4` and `roots16` are nonempty.

Notice that `roots4` and `roots16` are not equal to each other. They differ in these ways:

- `roots4` is a $GF(4)$ array, while `roots16` is a $GF(16)$ array. MATLAB keeps track of the underlying field of a Galois field array.
- The array elements in `roots4` and `roots16` differ because they use representations with respect to different primitive polynomials. For example, 2 (which represents a primitive element) is an element of the vector `roots4` because the default primitive

polynomial for GF(4) is the same polynomial that `gf4poly` represents. On the other hand, 2 is not an element of `roots16` because the primitive element of GF(16) is not a root of the polynomial that `gf16poly` represents.

Minimal Polynomials

The minimal polynomial of an element of $GF(2^m)$ is the smallest degree nonzero binary-coefficient polynomial having that element as a root in $GF(2^m)$. To find the minimal polynomial of an element or a column vector of elements, use the `minpol` function.

The code below finds that the minimal polynomial of `gf(6,4)` is $D^2 + D + 1$ and then checks that `gf(6,4)` is indeed among the roots of that polynomial in the field GF(16).

```
m = 4;
e = gf(6,4);
em = minpol(e) % Find minimal polynomial of e. em is in GF(2).

emr = roots(gf([0 0 1 1 1],m)) % Roots of D^2+D+1 in GF(2^m)
```

The output is

```
em = GF(2) array.
```

```
Array elements =
```

```
    0    0    1    1    1
```

```
emr = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
```

```
Array elements =
```

```
    6
    7
```

To find out which elements of a Galois field share the same minimal polynomial, use the `cosets` function.

Manipulating Galois Variables

- “Section Overview” on page 8-135
- “Determining Whether a Variable Is a Galois Field Array” on page 8-135
- “Extracting Information from a Galois Field Array” on page 8-135

Section Overview

This section describes techniques for manipulating Galois variables or for transferring information between Galois field arrays and ordinary MATLAB arrays.

Note These techniques are particularly relevant if you write MATLAB file functions that process Galois field arrays. For an example of this type of usage, enter `edit gf/conv` in the Command Window and examine the first several lines of code in the editor window.

Determining Whether a Variable Is a Galois Field Array

To find out whether a variable is a Galois field array rather than an ordinary MATLAB array, use the `isa` function. An illustration is below.

```
mlvar = eye(3);
gfvar = gf(mlvar,3);
no = isa(mlvar,'gf'); % False because mlvar is not a Galois array
yes = isa(gfvar,'gf'); % True because gfvar is a Galois array
```

Extracting Information from a Galois Field Array

To extract the array elements, field order, or primitive polynomial from a variable that is a Galois field array, append a suffix to the name of the variable. The table below lists the exact suffixes, which are independent of the name of the variable.

Information	Suffix	Output Value
Array elements	<code>.x</code>	MATLAB array of type <code>uint16</code> that contains the data values from the Galois field array.
Field order	<code>.m</code>	Integer of type <code>double</code> that indicates that the Galois field array is in $GF(2^m)$.

Information	Suffix	Output Value
Primitive polynomial	.prim_poly	Integer of type uint32 that represents the primitive polynomial. The representation is similar to the description in “How Integers Correspond to Galois Field Elements” on page 8-111.

Note If the output value is an integer data type and you want to convert it to double for later manipulation, use the `double` function.

The code below illustrates the use of these suffixes. The definition of `empr` uses a vector of binary coefficients of a polynomial to create a Galois field array in an extension field. Another part of the example retrieves the primitive polynomial for the field and converts it to a binary vector representation having the appropriate number of bits.

```
% Check that e solves its own minimal polynomial.
e = gf(6,4); % An element of GF(16)
emp = minpol(e); % The minimal polynomial, emp, is in GF(2).
empr = roots(gf(emp.x,e.m)); % Find roots of emp in GF(16).

% Check that the primitive element gf(2,m) is
% really a root of the primitive polynomial for the field.
primpoly_int = double(e.prim_poly);
mval = e.m;
primpoly_vect = gf(de2bi(primpoly_int,mval+1,'left-msb'),mval);
containstwo = roots(primpoly_vect); % Output vector includes 2.
```

Converting Galois Field Array to Doubles

```
a = gf([1,0])
b = double(a.x) %a.x is in uint16
```

MATLAB returns the following:

```
a = GF(2) array.
```

```
Array elements =
```

```

                1      0
b =
    1      0

```

Speed and Nondefault Primitive Polynomials

“Specifying the Primitive Polynomial” on page 8-113 describes how to represent elements of a Galois field with respect to a primitive polynomial of your choice. This section describes how you can increase the speed of computations involving a Galois field array that uses a primitive polynomial other than the default primitive polynomial. The technique is recommended if you perform many such computations.

The mechanism for increasing the speed is a data file, `userGfTable.mat`, that some computational functions use to avoid performing certain computations repeatedly. To take advantage of this mechanism for your combination of field order (`m`) and primitive polynomial (`prim_poly`):

- 1 Navigate in the MATLAB application to a folder to which you have write permission. You can use either the `cd` function or the Current Folder feature to navigate.

- 2 Define `m` and `prim_poly` as workspace variables. For example:

```
m = 3; prim_poly = 13; % Examples of valid values
```

- 3 Invoke the `gfTable` function:

```
gfTable(m,prim_poly); % If you previously defined m and prim_poly
```

The function revises or creates `userGfTable.mat` in your current working folder to include data relating to your combination of field order and primitive polynomial. After you initially invest the time to invoke `gfTable`, subsequent computations using those values of `m` and `prim_poly` should be faster.

Note If you change your current working directory after invoking `gfTable`, you must place `userGfTable.mat` on your MATLAB path to ensure that MATLAB can see it. Do this by using the `addpath` command to prefix the directory containing `userGfTable.mat` to your MATLAB path. If you have multiple copies of `userGfTable.mat` on your path, use `which('userGfTable.mat', '-all')` to find out where they are and which one MATLAB is using.

To see how much `gftable` improves the speed of your computations, you can surround your computations with the `tic` and `toc` functions. See the `gftable` reference page for an example.

Selected Bibliography for Galois Fields

- [1] Blahut, Richard E., *Theory and Practice of Error Control Codes*, Reading, MA, Addison-Wesley, 1983, p. 105.
- [2] Lang, Serge, *Algebra*, Third Edition, Reading, MA, Addison-Wesley, 1993.
- [3] Lin, Shu, and Daniel J. Costello, Jr., *Error Control Coding: Fundamentals and Applications*, Englewood Cliffs, NJ, Prentice-Hall, 1983.
- [4] van Lint, J. H., *Introduction to Coding Theory*, New York, Springer-Verlag, 1982.
- [5] Wicker, Stephen B., *Error Control Systems for Digital Communication and Storage*, Upper Saddle River, NJ, Prentice Hall, 1995.

Galois Fields of Odd Characteristic

A *Galois field* is an algebraic field having p^m elements, where p is prime and m is a positive integer. This chapter describes how to work with Galois fields in which p is *odd*. To work with Galois fields having an even number of elements, see Galois Field Computations on page 8-107. The sections in this chapter are as follows.

- “Galois Field Terminology” on page 8-138
- “Representing Elements of Galois Fields” on page 8-139
- “Default Primitive Polynomials” on page 8-142
- “Converting and Simplifying Element Formats” on page 8-143
- “Arithmetic in Galois Fields” on page 8-146
- “Polynomials over Prime Fields” on page 8-149
- “Other Galois Field Functions” on page 8-152
- “Selected Bibliography for Galois Fields” on page 8-153

Galois Field Terminology

Throughout this section, p is an odd prime number and m is a positive integer.

Also, this document uses a few terms that are not used consistently in the literature. The definitions adopted here appear in van Lint [5].

- A *primitive element* of $GF(p^m)$ is a cyclic generator of the group of nonzero elements of $GF(p^m)$. This means that every nonzero element of the field can be expressed as the primitive element raised to some integer power. Primitive elements are called A throughout this section.
- A *primitive polynomial* for $GF(p^m)$ is the minimal polynomial of some primitive element of $GF(p^m)$. As a consequence, it has degree m and is irreducible.

Representing Elements of Galois Fields

- “Section Overview” on page 8-139
- “Exponential Format” on page 8-139
- “Polynomial Format” on page 8-140
- “List of All Elements of a Galois Field” on page 8-141
- “Nonuniqueness of Representations” on page 8-142

Section Overview

This section discusses how to represent Galois field elements using this toolbox's exponential on page 8-139 format and polynomial on page 8-140 format. It also describes a way to list all elements on page 8-141 of the Galois field, because some functions use such a list as an input argument. Finally, it discusses the nonuniqueness on page 8-142 of representations of Galois field elements.

The elements of $GF(p)$ can be represented using the integers from 0 to $p-1$.

When m is at least 2, $GF(p^m)$ is called an extension field. Integers alone cannot represent the elements of $GF(p^m)$ in a straightforward way. MATLAB technical computing software uses two main conventions for representing elements of $GF(p^m)$: the exponential format and the polynomial format.

Note Both the exponential format and the polynomial format are relative to your choice of a particular primitive element A of $GF(p^m)$.

Exponential Format

This format uses the property that every nonzero element of $GF(p^m)$ can be expressed as A^c for some integer c between 0 and p^m-2 . Higher exponents are not needed, because the

theory of Galois fields implies that every nonzero element of $GF(p^m)$ satisfies the equation $x^{q-1} = 1$ where $q = p^m$.

The use of the exponential format is shown in the table below.

Element of $GF(p^m)$	MATLAB Representation of the Element
0	-Inf
$A^0 = 1$	0
A^1	1
...	...
A^{q-2} where $q = p^m$	q-2

Although -Inf is the standard exponential representation of the zero element, all negative integers are equivalent to -Inf when used as *input* arguments in exponential format. This equivalence can be useful; for example, see the concise line of code at the end of the section “Default Primitive Polynomials” on page 8-142.

Note The equivalence of all negative integers and -Inf as exponential formats means that, for example, -1 does *not* represent A^{-1} , the multiplicative inverse of A. Instead, -1 represents the zero element of the field.

Polynomial Format

The polynomial format uses the property that every element of $GF(p^m)$ can be expressed as a polynomial in A with exponents between 0 and m-1, and coefficients in $GF(p)$. In the polynomial format, the element

$$A(1) + A(2) A + A(3) A^2 + \dots + A(m) A^{m-1}$$

is represented in MATLAB by the vector

$$[A(1) \ A(2) \ A(3) \ \dots \ A(m)]$$

Note The Galois field functions in this toolbox represent a polynomial as a vector that lists the coefficients in order of *ascending* powers of the variable. This is the opposite of the order that other MATLAB functions use.

List of All Elements of a Galois Field

Some Galois field functions in this toolbox require an argument that lists all elements of an extension field $GF(p^m)$. This is again relative to a particular primitive element A of $GF(p^m)$. The proper format for the list of elements is that of a matrix having p^m rows, one for each element of the field. The matrix has m columns, one for each coefficient of a power of A in the polynomial format shown in “Polynomial Format” on page 8-140 above. The first row contains only zeros because it corresponds to the zero element in $GF(p^m)$. If k is between 2 and p^m , then the k th row specifies the polynomial format of the element A^{k-2} .

The minimal polynomial of A aids in the computation of this matrix, because it tells how to express A^m in terms of lower powers of A . For example, the table below lists the elements of $GF(3^2)$, where A is a root of the primitive polynomial $2 + 2x + x^2$. This polynomial allows repeated use of the substitution

$$A^2 = -2 - 2A = 1 + A$$

when performing the computations in the middle column of the table.

Elements of GF(9)

Exponential Format	Polynomial Format	Row of MATLAB Matrix of Elements
A^{-Inf}	0	0 0
A^0	1	1 0
A^1	A	0 1
A^2	$1+A$	1 1
A^3	$A + A^2 = A + 1 + A = 1 + 2A$	1 2
A^4	$A + 2A^2 = A + 2 + 2A = 2$	2 0
A^5	$2A$	0 2
A^6	$2A^2 = 2 + 2A$	2 2
A^7	$2A + 2A^2 = 2A + 2 + 2A = 2 + A$	2 1

Example

An automatic way to generate the matrix whose rows are in the third column of the table above is to use the code below.

```
p = 3; m = 2;  
% Use the primitive polynomial 2 + 2x + x^2 for GF(9).  
prim_poly = [2 2 1];  
field = gftuple([-1:p^m-2]', prim_poly, p);
```

The `gftuple` function is discussed in more detail in “Converting and Simplifying Element Formats” on page 8-143.

Nonuniqueness of Representations

A given field has more than one primitive element. If two primitive elements have different minimal polynomials, then the corresponding matrices of elements will have their rows in a different order. If the two primitive elements share the same minimal polynomial, then the matrix of elements of the field is the same.

Note You can use whatever primitive element you want, as long as you understand how the inputs and outputs of Galois field functions depend on the choice of *some* primitive polynomial. It is usually best to use the same primitive polynomial throughout a given script or function.

Other ways in which representations of elements are not unique arise from the equations that Galois field elements satisfy. For example, an exponential format of 8 in GF(9) is really the same as an exponential format of 0, because $A^8 = 1 = A^0$ in GF(9). As another example, the substitution mentioned just before the table Elements of GF(9) shows that the polynomial format [0 0 1] is really the same as the polynomial format [1 1].

Default Primitive Polynomials

This toolbox provides a *default* primitive polynomial for each extension field. You can retrieve this polynomial using the `gfprimdf` function. The command

```
prim_poly = gfprimdf(m,p); % If m and p are already defined
```

produces the standard row-vector representation of the default minimal polynomial for GF(p^m).

For example, the command below shows that the default primitive polynomial for GF(9) is $2 + x + x^2$, *not* the polynomial used in “List of All Elements of a Galois Field” on page 8-141.

```
poly1=gfprimdf(2,3);
```

```
poly1 =  
      2      1      1
```

To generate a list of elements of $GF(p^m)$ using the default primitive polynomial, use the command

```
field = gftuple([-1:p^m-2]',m,p);
```

Converting and Simplifying Element Formats

- “Converting to Simplest Polynomial Format” on page 8-143
- “Example: Generating a List of Galois Field Elements” on page 8-145
- “Converting to Simplest Exponential Format” on page 8-145

Converting to Simplest Polynomial Format

The `gftuple` function produces the simplest polynomial representation of an element of $GF(p^m)$, given either an exponential representation or a polynomial representation of that element. This can be useful for generating the list of elements of $GF(p^m)$ that other functions require.

Using `gftuple` requires three arguments: one representing an element of $GF(p^m)$, one indicating the primitive polynomial that MATLAB technical computing software should use when computing the output, and the prime p . The table below indicates how `gftuple` behaves when given the first two arguments in various formats.

Behavior of `gftuple` Depending on Format of First Two Inputs

How to Specify Element	How to Indicate Primitive Polynomial	What <code>gftuple</code> Produces
Exponential format; $c = \text{any integer}$	Integer $m > 1$	Polynomial format of A^c , where A is a root of the <i>default</i> primitive polynomial for $\text{GF}(p^m)$
Example: <code>tp = gftuple(6,2,3); % c = 6 here</code>		
Exponential format; $c = \text{any integer}$	Vector of coefficients of primitive polynomial	Polynomial format of A^c , where A is a root of the <i>given</i> primitive polynomial
Example: <code>polynomial = gfprimdf(2,3); tp = gftuple(6,polynomial,3); % c = 6 here</code>		
Polynomial format of any degree	Integer $m > 1$	Polynomial format of degree $< m$, using <i>default</i> primitive polynomial for $\text{GF}(p^m)$ to simplify
Example: <code>tp = gftuple([0 0 0 0 0 0 1],2,3);</code>		
Polynomial format of any degree	Vector of coefficients of primitive polynomial	Polynomial format of degree $< m$, using the <i>given</i> primitive polynomial for $\text{GF}(p^m)$ to simplify
Example: <code>polynomial = gfprimdf(2,3); tp = gftuple([0 0 0 0 0 0 1],polynomial,3);</code>		

The four examples that appear in the table above all produce the same vector `tp = [2, 1]`, but their different inputs to `gftuple` correspond to the lines of the table. Each example expresses the fact that $A^6 = 2+A$, where A is a root of the (default) primitive polynomial $2 + x + x^2$ for $\text{GF}(3^2)$.

Example

This example shows how `gfconv` and `gftuple` combine to multiply two polynomial-format elements of $\text{GF}(3^4)$. Initially, `gfconv` multiplies the two polynomials, treating the primitive element as if it were a variable. This produces a high-order polynomial, which `gftuple` simplifies using the polynomial equation that the primitive element satisfies. The final result is the simplest polynomial format of the product.

```

p = 3; m = 4;
a = [1 2 0 1]; b = [2 2 1 2];
notsimple = gfconv(a,b,p) % a times b, using high powers of alpha
simple = gftuple(notsimple,m,p) %Highest exponent of alpha is m-1

```

The output is below.

```

notsimple =
      2      0      2      0      0      1      2

simple =
      2      1      0      1

```

Example: Generating a List of Galois Field Elements

This example applies the conversion functionality to the task of generating a matrix that lists all elements of a Galois field. A matrix that lists all field elements is an input argument in functions such as `gfadd` and `gfmul`. The variables `field1` and `field2` below have the format that such functions expect.

```

p = 5; % Or any prime number
m = 4; % Or any positive integer
field1 = gftuple([-1:p^m-2]',m,p);

prim_poly = gfprimd(m,p); % Or any primitive polynomial
% for GF(p^m)
field2 = gftuple([-1:p^m-2]',prim_poly,p);

```

Converting to Simplest Exponential Format

The same function `gftuple` also produces the simplest exponential representation of an element of $GF(p^m)$, given either an exponential representation or a polynomial representation of that element. To retrieve this output, use the syntax

```
[polyformat, expformat] = gftuple(...)
```

The input format and the output `polyformat` are as in the table Behavior of `gftuple` Depending on Format of First Two Inputs. In addition, the variable `expformat` contains the simplest exponential format of the element represented in `polyformat`. It is *simplest* in the sense that the exponent is either $-\infty$ or a number between 0 and p^m-2 .

Example

To recover the exponential format of the element $2 + A$ that the previous section considered, use the commands below. In this case, `polyformat` contains redundant information, while `expformat` contains the desired result.

```
[polyformat, expformat] = gftuple([2 1],2,3)
```

```
polyformat =
```

```
    2    1
```

```
expformat =
```

```
    6
```

This output appears at first to contradict the information in the table Elements of GF(9), but in fact it does not. The table uses a different primitive element; two plus that primitive element has the polynomial and exponential formats shown below.

```
prim_poly = [2 2 1];
```

```
[polyformat2, expformat2] = gftuple([2 1],prim_poly,3)
```

The output below reflects the information in the bottom line of the table.

```
polyformat2 =
```

```
    2    1
```

```
expformat2 =
```

```
    7
```

Arithmetic in Galois Fields

- “Section Overview” on page 8-146
- “Arithmetic in Prime Fields” on page 8-147
- “Arithmetic in Extension Fields” on page 8-147

Section Overview

You can add, subtract, multiply, and divide elements of Galois fields using the functions `gfadd`, `gfsub`, `gfmul`, and `gfdiv`, respectively. Each of these functions has a mode for prime fields on page 8-147 and a mode for extension fields on page 8-147.

Arithmetic in Prime Fields

Arithmetic in $GF(p)$ is the same as arithmetic modulo p . The functions `gfadd`, `gfmul`, `gfsub`, and `gfdiv` accept two arguments that represent elements of $GF(p)$ as integers between 0 and $p-1$. The third argument specifies p .

Example: Addition Table for $GF(5)$

The code below constructs an addition table for $GF(5)$. If a and b are between 0 and 4, then the element `gfp_add(a+1,b+1)` represents the sum $a+b$ in $GF(5)$. For example, `gfp_add(3,5) = 1` because $2+4$ is 1 modulo 5.

```
p = 5;
row = 0:p-1;
table = ones(p,1)*row;
gfp_add = gfadd(table,table',p)
```

The output for this example follows.

`gfp_add =`

0	1	2	3	4
1	2	3	4	0
2	3	4	0	1
3	4	0	1	2
4	0	1	2	3

Other values of p produce tables for different prime fields $GF(p)$. Replacing `gfadd` by `gfmul`, `gfsub`, or `gfdiv` produces a table for the corresponding arithmetic operation in $GF(p)$.

Arithmetic in Extension Fields

The same arithmetic functions can add elements of $GF(p^m)$ when $m > 1$, but the format of the arguments is more complicated than in the case above. In general, arithmetic in extension fields is more complicated than arithmetic in prime fields; see the works listed in “Selected Bibliography for Galois Fields” on page 8-153 for details about how the arithmetic operations work.

When working in extension fields, the functions `gfadd`, `gfmul`, `gfsub`, and `gfdiv` use the first two arguments to represent elements of $GF(p^m)$ in exponential format. The third argument, which is required, lists all elements of $GF(p^m)$ as described in “List of All Elements of a Galois Field” on page 8-141. The result is in exponential format.

Example: Addition Table for GF(9)

The code below constructs an addition table for $GF(3^2)$, using exponential formats relative to a root of the default primitive polynomial for $GF(9)$. If a and b are between -1 and 7 , then the element `gfpm_add(a+2,b+2)` represents the sum of A^a and A^b in $GF(9)$. For example, `gfpm_add(4,6) = 5` because

$$A^2 + A^4 = A^5$$

Using the fourth and sixth rows of the matrix `field`, you can verify that

$$A^2 + A^4 = (1 + 2A) + (2 + 0A) = 3 + 2A = 0 + 2A = A^5 \text{ modulo } 3.$$

```
p = 3; m = 2; % Work in GF(3^2).
field = gftuple([-1:p^m-2]',m,p); % Construct list of elements.
row = -1:p^m-2;
table = ones(p^m,1)*row;
gfpm_add = gfadd(table,table',field)
```

The output is below.

`gfpm_add =`

-Inf	0	1	2	3	4	5	6	7
0	4	7	3	5	-Inf	2	1	6
1	7	5	0	4	6	-Inf	3	2
2	3	0	6	1	5	7	-Inf	4
3	5	4	1	7	2	6	0	-Inf
4	-Inf	6	5	2	0	3	7	1
5	2	-Inf	7	6	3	1	4	0
6	1	3	-Inf	0	7	4	2	5
7	6	2	4	-Inf	1	0	5	3

Note If you used a different primitive polynomial, then the tables would look different. This makes sense because the ordering of the rows and columns of the tables was based on that particular choice of primitive polynomial and not on any natural ordering of the elements of $GF(9)$.

Other values of p and m produce tables for different extension fields $GF(p^m)$. Replacing `gfadd` by `gfmul`, `gfsub`, or `gfdiv` produces a table for the corresponding arithmetic operation in $GF(p^m)$.

Polynomials over Prime Fields

- “Section Overview” on page 8-149
- “Cosmetic Changes of Polynomials” on page 8-149
- “Polynomial Arithmetic” on page 8-150
- “Characterization of Polynomials” on page 8-150
- “Roots of Polynomials” on page 8-151

Section Overview

A polynomial over GF(p) is a polynomial whose coefficients are elements of GF(p). Communications Toolbox software provides functions for

- Changing polynomials in cosmetic on page 8-149 ways
- Performing polynomial arithmetic on page 8-150
- Characterizing polynomials as primitive or irreducible on page 8-150
- Finding roots on page 8-151 of polynomials in a Galois field

Note The Galois field functions in this toolbox represent a polynomial over GF(p) for odd values of p as a vector that lists the coefficients in order of *ascending* powers of the variable. This is the opposite of the order that other MATLAB functions use.

Cosmetic Changes of Polynomials

To display the traditionally formatted polynomial that corresponds to a row vector containing coefficients, use `gfpretty`. To truncate a polynomial by removing all zero-coefficient terms that have exponents *higher* than the degree of the polynomial, use `gftrunc`. For example,

```
polynom = gftrunc([1 20 394 10 0 0 29 3 0 0])
gfpretty(polynom)
```

The output is below.

```
polynom =
```

```
1    20    394    10    0    0    29    3
```

```
1 + 20 X + 394 X2 + 10 X3 + 29 X6 + 3 X7
```

Note If you do not use a fixed-width font, then the spacing in the display might not look correct.

Polynomial Arithmetic

The functions `gfadd` and `gfsub` add and subtract, respectively, polynomials over $GF(p)$. The `gfconv` function multiplies polynomials over $GF(p)$. The `gfdeconv` function divides polynomials in $GF(p)$, producing a quotient polynomial and a remainder polynomial. For example, the commands below show that $2 + x + x^2$ times $1 + x$ over the field $GF(3)$ is $2 + 2x^2 + x^3$.

```
a = gfconv([2 1 1],[1 1],3)
[quot, remd] = gfdeconv(a,[2 1 1],3)
```

The output is below.

```
a =
    2     0     2     1

quot =
    1     1

remd =
    0
```

The previously discussed functions `gfadd` and `gfsub` add and subtract, respectively, polynomials. Because it uses a vector of coefficients to represent a polynomial, MATLAB does not distinguish between adding two polynomials and adding two row vectors elementwise.

Characterization of Polynomials

Given a polynomial over $GF(p)$, the `gfprimck` function determines whether it is irreducible and/or primitive. By definition, if it is primitive then it is irreducible; however, the reverse is not necessarily true. The `gfprimdf` and `gfprimfd` functions return primitive polynomials.

Given an element of $GF(p^m)$, the `gfminpol` function computes its minimal polynomial over $GF(p)$.

Example

For example, the code below reflects the irreducibility of all minimal polynomials. However, the minimal polynomial of a nonprimitive element is not a primitive polynomial.

```
p = 3; m = 4;
% Use default primitive polynomial here.

prim_poly = gfminpol(1,m,p);
ckprim = gfprimck(prim_poly,p);
% ckprim = 1, since prim_poly represents a primitive polynomial.

notprimpoly = gfminpol(5,m,p);
cknotprim = gfprimck(notprimpoly,p);
% cknotprim = 0 (irreducible but not primitive)
% since alpha^5 is not a primitive element when p = 3.

ckreducible = gfprimck([0 1 1],p);
% ckreducible = -1 since the polynomial is reducible.
```

Roots of Polynomials

Given a polynomial over $GF(p)$, the `gfroots` function finds the roots of the polynomial in a suitable extension field $GF(p^m)$. There are two ways to tell MATLAB the degree m of the extension field $GF(p^m)$, as shown in the following table.

Formats for Second Argument of `gfroots`

Second Argument	Represents
A positive integer	m as in $GF(p^m)$. MATLAB uses the default primitive polynomial in its computations.
A row vector	A primitive polynomial for $GF(p^m)$. Here m is the degree of this primitive polynomial.

Example: Roots of a Polynomial in $GF(9)$

The code below finds roots of the polynomial $1 + x^2 + x^3$ in $GF(9)$ and then checks that they are indeed roots. The exponential format of elements of $GF(9)$ is used throughout.

```
p = 3; m = 2;
field = gftuple([-1:p^m-2]',m,p); % List of all elements of GF(9)
% Use default primitive polynomial here.
polynomial = [1 0 1 1]; % 1 + x^2 + x^3
```

```
rts =groots(polynomial,m,p) % Find roots in exponential format
% Check that each one is actually a root.
for ii = 1:3
    root = rts(ii);
    rootsquared = gfmul(root,root,field);
    rootcubed = gfmul(root,rootsquared,field);
    answer(ii)= gfadd(gfadd(0,rootsquared,field),rootcubed,field);
    % Recall that 1 is really alpha to the zero power.
    % If answer = -Inf, then the variable root represents
    % a root of the polynomial.
end
answer
```

The output shows that A^0 (which equals 1), A^5 , and A^7 are roots.

```
roots =
```

```
    0
    5
    7
```

```
answer =
```

```
 -Inf  -Inf  -Inf
```

See the reference page for `groots` to see how `groots` can also provide you with the polynomial formats of the roots and the list of all elements of the field.

Other Galois Field Functions

See the online reference pages for information about these other Galois field functions in Communications Toolbox software:

- `gfcosets`, which produces cyclotomic cosets
- `gffilter`, which filters data using $GF(p)$ polynomials
- `gfprimfd`, which finds primitive polynomials
- `gfrank`, which computes the rank of a matrix over $GF(p)$
- `gfrepconv`, which converts one binary polynomial representation to another

Selected Bibliography for Galois Fields

- [1] Blahut, Richard E., *Theory and Practice of Error Control Codes*, Reading, Mass., Addison-Wesley, 1983.
- [2] Lang, Serge, *Algebra*, Third Edition, Reading, Mass., Addison-Wesley, 1993.
- [3] Lin, Shu, and Daniel J. Costello, Jr., *Error Control Coding: Fundamentals and Applications*, Englewood Cliffs, N.J., Prentice-Hall, 1983.
- [4] van Lint, J. H., *Introduction to Coding Theory*, New York, Springer-Verlag, 1982.

Interleaving

In this section...
“Block Interleaving” on page 8-154
“Convolutional Interleaving” on page 8-160
“Selected Bibliography for Interleaving” on page 8-171

Block Interleaving

- “Block Interleaving Features” on page 8-154
- “Improve Error Rate Using Block Interleaving in MATLAB” on page 8-156
- “Improve Error Rate Using Block Interleaving in Simulink” on page 8-157

Block Interleaving Features

A block interleaver accepts a set of symbols and rearranges them, without repeating or omitting any of the symbols in the set. The number of symbols in each set is fixed for a given interleaver. The interleaver's operation on a set of symbols is independent of its operation on all other sets of symbols.

An interleaver permutes symbols according to a mapping. A corresponding deinterleaver uses the inverse mapping to restore the original sequence of symbols. Interleaving and deinterleaving can be useful for reducing errors caused by burst errors in a communication system.

Each interleaver function has a corresponding deinterleaver function. In typical usage of the interleaver/deinterleaver pairs, the inputs of the deinterleaver match those of the interleaver, except for the data being rearranged.

A block interleaver accepts a set of symbols and rearranges them, without repeating or omitting any of the symbols in the set. The number of symbols in each set is fixed for a given interleaver.

The set of block interleavers in this toolbox includes a general block interleaver as well as several special cases. Each special-case interleaver function uses the same computational code that the general block interleaver function uses, but provides a syntax that is more suitable for the special case. The interleaver functions are described below.

Type of Interleaver	Interleaver Function	Description
General block interleaver	<code>intrlv</code>	Uses the permutation table given explicitly as an input argument.
Algebraic interleaver	<code>algintrlv</code>	Derives a permutation table algebraically, using the Takeshita-Costello or Welch-Costas method. These methods are described in [4].
Helical scan interleaver	<code>helscanintrlv</code>	Fills a matrix with data row by row and then sends the matrix contents to the output in a helical fashion.
Matrix interleaver	<code>matintrlv</code>	Fills a matrix with data elements row by row and then sends the matrix contents to the output column by column.
Random interleaver	<code>randintrlv</code>	Chooses a permutation table randomly using the initial state input that you provide.

Types of Block Interleavers

The set of block interleavers in this library includes a general interleaver/deinterleaver pair as well as several special cases. Each special-case block uses the same computational code that its more general counterpart uses, but provides an interface that is more suitable for the special case.

The Matrix Interleaver block accomplishes block interleaving by filling a matrix with the input symbols row by row and then sending the matrix contents to the output port column by column. For example, if the interleaver uses a 2-by-3 matrix to do its internal computations, then for an input of [1 2 3 4 5 6], the block produces an output of [1 4 2 5 3 6].

The Random Interleaver block chooses a permutation table randomly using the **Initial seed** parameter that you provide in the block mask. By using the same **Initial seed** value in the corresponding Random Deinterleaver block, you can restore the permuted symbols to their original ordering.

The Algebraic Interleaver block uses a permutation table that is algebraically derived. It supports Takeshita-Costello interleavers and Welch-Costas interleavers. These interleavers are described in [4].

Improve Error Rate Using Block Interleaving in MATLAB

The following example illustrates how an interleaver improves the error rate in a communication system whose channel produces a burst of errors. A random interleaver rearranges the bits of numerous codewords before two adjacent codewords are each corrupted by three errors.

Three errors exceed the error-correction capability of the Hamming code. However, the example shows that when the Hamming code is combined with an interleaver, this system is able to recover the original message despite the 6-bit burst of errors. The improvement in performance occurs because the interleaving effectively spreads the errors among different codewords so that the number of errors per codeword is within the error-correction capability of the code.

```
st1 = 27221; st2 = 4831; % States for random number generator
n = 7; k = 4; % Parameters for Hamming code
msg = randi([0 1],k*500,1); % Data to encode
code = encode(msg,n,k,'hamming/binary'); % Encoded data
% Create a burst error that will corrupt two adjacent codewords.
errors = zeros(size(code)); errors(n-2:n+3) = [1 1 1 1 1 1];

% With Interleaving
%-----
inter = randintrlv(code,st2); % Interleave.
inter_err = bitxor(inter,errors); % Include burst error.
deinter = randdeintrlv(inter_err,st2); % Deinterleave.
decoded = decode(deinter,n,k,'hamming/binary'); % Decode.
disp('Number of errors and error rate, with interleaving:');
[number_with,rate_with] = biterr(msg,decoded) % Error statistics

% Without Interleaving
%-----
code_err = bitxor(code,errors); % Include burst error.
decoded = decode(code_err,n,k,'hamming/binary'); % Decode.
disp('Number of errors and error rate, without interleaving:');
[number_without,rate_without] = biterr(msg,decoded) % Error statistics
```

The output from the example follows.

Number of errors and error rate, with interleaving:

number_with =

0

rate_with =

0

Number of errors and error rate, without interleaving:

number_without =

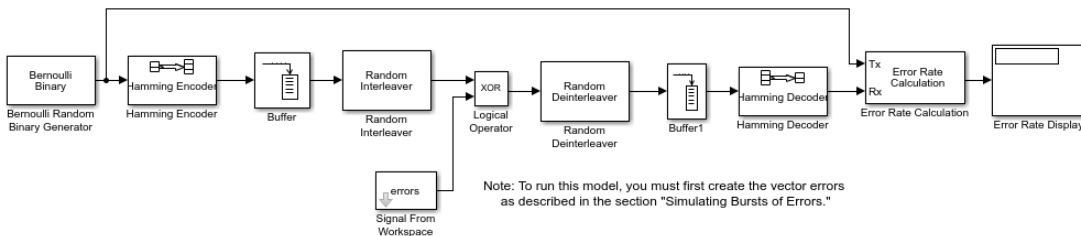
4

rate_without =

0.0020

Improve Error Rate Using Block Interleaving in Simulink

The following example shows how to use an interleaver to improve the error rate when the channel produces bursts of errors.



Before running the model, you must create a binary vector that simulates bursts of errors, as described in "Improve Error Rate Using Block Interleaving in Simulink" on page 8-157. The Signal From Workspace block imports this vector from the MATLAB workspace into the model, where the Logical Operator block performs an XOR of the vector with the signal.

To open the completed model, type `doc_interleaver` at the MATLAB command line. To build the model, gather and configure these blocks:

- Bernoulli Binary Generator, in the Random Data Sources sublibrary of the Comm Sources library
 - Check the box next to **Frame-based outputs**.
 - Set **Samples per frame** to 4.
- Hamming Encoder, in the Block sublibrary of the Error Detection and Correction library. Use default parameters
- Buffer, in the Buffers sublibrary of the Signal Management library in DSP System Toolbox
 - Set **Output buffer size (per channel)** to 84.
- Random Interleaver, in the Block sublibrary of the Interleaving library in Communications Toolbox
 - Set **Number of elements** to 84.
- Logical Operator, in the Simulink Math Operations library
 - Set **Operator** to XOR.
- Signal From Workspace, in the Sources library of the DSP System Toolbox product
 - Set **Signal** to errors.
 - Set **Sample time** to 4/7.
 - Set **Samples per frame** to 84.
- Random Deinterleaver, in the Block sublibrary of the Interleaving library in Communications Toolbox
 - Set **Number of elements** to 84.
- Buffer, in the Buffers sublibrary of the Signal Management library in DSP System Toolbox
 - Set **Output buffer size (per channel)** to 7.
- Hamming Decoder, in the Block sublibrary of the Error Detection and Correction library. Use default parameters.
- Error Rate Calculation, in the Comm Sinks library

- Set **Receive delay** to $(4/7)*84$.
- Set **Computation delay** to 100.
- Set **Output data** to Port.
- Display, in the Simulink Sinks library. Use default parameters.

Click the **Simulation** menu and select **Model Configuration parameters**. Set **Stop time** to `length(errors)`.

Creating the Vector of Errors

Before running the model, use the following code to create a binary vector in the MATLAB workspace. The model uses this vector to simulate bursts of errors. The vector contains blocks of three 1s, representing bursts of errors, at random intervals. The distance between two consecutive blocks of 1s is a random integer between 1 and 80.

```
errors=zeros(1,10^4);
n=1;
while n<10^4-80;
n=n+floor(79*rand(1))+3;
errors(n:n+2)=[1 1 1];
end
```

To determine the ratio of the number of 1s to the total number of symbols in the vector `errors` enter

```
sum(errors)/length(errors)
```

Your answer should be approximately 3/43, or .0698, since after each sequence of three 1s, the expected distance to the next sequence of 1s is 40. Consequently, you expect to see three 1s in 43 terms of the sequence. If there were no error correction in the model, the bit error rate would be approximately .0698.

When you run a simulation with the model, the error rate is approximately .019, which shows the improvement due to error correction and interleaving. You can see the effect of interleaving by deleting the Random Interleaver and Random Deinterleaver blocks from the model, connecting the lines, and running another simulation. The bit error rate is higher without interleaving because the Hamming code can only correct one error in each codeword.

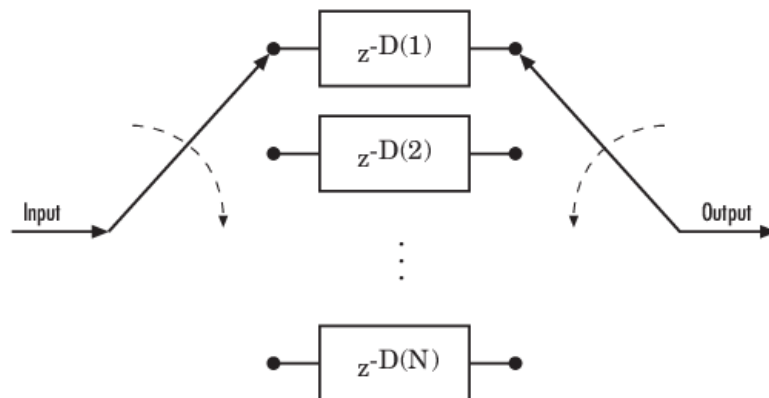
Convolutional Interleaving

- “Convolutional Interleaving Features” on page 8-160
- “Delays of Convolutional Interleavers” on page 8-162
- “Convolutional Interleaving and Deinterleaving Using a Sequence of Consecutive Integers in MATLAB” on page 8-166
- “Convolutional Interleaving and Deinterleaving Using a Sequence of Consecutive Integers in Simulink” on page 8-169

Convolutional Interleaving Features

A convolutional interleaver consists of a set of shift registers, each with a fixed delay. In a typical convolutional interleaver, the delays are nonnegative integer multiples of a fixed integer (although a general multiplexed interleaver allows unrestricted delay values). Each new symbol from an input vector feeds into the next shift register and the oldest symbol in that register becomes part of the output vector. A convolutional interleaver has memory; that is, its operation depends not only on current symbols but also on previous symbols.

The schematic below depicts the structure of a general convolutional interleaver by showing the set of shift registers and their delay values $D(1)$, $D(2), \dots, D(N)$. The k th shift register holds $D(k)$ symbols, where $k = 1, 2, \dots, N$. The convolutional interleaving functions in this toolbox have input arguments that indicate the number of shift registers and the delay for each shift register.



Communications Toolbox implements convolutional interleaving functionality using Simulink blocks, System objects, and MATLAB functions.

The set of convolutional interleavers in this product includes a general interleaver/deinterleaver pair as well as several special cases. Each special-case function uses the same computational code that its more general counterpart uses, but provides a syntax that is more suitable for the special case. The special cases are described below.

Type of Interleaver	Interleaving Function	Description
General multiplexed interleaver	<code>muxintrlv</code>	Allows unrestricted delay values for the set of shift registers.
Convolutional interleaver	<code>convintrlv</code>	The delay values for the set of shift registers are nonnegative integer multiples of a fixed integer that you specify.
Helical interleaver	<code>helintrlv</code>	Fills an array with input symbols in a helical fashion and empties the array row by row.

The `helscanintrlv` function and the `helintrlv` function both use a helical array for internal computations. However, the two functions have some important differences:

- `helintrlv` uses an unlimited-row array, arranges input symbols in the array along columns, outputs some symbols that are not from the current input, and leaves some input symbols in the array without placing them in the output.
- `helscanintrlv` uses a fixed-size matrix, arranges input symbols in the array across rows, and outputs all the input symbols without using any default values or values from a previous call.

Types of Convolutional Interleavers

The set of convolutional interleavers in this library includes a general interleaver/deinterleaver pair as well as several special cases. Each special-case block uses the same computational code that its more general counterpart uses, but provides an interface that is more suitable for the special case.

The most general block in this library is the General Multiplexed Interleaver block, which allows arbitrary delay values for the set of shift registers. To implement the preceding schematic using this block, use an **Interleaver delay** parameter of [D(1); D(2); ...; D(N)].

More specific is the Convolutional Interleaver block, in which the delay value for the k th shift register is $(k-1)$ times the block's **Register length step** parameter. The number of shift registers in this block is the value of the **Rows of shift registers** parameter.

Finally, the Helical Interleaver block supports a special case of convolutional interleaving that fills an array with symbols in a helical fashion and empties the array row by row. To configure this interleaver, use the **Number of columns of helical array** parameter to set the width of the array, and use the **Group size** and **Helical array step size** parameters to determine how symbols are placed in the array. See the reference page for the Helical Interleaver block for more details and an example.

Delays of Convolutional Interleavers

After a sequence of symbols passes through a convolutional interleaver and a corresponding convolutional deinterleaver, the restored sequence lags behind the original sequence. The delay, measured in symbols, between the original and restored sequences is indicated in the table below. The variable names in the second column (`delay`, `nrows`, `slope`, `col`, `ngrp`, and `stp`) refer to the inputs named on each function's reference page.

Delays of Interleaver/Deinterleaver Pairs

Interleaver/Deinterleaver Pair	Delay Between Original and Restored Sequences
<code>muxintrlv</code> , <code>muxdeintrlv</code>	<code>length(delay)*max(delay)</code>
<code>convintrlv</code> , <code>convdeintrlv</code>	<code>nrows*(nrows-1)*slope</code>
<code>helintrlv</code> , <code>heldeintrlv</code>	<code>col*ngrp*ceil(stp*(col-1)/ngrp)</code>

Delays of Convolutional Interleavers

After a sequence of symbols passes through a convolutional interleaver and a corresponding convolutional deinterleaver, the restored sequence lags behind the original sequence. The delay, measured in symbols, between the original and restored sequences is

Number of shift registers \times Maximum delay among all shift registers

for the most general multiplexed interleaver. If your model incurs an additional delay between the interleaver output and the deinterleaver input, the restored sequence lags behind the original sequence by the sum of the additional delay and the amount in the preceding formula.

Note For proper synchronization, the delay in your model between the interleaver output and the deinterleaver input must be an integer multiple of the number of shift registers. You can use the DSP System Toolbox Delay block to adjust delays manually, if necessary.

Convolutional Interleaver block

In the special case implemented by the Convolutional Interleaver/Convolutional Deinterleaver pair, the number of shift registers is the **Rows of shift registers** parameter, while the maximum delay among all shift registers is

$$B \times (N-1) \quad (N-1)$$

where B is the **Register length step** parameter and N is the **Rows of shift registers** parameter.

Helical Interleaver block

In the special case implemented by the Helical Interleaver/Helical Deinterleaver pair, the delay between the restored sequence and the original sequence is

$$CN \left\lfloor \frac{s(C-1)}{N} \right\rfloor$$

where C is the **Number of columns in helical array** parameter, N is the **Group size** parameter, and s is the **Helical array step size** parameter.

Effect of Delays on Recovery of Convolutionally Interleaved Data Using MATLAB

If you use a convolutional interleaver followed by a corresponding convolutional deinterleaver, then a nonzero delay means that the recovered data (that is, the output from the deinterleaver) is not the same as the original data (that is, the input to the interleaver). If you compare the two data sets directly, then you must take the delay into account by using appropriate truncating or padding operations.

Here are some typical ways to compensate for a delay of D in an interleaver/deinterleaver pair:

- Interleave a version of the original data that is padded with D extra symbols at the end. Before comparing the original data with the recovered data, omit the first D symbols of the recovered data. In this approach, all the original symbols appear in the recovered data.
- Before comparing the original data with the recovered data, omit the last D symbols of the original data and the first D symbols of the recovered data. In this approach, some of the original symbols are left in the deinterleaver's shift registers and do not appear in the recovered data.

The following code illustrates these approaches by computing a symbol error rate for the interleaving/deinterleaving operation.

```
x = randi([0 63],20,1); % Original data
nrows = 3; slope = 2; % Interleaver parameters
D = nrows*(nrows-1)*slope; % Delay of interleaver/deinterleaver pair
hInt = comm.ConvolutionalInterleaver('NumRegisters', nrows, ...
    'RegisterLengthStep', slope);
hDeint = comm.ConvolutionalDeinterleaver('NumRegisters', nrows, ...
    'RegisterLengthStep', slope);

% First approach.
x_padded = [x; zeros(D,1)]; % Pad x at the end before interleaving.
a1 = step(hInt, x_padded); % Interleave padded data.

b1 = step(hDeint, a1)
% Omit input padding and the first D symbols of the recovered data and
% compare
servec1 = step(comm.ErrorRate('ReceiveDelay',D),x_padded,b1);
ser1 = servec1(1)

% Second approach.
release(hInt); release(hDeint)
a2 = step(hInt,x); % Interleave original data.
b2 = step(hDeint,a2)
% Omit the last D symbols of the original data and the first D symbols of
% the recovered data and compare.
servec2 = step(comm.ErrorRate('ReceiveDelay',D),x,b2);
ser2 = servec2(1)
```

The output is shown below. The zero values of `ser1` and `ser2` indicate that the script correctly aligned the original and recovered data before computing the symbol error rates. However, notice from the lengths of `b1` and `b2` that the two approaches to alignment result in different amounts of deinterleaved data.

```
b1 =
    0
    0
    0
    0
    0
    0
    0
    0
```

```
0
0
0
0
0
59
42
1
28
52
54
43
8
56
5
35
37
48
17
28
62
10
31
61
39
```

```
ser1 =
```

```
0
```

```
b2 =
```

```
0
0
0
0
0
0
0
0
0
0
0
```

```
0
0
59
42
1
28
52
54
43
8

ser2 =

0
```

Combining Interleaving Delays and Other Delays

If you use convolutional interleavers in a script that incurs an additional delay, d , between the interleaver output and the deinterleaver input (for example, a delay from a filter), then the restored sequence lags behind the original sequence by the sum of d and the amount from the table Delays of Interleaver/Deinterleaver Pairs on page 8-162. In this case, d must be an integer multiple of the number of shift registers, or else the convolutional deinterleaver cannot recover the original symbols properly. If d is not naturally an integer multiple of the number of shift registers, then you can adjust the delay manually by padding the vector that forms the input to the deinterleaver.

Convolutional Interleaving and Deinterleaving Using a Sequence of Consecutive Integers in MATLAB

The example below illustrates convolutional interleaving and deinterleaving using a sequence of consecutive integers. It also illustrates the inherent delay of the interleaver/deinterleaver pair.

```
x = [1:10]'; % Original data
delay = [0; 1; 2]; % Set delays of three shift registers.
hInt = comm.MultplexedInterleaver('Delay', delay);
hDeint = comm.MultplexedDeinterleaver('Delay', delay);
y = step(hInt,x) % Interleave.
z = step(hDeint,y) % Deinterleave.
```

In this example, the `muxintrlv` function initializes the three shift registers to the values `[]`, `[0]`, and `[0 0]`, respectively. Then the function processes the input data `[1:10]'`, performing internal computations as indicated in the table below.

Current Input	Current Shift Register	Current Output	Contents of Shift Registers
1	1	1	[1] [0] [0 0]
2	2	0	[1] [2] [0 0]
3	3	0	[1] [2] [0 3]
4	1	4	[1] [2] [0 3]
5	2	2	[1] [5] [0 3]
6	3	0	[1] [5] [3 6]
7	1	7	[1] [5] [3 6]
8	2	5	[1] [8] [3 6]
9	3	3	[1] [8] [6 9]
10	1	10	[1] [8] [6 9]

The output from the example is below.

y =

1
0
0
4
2
0
7
5
3
10

state_y =

value: {3x1 cell}
index: 2

z =

0
0
0
0
0
0
1
2
3
4

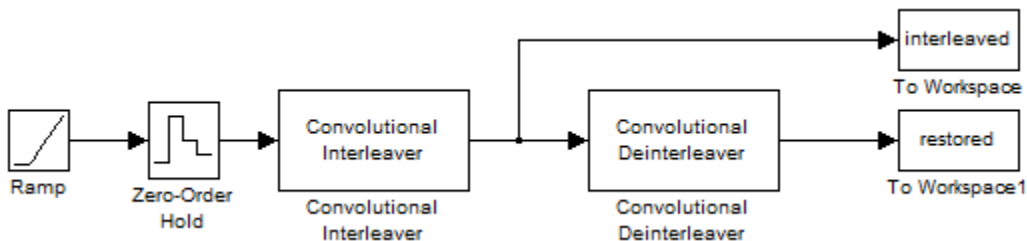
Notice that the “Current Output” column of the table above agrees with the values in the vector `y`. Also, the last row of the table above indicates that the last shift register processed for the given data set is the first shift register. This agrees with the value of 2 for `state_y.index`, which indicates that any additional input data would be directed to the second shift register. You can optionally check that the state values listed in `state_y.value` match the “Contents of Shift Registers” entry in the last row of the table by typing `state_y.value{:}` in the Command Window after executing the example.

Another feature to notice about the example output is that `z` contains six zeros at the beginning before containing any of the symbols from the original data set. The six zeros illustrate that the delay of this convolutional interleaver/deinterleaver pair is

$\text{length}(\text{delay}) * \text{max}(\text{delay}) = 3 * 2 = 6$. For more information about delays, see "Delays of Convolutional Interleavers" on page 8-162.

Convolutional Interleaving and Deinterleaving Using a Sequence of Consecutive Integers in Simulink

The example below illustrates convolutional interleaving and deinterleaving using a sequence of consecutive integers. It also illustrates the inherent delay and the effect of the interleaving blocks' initial conditions.



Open the model by typing `doc_convinterleaver` at the MATLAB command line. To build the model, gather and configure these blocks:

- Ramp, in the Simulink Sources library. Use default parameters.
- Zero-Order Hold, in the Simulink Discrete library. Use default parameters.
- Convolutional Interleaver
 - Set **Rows of shift registers** to 3.
 - Set **Initial conditions** to `[-1 -2 -3]'`.
- Convolutional Deinterleaver
 - Set **Rows of shift registers** to 3.
 - Set **Initial conditions** to `[-1 -2 -3]'`.
- Two copies of To Workspace, in the Simulink Sinks library
 - Set **Variable name** to `interleaved` and `restored`, respectively, in the two copies of this block.
 - Set **Save format** to Array in each of the two copies of this block.

Connect the blocks as shown in the preceding diagram. From the model window's **Simulation** menu, select **Model Configuration parameters**. In the Configuration

Parameters dialog box, set **Stop time** to 20. Run the simulation and execute the following command:

```
comparison = [[0:20]', interleaved, restored]
```

```
comparison =
```

0	0	-1
1	-2	-2
2	-3	-3
3	3	-1
4	-2	-2
5	-3	-3
6	6	-1
7	1	-2
8	-3	-3
9	9	-1
10	4	-2
11	-3	-3
12	12	0
13	7	1
14	2	2
15	15	3
16	10	4
17	5	5
18	18	6
19	13	7
20	8	8

In this output, the first column contains the original symbol sequence. The second column contains the interleaved sequence, while the third column contains the restored sequence.

The negative numbers in the interleaved and restored sequences come from the interleaving blocks' initial conditions, not from the original data. The first of the original symbols appears in the restored sequence only after a delay of 12 symbols. The delay of the interleaver-deinterleaver combination is the product of the number of shift registers (3) and the maximum delay among all shift registers (4).

For a similar example that also indicates the contents of the shift registers at each step of the process, see “Convolutional Interleaving and Deinterleaving Using a Sequence of Consecutive Integers in MATLAB” on page 8-166.

Selected Bibliography for Interleaving

- [1] Berlekamp, E.R., and P. Tong, "Improved Interleavers for Algebraic Block Codes," U. S. Patent 4559625, Dec. 17, 1985.
- [2] Clark, George C. Jr., and J. Bibb Cain, *Error-Correction Coding for Digital Communications*, New York, Plenum Press, 1981.
- [3] Forney, G. D. Jr., "Burst-Correcting Codes for the Classic Bursty Channel," *IEEE Transactions on Communications*, vol. COM-19, October 1971, pp. 772-781.
- [4] Heegard, Chris and Stephen B. Wicker, *Turbo Coding*, Boston, Kluwer Academic Publishers, 1999.
- [5] Ramsey, J. L., "Realization of Optimum Interleavers," *IEEE Transactions on Information Theory*, IT-16 (3), May 1970, pp. 338-345.
- [6] Takeshita, O. Y. and D. J. Costello, Jr., "New Classes Of Algebraic Interleavers for Turbo-Codes," *Proc. 1998 IEEE International Symposium on Information Theory*, Boston, Aug. 16-21, 1998. pp. 419.

Digital Modulation

In most media for communication, only a fixed range of frequencies is available for transmission. One way to communicate a message signal whose frequency spectrum does not fall within that fixed frequency range, or one that is otherwise unsuitable for the channel, is to alter a transmittable signal according to the information in your message signal. This alteration is called *modulation*, and it is the modulated signal that you transmit. The receiver then recovers the original signal through a process called *demodulation*.

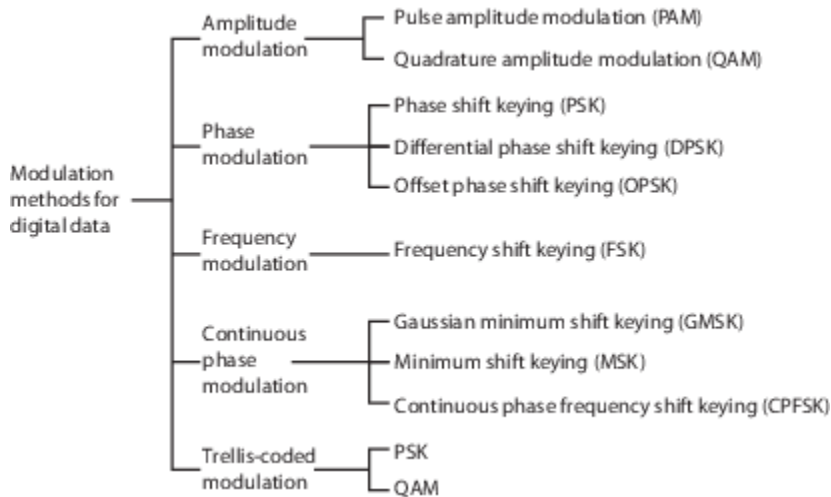
In this section...
“Digital Modulation Features” on page 8-172
“Signals and Delays” on page 8-177
“PM Modulation” on page 8-184
“AM Modulation” on page 8-185
“CPM Modulation” on page 8-189
“Exact LLR Algorithm” on page 8-192
“Approximate LLR Algorithm” on page 8-193
“Delays in Digital Modulation” on page 8-193
“Selected Bibliography for Digital Modulation” on page 8-196

Digital Modulation Features

- “Modulation Techniques” on page 8-172
- “Baseband and Passband Simulation” on page 8-176
- “Modulation Terminology” on page 8-176
- “Representing Digital Signals” on page 8-177

Modulation Techniques

The Communications Toolbox supports these modulation techniques for digital data. All the methods at the far right are implemented in library blocks.



Like analog modulation, digital modulation alters a transmittable signal according to the information in a message signal. However, for digital modulation the message signal is restricted to a finite set. Modulation functions output the complex envelope of the modulated signal. Using the Communications Toolbox, you can modulate or demodulate signals using various digital modulation techniques, and plot signal constellations.

Note Unless otherwise indicated, the modulation and demodulation functions do not perform pulse shaping or filtering. See Combining Pulse Shaping and Filtering with Modulation on page 8-188 for more information about filtering.

The available methods of modulation depend on whether the input signal is analog or digital. For a list of the modulation techniques the Communications Toolbox supports see Digital Baseband Modulation, Analog Baseband Modulation, and Analog Passband Modulation.

Accessing Digital Modulation Blocks

Open the Modulation library by double-clicking the icon in the main block library. Then open the Digital Baseband sublibrary by double-clicking its icon in the Modulation library.

The Digital Baseband library has sublibraries of its own. Open each of these sublibraries by double-clicking the icon listed in the table below.

Kind of Modulation	Icon in Digital Baseband Library
Amplitude modulation	AM
Phase modulation	PM
Frequency modulation	FM
Continuous phase modulation	CPM
Trellis-coded modulation	TCM

Some digital modulation sublibraries contain blocks that implement specific modulation techniques. These specific-case modulation blocks use the same computational code that their general counterparts use, but provide an interface that is either simpler or more suitable for the specific case. This table lists general modulators along with the conditions under which a general modulator is equivalent to a specific modulator. The situation is analogous for demodulators.

General and Specific Blocks

General Modulator	General Modulator Conditions	Specific Modulator
General QAM Modulator Baseband	Predefined constellation containing 2^K points on a rectangular lattice. K is the modulation order.	Rectangular QAM Modulator Baseband
M-PSK Modulator Baseband	M-ary number parameter is 2.	BPSK Modulator Baseband
	M-ary number parameter is 4.	QPSK Modulator Baseband
M-DPSK Modulator Baseband	M-ary number parameter is 2.	DBPSK Modulator Baseband
	M-ary number parameter is 4.	DQPSK Modulator Baseband
CPM Modulator Baseband	M-ary number parameter is 2, Frequency pulse shape parameter is Gaussian.	GMSK Modulator Baseband
	M-ary number parameter is 2, Frequency pulse shape parameter is Rectangular, Pulse length parameter is 1.	MSK Modulator Baseband
	Frequency pulse shape parameter is Rectangular, Pulse length parameter is 1.	CPFSK Modulator Baseband
General TCM Encoder	Predefined signal constellation containing 2^K points on a rectangular lattice.	Rectangular QAM TCM Encoder
	Predefined signal constellation containing 2^K points on a circle.	M-PSK TCM Encoder

Furthermore, the CPFSK Modulator Baseband block is similar to the M-FSK Modulator Baseband block, when the M-FSK block uses continuous phase transitions. However, the M-FSK features of this product differ from the CPFSK features in their mask interfaces and in the demodulator implementations.

Baseband and Passband Simulation

For a given modulation technique, two ways to simulate modulation techniques are called *baseband* and *passband*. Baseband simulation, also known as the *lowpass equivalent method*, requires less computation. The Communications Toolbox supports baseband simulation for digital modulation and passband simulation for analog modulation.

Baseband Modulated Signals Defined

If you use baseband modulation to produce the complex envelope y of the modulation of a message signal x , then y is a *complex-valued* signal that is related to the output of a passband modulator. If the modulated signal has the waveform

$$Y_1(t)\cos(2\pi f_c t + \theta) - Y_2(t)\sin(2\pi f_c t + \theta),$$

where f_c is the carrier frequency and θ is the carrier signal's initial phase, then a baseband simulation recognizes that this equals the real part of

$$[(Y_1(t) + jY_2(t))e^{j\theta}]\exp(j2\pi f_c t).$$

and models only the part inside the square brackets. Here j is the square root of -1. The complex vector y is a sampling of the complex signal

$$(Y_1(t) + jY_2(t))e^{j\theta}.$$

If you prefer to work with passband signals instead of baseband signals, then you can build functions that convert between the two. Be aware that passband modulation tends to be more computationally intensive than baseband modulation because the carrier signal typically needs to be sampled at a high rate.

Modulation Terminology

Modulation is a process by which a *carrier signal* is altered according to information in a *message signal*. The *carrier frequency*, F_c , is the frequency of the carrier signal. The *sampling rate* is the rate at which the message signal is sampled during the simulation.

The frequency of the carrier signal is usually much greater than the highest frequency of the input message signal. The Nyquist sampling theorem requires that the simulation sampling rate, F_s , be greater than two times the sum of the carrier frequency and the highest frequency of the modulated signal in order for the demodulator to recover the message correctly.

Representing Digital Signals

To modulate a signal using digital modulation with an alphabet having M symbols, start with a real message signal whose values are integers from 0 to $M-1$. Represent the signal by listing its values in a vector, x . Alternatively, you can use a matrix to represent a multichannel signal, where each column of the matrix represents one channel.

For example, if the modulation uses an alphabet with eight symbols, then the vector $[2 \ 3 \ 7 \ 1 \ 0 \ 5 \ 5 \ 2 \ 6]'$ is a valid single-channel input to the modulator. As a multichannel example, the two-column matrix

```
[2 3;
 3 3;
 7 3;
 0 3;]
```

defines a two-channel signal in which the second channel has a constant value of 3.

Signals and Delays

All digital modulation blocks process only discrete-time signals and use the baseband representation. The data types of inputs and outputs are depicted in the following figure.



Note If you want to separate the in-phase and quadrature components of the complex modulated signal, use the Complex to Real-Imag block in the Simulink Math Operations library.

Integer-Valued Signals and Binary-Valued Signals

Some digital modulation blocks can accept either integer-valued or binary-valued signals. The corresponding demodulation blocks can output either integers or groups of individual bits that represent integers. This section describes how modulation blocks process integer or binary inputs; the case for demodulation blocks is the reverse. You should note that modulation blocks have an **Input type** parameter and that demodulation blocks have an **Output type** parameter.

When you set the **Input type** parameter to Integer, the block accepts integer values from 0 to $M-1$. M represents the **M-ary number** block parameter.

When you set the **Input type** parameter to Bit, the block accepts binary-valued inputs that represent integers. The block collects binary-valued signals into groups of $K = \log_2(M)$ bits

where

K represents the number of bits per symbol. Since $M = 2^K$, K is commonly referred to as the modulation order.

The input vector length must be an integer multiple of K . In this configuration, the block accepts a group of K bits and maps that group onto a symbol at the block output. The block outputs one modulated symbol for each group of K bits.

Constellation Ordering (or Symbol Set Ordering)

Depending on the modulation scheme, the **Constellation ordering** or **Symbol set ordering** parameter indicates how the block maps a group of K input bits to a corresponding symbol. When you set the parameter to Binary, the block maps $[u(1) u(2) \dots u(K)]$ to the integer

$$\sum_{i=1}^K u(i)2^{K-i}$$

and assumes that this integer is the input value. $u(1)$ is the most significant bit.

If you set $M = 8$, **Constellation ordering** (or **Symbol set ordering**) to Binary, and the binary input word is $[1 1 0]$, the block converts $[1 1 0]$ to the integer 6. The block produces the same output when the input is 6 and the **Input type** parameter is Integer.

When you set **Constellation ordering** (or **Symbol set ordering** or **Symbol mapping**) to Gray, the block uses a Gray-coded arrangement and assigns binary inputs to points of a predefined Gray-coded signal constellation. The predefined M-ary Gray-coded signal constellation assigns the binary representation

```
M = 8; P = [0:M-1]';
de2bi(bitxor(P, floor(P/2)), log2(M), 'left-msb')
```

to the P^{th} integer.

The following tables show the typical Binary to Gray mapping for $M = 8$.

Binary to Gray Mapping for Bits

Binary Code	Gray Code
000	000
001	001
010	011
011	010
100	110
101	111
110	101
111	100

Gray to Binary Mapping for Integers

Binary Code	Gray Code
0	0
1	1
2	3
3	2
4	6
5	7
6	5
7	4

Gray Encoding a Modulated Signal

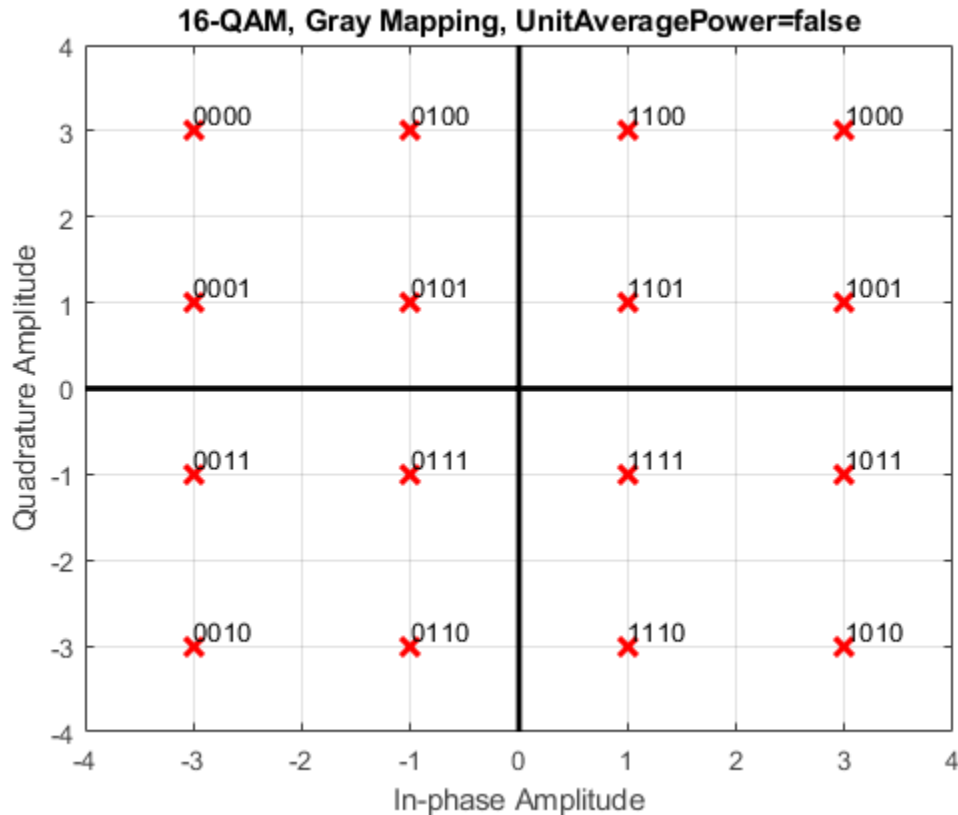
For the PSK, DPSK, FSK, QAM, and PAM modulation types, Gray constellations are obtained by setting the symbol mapping to Gray-encoding in the corresponding modulation function or System object@.

For modulation functions, you can specify 'gray' for the symbol order input argument to obtain Gray-encoded modulation.

The following example uses the `qammod` function with Gray-encoded symbol mapping.

```
y = [0:15];
y = de2bi(y);
```

```
M = 16;  
symorder = 'gray';  
xmap = qammod(y,M,symorder,'InputType','bit','PlotConstellation',true);
```



Checking the constellation plot, you can see the modulated symbols are Gray-encoded because all adjacent elements differ by only one bit.

Upsample Signals and Rate Changes

Some digital modulation blocks can output an upsampled version of the modulated signal, while their corresponding digital demodulation blocks can accept an upsampled version of the modulated signal as input. In both cases, the **Rate options** parameter represents the upsampling factor, which must be a positive integer. Depending on whether the input

signal is single-rate mode or multirate mode, the block either changes the signal's vector size or its sample time, as the following table indicates. Only the QPSK blocks deviate from the information in the table, in that S is replaced by $2S$ in the scaling factors.

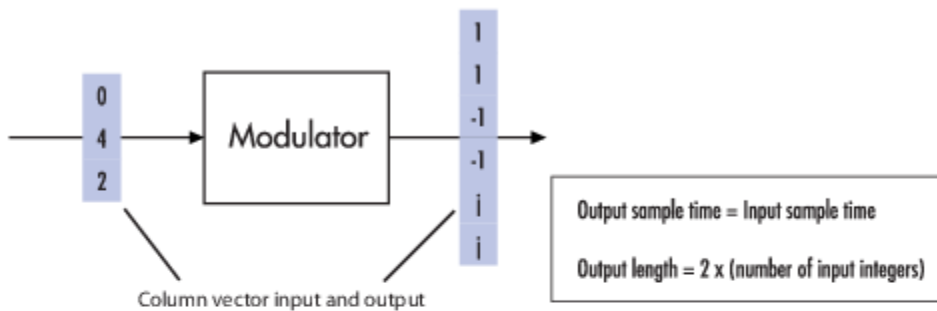
Process Upsampled Modulated Data (Except QPSK Method)

Computation Type	Input Status	Result
Modulation	Single-rate processing	Output vector length is S times the number of integers or binary words in the input vector. Output sample time equals the input sample time.
Modulation	Multirate processing	Output vector is a scalar. Output sample time is $1/S$ times the input sample time.
Demodulation	Single-rate processing	Number of integers or binary words in the output vector is $1/S$ times the number of samples in the input vector. Output sample time equals the input sample time.
Demodulation	Multirate processing	Output signal contains one integer or one binary word. Output sample time is S times the input sample time. Furthermore, if $S > 1$ and the demodulator is from the AM, PM, or FM sublibrary, the demodulated signal is delayed by one output sample period. There is no delay if $S = 1$ or if the demodulator is from the CPM sublibrary.

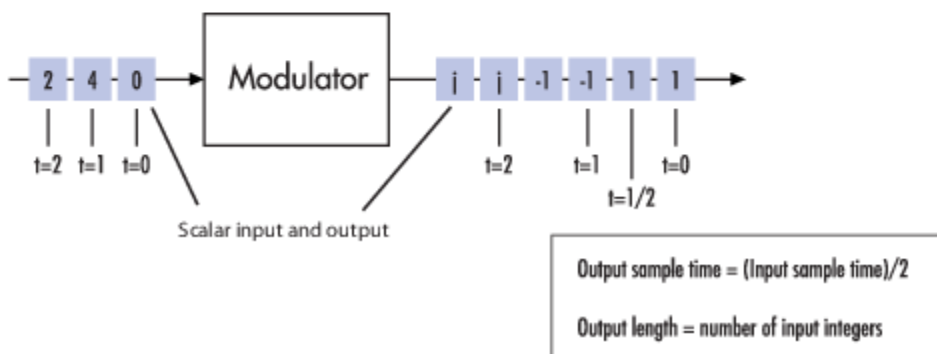
Illustrations of Size or Rate Changes

The following schematics illustrate how a modulator (other than OQPSK) upsamples a triplet of frame-based and sample-based integers. In both cases, the **Samples per symbol** parameter is 2.

Upsample Output: Single-Rate Processing



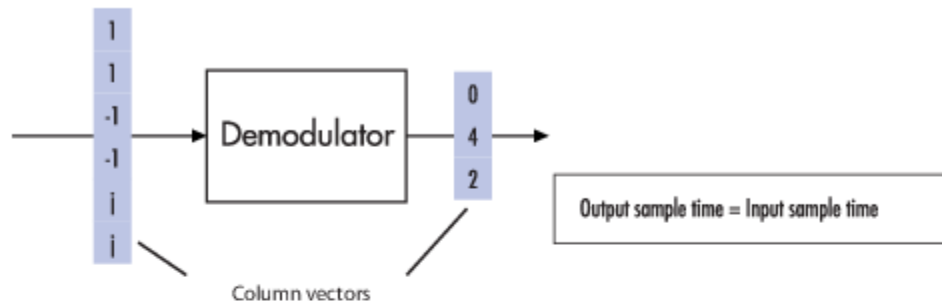
Upsample Output: Multirate Processing



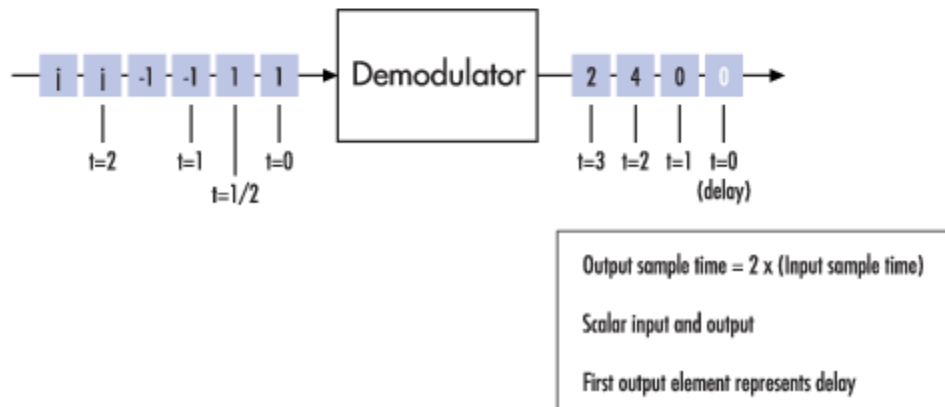
The following schematics illustrate how a demodulator (other than OQPSK or one from the CPM sublibrary) processes three doubly sampled symbols using both frame-based and

sample-based inputs. In both cases, the **Samples per symbol** parameter is 2. The sample-based schematic includes an output delay of one sample period.

Upsampled Input: Single Rate Processing



Upsampled Input: Multirate Processing

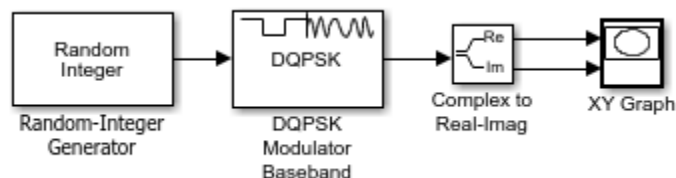


For more information on delays, see Delays in Digital Modulation on page 8-193.

PM Modulation

DQPSK Signal Constellation Points and Transitions

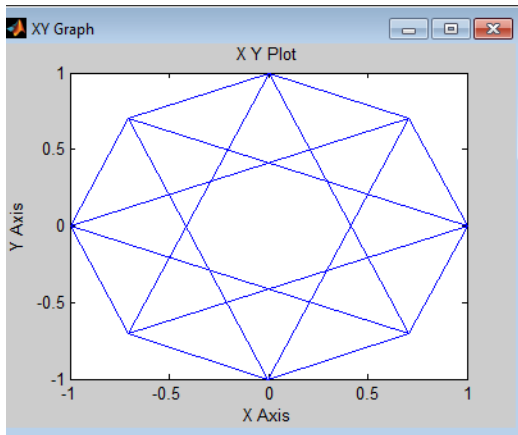
The model below plots the output of the DQPSK Modulator Baseband block. The image shows the possible transitions from each symbol in the DQPSK signal constellation to the next symbol.



To open this model enter `doc_dqpsk_plot` at the MATLAB command line. To build the model, gather and configure these blocks:

- Random Integer Generator, in the Random Data Sources sublibrary of the Comm Sources library
 - Set **M-ary number** to 4.
 - Set **Initial seed** to any positive integer scalar, preferably the output of the `randn` function.
 - Set **Sample time** to `.01`.
- DQPSK Modulator Baseband, in the PM sublibrary of the Digital Baseband sublibrary of Modulation
- Complex to Real-Imag, in the Simulink Math Operations library
- XY Graph, in the Simulink Sinks library

Use the blocks' default parameters unless otherwise instructed. Connect the blocks as in the figure above. Running the model produces the following plot. The plot reflects the transitions among the eight DQPSK constellation points.

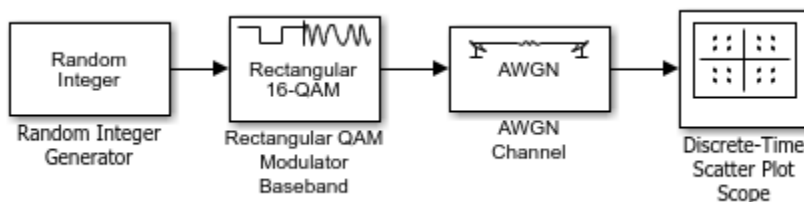


This plot illustrates $\pi/4$ -DQPSK modulation, because the default **Phase offset** parameter in the DQPSK Modulator Baseband block is $\pi/4$. To see how the phase offset influences the signal constellation, change the **Phase offset** parameter in the DQPSK Modulator Baseband block to $\pi/8$ or another value. Run the model again and observe how the plot changes.

AM Modulation

Rectangular QAM Modulation and Scatter Diagram

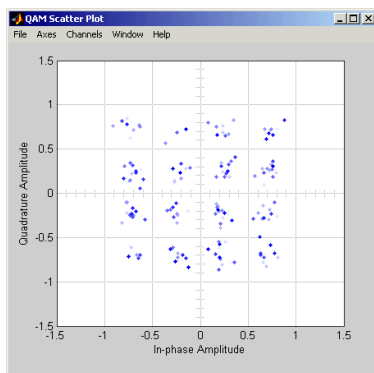
The model below uses the M-QAM Modulator Baseband block to modulate random data. After passing the symbols through a noisy channel, the model produces a scatter diagram of the noisy data. The diagram suggests what the underlying signal constellation looks like and shows that the noise distorts the modulated signal from the constellation.



To open this model, enter `doc_qam_scatter` at the MATLAB command line. To build the model, gather and configure these blocks:

- Random Integer Generator, in the Random Data Sources sublibrary of the Comm Sources library
 - Set **M-ary number** to 16.
 - Set **Initial seed** to any positive integer scalar, preferably the output of the randn function.
 - Set **Sample time** to . 1.
- Rectangular QAM Modulator Baseband, in the AM sublibrary of the Digital Baseband sublibrary of Modulation
 - Set **Normalization method** to Peak Power.
- AWGN Channel, in the Channels library
 - Set **Es/No** to 20.
 - Set **Symbol period** to . 1.
- Constellation Diagram, in the Comm Sinks library
 - Set **Symbols to display** to 160.

Connect the blocks as in the figure. From the model window's **Simulation** menu, select **Model Configuration parameters**. In the Configuration Parameters dialog box, set **Stop time** to 250. Running the model produces a scatter diagram like the following one. Your plot might look somewhat different, depending on your **Initial seed** value in the Random Integer Generator block. Because the modulation technique is 16-QAM, the plot shows 16 clusters of points. If there were no noise, the plot would show the 16 exact constellation points instead of clusters around the constellation points.



Compute Symbol Error Rate

The example generates a random digital signal, modulates it, adds noise, demodulates the noisy signal, and computes the symbol error rate. The noisy, modulated data is plotted in a constellation diagram. Numerical results and plot may vary due to the random input data.

Create a random digital message and a constellation diagram System object.

```
M = 16; % Alphabet size, 16-QAM
x = randi([0 M-1],5000,1);

cpts = qammod(0:M-1,M);
constDiag = comm.ConstellationDiagram('ReferenceConstellation',cpts, ...
    'XLimits',[-4 4], 'YLimits',[-4 4]);
```

Apply 16-QAM modulation and transmit signal through an AWGN channel.

```
y = qammod(x,M);
ynoisy = awgn(y,15, 'measured');
```

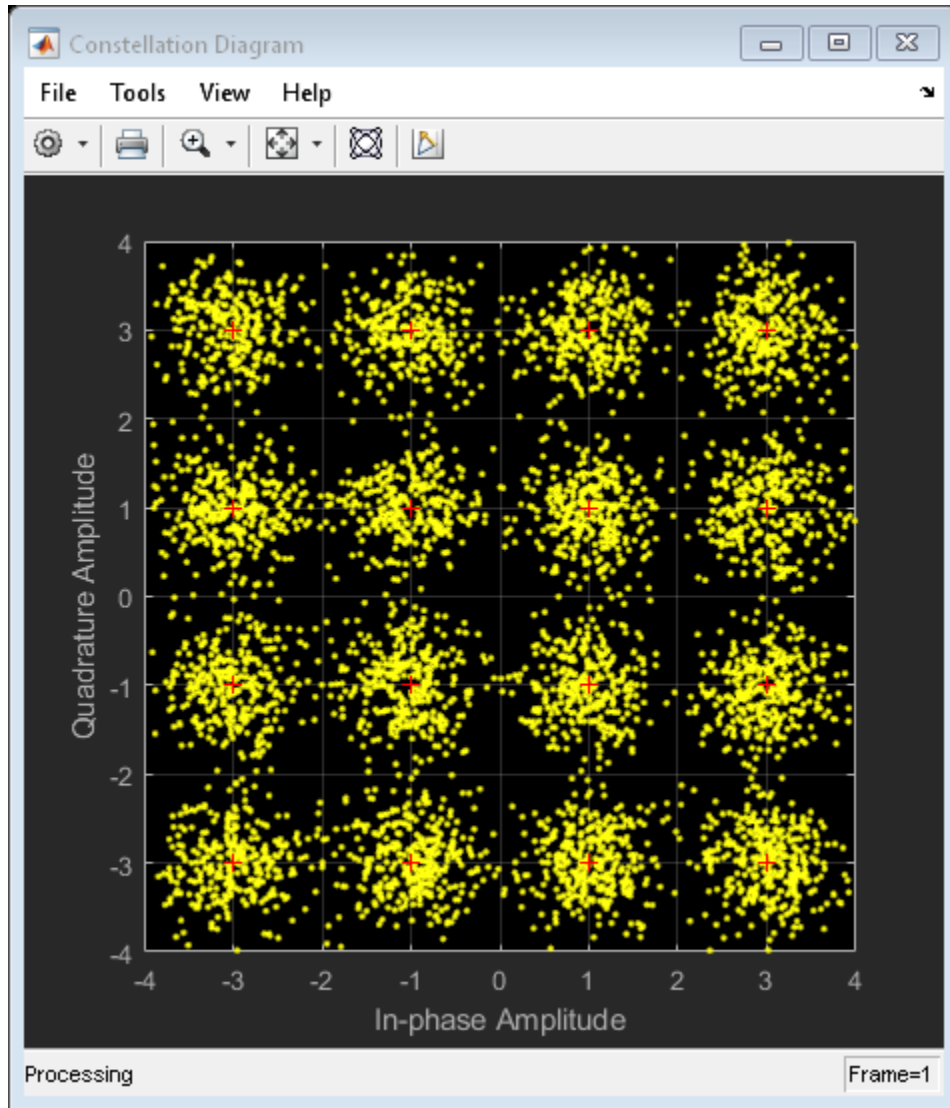
Demodulate ynoisy to recover the message and check the symbol error rate.

```
z = qamdemod(ynoisyy,M);
[num,rt] = symerr(x,z)

num = 79
rt = 0.0158
```

Create constellation diagram from noisy data. The signal reference constellation has 16 precisely located points but the transmitted symbols with the noise added causes the scatter plot to have a small cluster of points scattered around each reference constellation point.

```
constDiag(ynoisyy)
```



Combine Pulse Shaping and Filtering with Modulation

Modulation is often followed by pulse shaping, and demodulation is often preceded by a filtering or an integrate-and-dump operation. This section presents an example involving

rectangular pulse shaping. For an example that uses raised cosine pulse shaping, see “Pulse Shaping Using a Raised Cosine Filter” on page 16-7.

Rectangular Pulse Shaping

Rectangular pulse shaping repeats each output from the modulator a fixed number of times to create an upsampled signal. Although it is less realistic than other kinds of pulse shaping, rectangular pulse shaping can be a first step or an exploratory step in algorithm development. If the transmitter upsamples the modulated signal, then the receiver should downsample the received signal before demodulating. The code below uses the `rectpulse` function for rectangular pulse shaping at the transmitter and the `intdump` function for downsampling at the receiver. The “integrate and dump” operation is one way to downsample the received signal.

```
% Create a random digital message and a constellation diagram System
% object.
M = 16; % Alphabet size, 16-QAM
x = randi([0 M-1],5000,1); % Message signal
Nsamp = 4; % Oversampling rate

% Apply 16-QAM modulation and rectangular pulse shaping. Transmit signal
% through an AWGN channel.
y = qammod(x,M);
ypulse = rectpulse(y,Nsamp);
ynoisyy = awgn(ypulse,15,'measured');

% Downsample at the receiver.
ydownsamp = intdump(ynoisyy,Nsamp);

% Demodulate to recover the message.
z = qamdmod(ydownsamp,M);
```

CPM Modulation

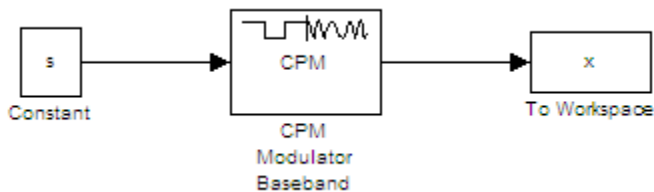
Phase Tree for Continuous Phase Modulation

This example plots a phase tree associated with a continuous phase modulation scheme. A phase tree is a diagram that superimposes many curves, each of which plots the phase of a modulated signal over time. The distinct curves result from different inputs to the modulator.

This example uses the CPM Modulator Baseband block for its numerical computations. The block is configured using a raised cosine filter pulse shape. The example also

illustrates how you can use Simulink and MATLAB together. The example uses MATLAB commands to run a series of simulations with different input signals, to collect the simulation results, and to plot the full data set.

Note In contrast to this example's approach using both MATLAB and Simulink, the `commcpmphasetree` example produces a phase tree using a Simulink model without additional lines of MATLAB code.



Open the model by typing `doc_phasetree` at the MATLAB command line. To build the model, gather and configure these blocks:

- Constant, in the Simulink Commonly Used Blocks library
 - Set **Constant value** to `s` (which will appear in the MATLAB workspace).
 - Set **Sampling mode** to Frame-based.
 - Set **Frame period** to 1.
- CPM Modulator Baseband
 - Set **M-ary number** to 2.
 - Set **Modulation index** to $2/3$.
 - Set **Frequency pulse shape** to Raised Cosine.
 - Set **Pulse length** to 2.
- To Workspace, in the Simulink Sinks library
 - Set **Variable name** to `x`.
 - Set **Save format** to Array.

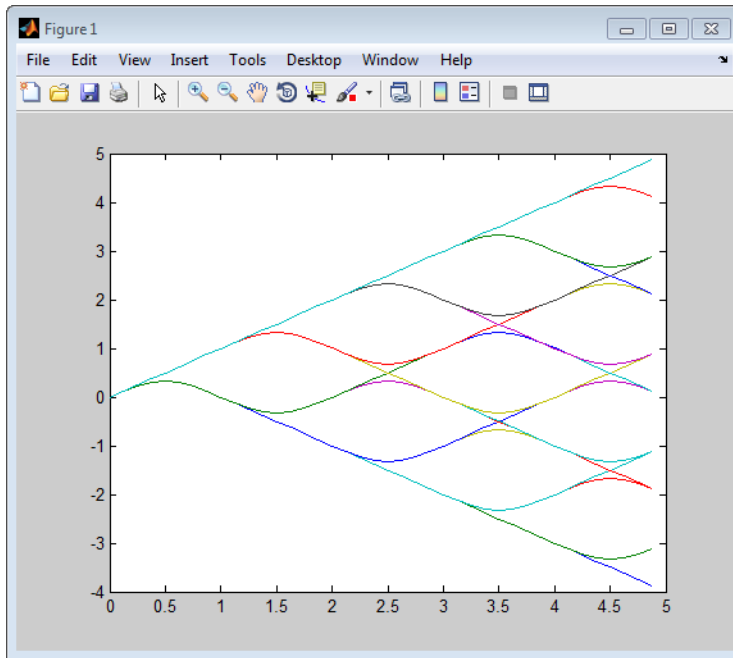
Do not run the model, because the variable `s` is not yet defined in the MATLAB workspace. Instead, save the model to a folder on your MATLAB path, using the filename `doc_phasetree`.

The second step of this example is to execute the following MATLAB code:

```
% Parameters from the CPM Modulator Baseband block
M_ary_number = 2;
modulation_index = 2/3;
pulse_length = 2;
samples_per_symbol = 8;

L = 5; % Symbols to display
pmat = [];
for ip_sig = 0:(M_ary_number^L)-1
    s = de2bi(ip_sig,L,M_ary_number,'left-msb');
    % Apply the mapping of the input symbol to the CPM
    % symbol 0 -> -(M-1), 1 -> -(M-2), etc.
    s = 2*s'+1-M_ary_number;
    sim('doc_phasetree', .9); % Run model to generate x.
    % Next column of pmat
    pmat(:,ip_sig+1) = unwrap(angle(x(:)));
end;
pmat = pmat/(pi*modulation_index);
t = (0:L*samples_per_symbol-1)'/samples_per_symbol;
plot(t,pmat); figure(gcf); % Plot phase tree.
```

This code defines the parameters for the CPM Modulator, applies symbol mapping, and plots the results. Each curve represents a different instance of simulating the CPM Modulator Baseband block with a distinct (constant) input signal.



Exact LLR Algorithm

The log-likelihood ratio (LLR) is the logarithm of the ratio of probabilities of a 0 bit being transmitted versus a 1 bit being transmitted for a received signal. The LLR for a bit b is defined as:

$$L(b) = \log\left(\frac{\Pr(b = 0 | r = (x, y))}{\Pr(b = 1 | r = (x, y))}\right)$$

Assuming equal probability for all symbols, the LLR for an AWGN channel can be expressed as:

$$L(b) = \log\left(\frac{\sum_{s \in S_0} e^{-\frac{1}{\sigma^2}((x - s_x)^2 + (y - s_y)^2)}}{\sum_{s \in S_1} e^{-\frac{1}{\sigma^2}((x - s_x)^2 + (y - s_y)^2)}}\right)$$

where the variables represent the values shown in the following table.

Variable	What the Variable Represents
r	Received signal with coordinates (x, y) .
b	Transmitted bit (one of the K bits in an M -ary symbol, assuming all M symbols are equally probable).
S_0	Ideal symbols or constellation points with bit 0, at the given bit position.
S_1	Ideal symbols or constellation points with bit 1, at the given bit position.
s_x	In-phase coordinate of ideal symbol or constellation point.
s_y	Quadrature coordinate of ideal symbol or constellation point.
σ^2	Noise variance of baseband signal.
σ_x^2	Noise variance along in-phase axis.
σ_y^2	Noise variance along quadrature axis.

Note Noise components along the in-phase and quadrature axes are assumed to be independent and of equal power (i.e., $\sigma_x^2 = \sigma_y^2 = \sigma^2/2$).

Approximate LLR Algorithm

Approximate LLR is computed by taking into consideration only the nearest constellation point to the received signal with a 0 (or 1) at that bit position, rather than all the constellation points as done in exact LLR. It is defined as [8]:

$$L(b) = -\frac{1}{\sigma^2} \left(\min_{s \in S_0} \left((x - s_x)^2 + (y - s_y)^2 \right) - \min_{s \in S_1} \left((x - s_x)^2 + (y - s_y)^2 \right) \right)$$

Delays in Digital Modulation

Digital modulation and demodulation blocks sometimes incur delays between their inputs and outputs, depending on their configuration and on properties of their signals. The following table lists sources of delay and the situations in which they occur.

Delays Resulting from Digital Modulation or Demodulation

Modulation or Demodulation Type	Situation in Which Delay Occurs	Amount of Delay
FM demodulator	Sample-based processing	One output period
All demodulators in CPM sublibrary	Multirate processing, and the model uses a variable-step solver or a fixed-step solver with the Tasking Mode parameter set to SingleTasking D = Traceback length parameter	D+1 output periods
	Single-rate processing, D = Traceback depth parameter	D output periods
OQPSK demodulator	Single-rate processing	For more information, see OQPSK Demodulator Baseband.
	Multirate processing, and the model uses a fixed-step solver with Tasking Mode parameter set to Auto or MultiTasking .	
	Multirate processing processing, and the model uses a variable-step solver or the Tasking Mode parameter is set to SingleTasking .	
All decoders in TCM sublibrary	Operation mode set to Continuous , Tr = Traceback depth parameter, and code rate k/n	Tr*k output bits

As a result of delays, data that enters a modulation or demodulation block at time T appears in the output at time T+delay. In particular, if your simulation computes error statistics or compares transmitted with received data, it must take the delay into account when performing such computations or comparisons.

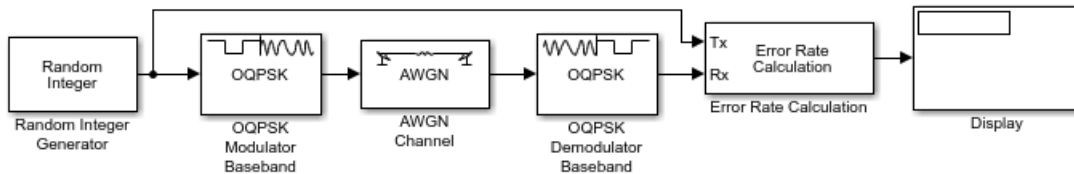
First Output Sample in DPSK Demodulation

In addition to the delays mentioned above, the M-DPSK, DQPSK, and DBPSK demodulators produce output whose first sample is unrelated to the input. This is related to the differential modulation technique, not the particular implementation of it.

Example: Delays from Demodulation

Demodulation in the model below causes the demodulated signal to lag, compared to the unmodulated signal. When computing error statistics, the model accounts for the delay by

setting the Error Rate Calculation block's **Receive delay** parameter to 0. If the **Receive delay** parameter had a different value, then the error rate showing at the top of the Display block would be close to 1/2.



To open this model , enter `doc_oqpsk_modulation_delay` at the MATLAB command line. To build the model, gather and configure these blocks:

- Random Integer Generator, in the Random Data Sources sublibrary of the Comm Sources library
 - Set **M-ary number** to 4.
 - Set **Initial seed** to any positive integer scalar.
- OQPSK Modulator Baseband, in the PM sublibrary of the Digital Baseband sublibrary of Modulation
- AWGN Channel, in the Channels library
 - Set **Es/No** to 6.
- OQPSK Demodulator Baseband, in the PM sublibrary of the Digital Baseband sublibrary of Modulation
- Error Rate Calculation, in the Comm Sinks library
 - Set **Receive delay** to 1.
 - Set **Computation delay** to 0.
 - Set **Output data** to Port.
- Display, in the Simulink Sinks library
 - Drag the bottom edge of the icon to make the display big enough for three entries.

Connect the blocks as shown above. From the model window's **Simulation**, select **Model Configuration parameters**. In the **Configuration Parameters** dialog box, set **Stop time** to 1000. Then run the model and observe the error rate at the top of the Display block's icon. Your error rate will vary depending on your **Initial seed** value in the Random Integer Generator block.

Selected Bibliography for Digital Modulation

- [1] Jeruchim, M. C., P. Balaban, and K. S. Shanmugan, *Simulation of Communication Systems*, New York, Plenum Press, 1992.
- [2] Proakis, J. G., *Digital Communications*, 3rd ed., New York, McGraw-Hill, 1995.
- [3] Sklar, B., *Digital Communications: Fundamentals and Applications*, Englewood Cliffs, NJ, Prentice-Hall, 1988.
- [4] Anderson, J. B., T. Aulin, and C.-E. Sundberg, *Digital Phase Modulation*, New York, Plenum Press, 1986.
- [5] Biglieri, E., D. Divsalar, P.J. McLane, and M.K. Simon, *Introduction to Trellis-Coded Modulation with Applications*, New York, Macmillan, 1991.
- [6] Pawula, R.F., "On M-ary DPSK Transmission Over Terrestrial and Satellite Channels," *IEEE Transactions on Communications*, Vol. COM-32, July 1984, pp. 752-761.
- [7] Smith, J. G., "Odd-Bit Quadrature Amplitude-Shift Keying," *IEEE Transactions on Communications*, Vol. COM-23, March 1975, pp. 385-389.
- [8] Viterbi, A. J., "An Intuitive Justification and a Simplified Implementation of the MAP Decoder for Convolutional Codes," *IEEE Journal on Selected Areas in Communications*, vol. 16, No. 2, pp 260-264, Feb. 1998

Analog Passband Modulation

In this section...

“Analog Modulation Features” on page 8-197

“Represent Signals for Analog Modulation” on page 8-198

“Sampling Issues in Analog Modulation” on page 8-201

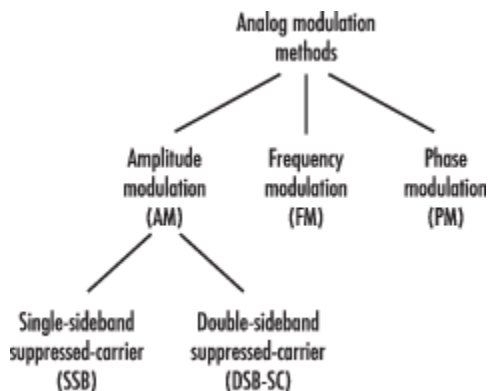
“Filter Design Issues” on page 8-201

Analog Modulation Features

In most communication medium, only a fixed range of frequencies is available for transmission. One way to communicate a message signal whose frequency spectrum does not fall within that fixed frequency range, or one that is otherwise unsuitable for the channel, is to alter a transmittable signal according to the information in your message signal. This alteration is called *modulation*, and it is the modulated signal that you transmit. The receiver then recovers the original signal through a process called *demodulation*. This section describes how to modulate and demodulate analog signals using blocks.

Open the Modulation library by double-clicking its icon in the main Communications Toolbox block library. Then, open the Analog Passband sublibrary by double-clicking its icon in the Modulation library.

The following figure shows the modulation techniques that Communications Toolbox supports for analog signals. As the figure suggests, some categories of techniques include named special cases.



For a given modulation technique, two ways to simulate modulation techniques are called *baseband* and *passband*. This product supports passband simulation for analog modulation.

The modulation and demodulation blocks also let you control such features as the initial phase of the modulated signal and post-demodulation filtering.

Represent Signals for Analog Modulation

Analog modulation blocks in this product process only sample-based scalar signals. The input and output of the analog modulator and demodulator are all real signals.

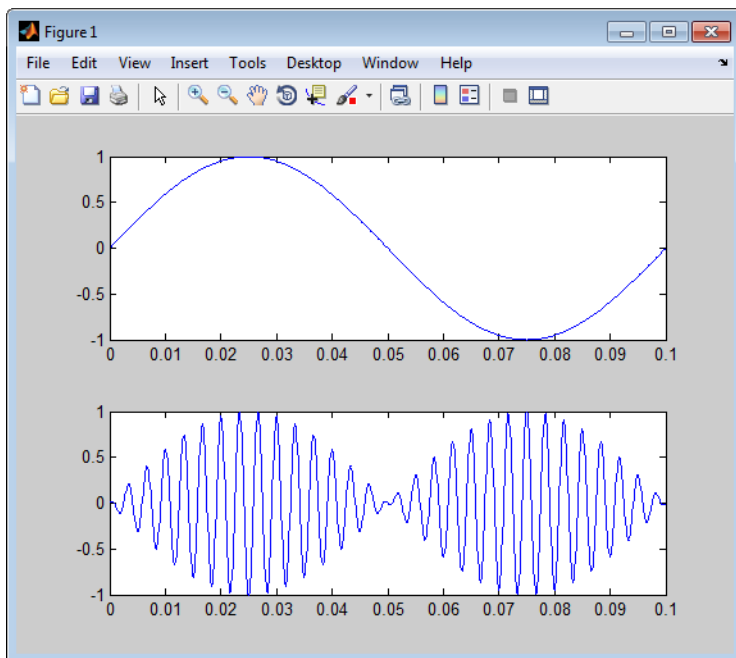
All analog demodulators in this product produce discrete-time, not continuous-time, output.

Representing Analog Signals Using MATLAB

To modulate an analog signal using MATLAB, start with a real message signal and a sampling rate F_s in hertz. Represent the signal using a vector x , the entries of which give the signal's values in time increments of $1/F_s$. Alternatively, you can use a matrix to represent a multichannel signal, where each column of the matrix represents one channel.

For example, if t measures time in seconds, then the vector x below is the result of sampling a sine wave 8000 times per second for 0.1 seconds. The vector y represents the modulated signal.

```
Fs = 8000; % Sampling rate is 8000 samples per second.  
Fc = 300; % Carrier frequency in Hz  
t = [0:.1*Fs]'/Fs; % Sampling times for .1 second  
x = sin(20*pi*t); % Representation of the signal  
y = ammod(x,Fc,Fs); % Modulate x to produce y.  
figure;  
subplot(2,1,1); plot(t,x); % Plot x on top.  
subplot(2,1,2); plot(t,y)% Plot y below.
```



As a multichannel example, the code below defines a two-channel signal in which one channel is a sinusoid with zero initial phase and the second channel is a sinusoid with an initial phase of $\pi/8$.

```
Fs = 8000;
t = [0:.1*Fs]'/Fs;
x = [sin(20*pi*t), sin(20*pi*t+pi/8)];
```

Analog Modulation with Additive White Gaussian Noise (AWGN) Using MATLAB

This example illustrates the basic format of the analog modulation and demodulation functions. Although the example uses phase modulation, most elements of this example apply to other analog modulation techniques as well.

The example samples an analog signal and modulates it. Then it simulates an additive white Gaussian noise (AWGN) channel, demodulates the received signal, and plots the original and demodulated signals.

```
% Prepare to sample a signal for two seconds,
% at a rate of 100 samples per second.
```

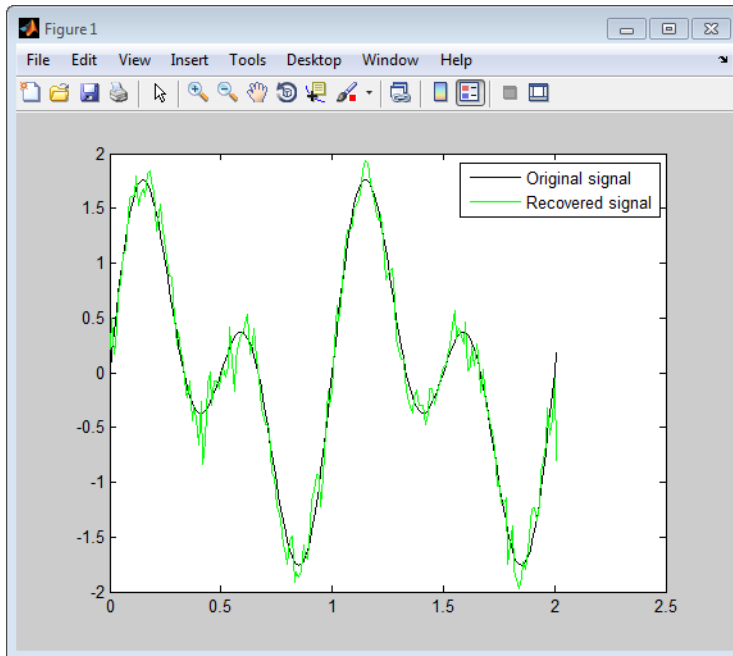
```
Fs = 100; % Sampling rate
t = [0:2*Fs+1]/Fs; % Time points for sampling

% Create the signal, a sum of sinusoids.
x = sin(2*pi*t) + sin(4*pi*t);

Fc = 10; % Carrier frequency in modulation
phasedev = pi/2; % Phase deviation for phase modulation

y = pmod(x,Fc,Fs,phasedev); % Modulate.
y = awgn(y,10,'measured',103); % Add noise.
z = pmdemod(y,Fc,Fs,phasedev); % Demodulate.

% Plot the original and recovered signals.
figure; plot(t,x,'k-',t,z,'g-');
legend('Original signal','Recovered signal');
```



Other examples using analog modulation functions appear in the reference pages for `ammod`, `amdemod`, `ssbmod`, and `fmmod`.

Sampling Issues in Analog Modulation

The proper simulation of analog modulation requires that the Nyquist criterion be satisfied, taking into account the signal bandwidth.

Specifically, the sample rate of the system must be greater than twice the sum of the carrier frequency and the signal bandwidth.

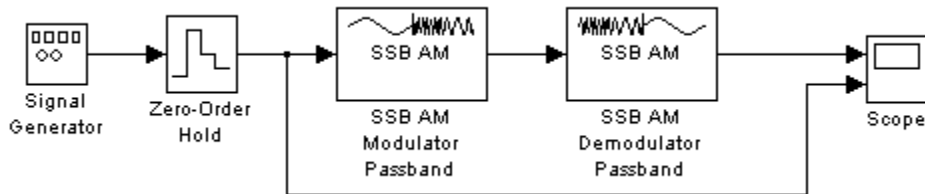
Filter Design Issues

After demodulating, you might want to filter out the carrier signal. The particular filter used, such as `butter`, `cheby1`, `cheby2`, and `ellip`, can be selected on the mask of the demodulator block. Different filtering methods have different properties, and you might need to test your application with several filters before deciding which is most suitable.

Varying Filter's Cutoff Frequency Using Simulink

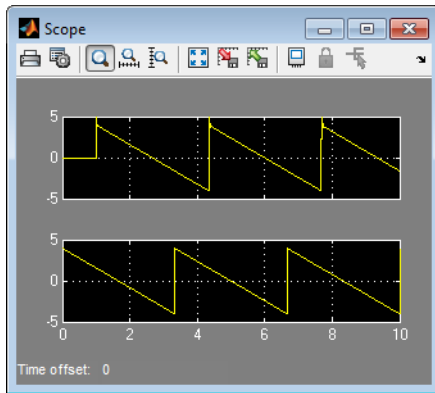
In many situations, a suitable cutoff frequency is half the carrier frequency. Because the carrier frequency must be higher than the bandwidth of the message signal, a cutoff frequency chosen in this way properly filters out unwanted frequency components. If the cutoff frequency is too high, the unwanted components may not be filtered out. If the cutoff frequency is too low, it might narrow the bandwidth of the message signal.

The following example modulates a sawtooth message signal, demodulates the resulting signal using a Butterworth filter, and plots the original and recovered signals. The Butterworth filter is implemented within the SSB AM Demodulator Passband block.



To open this model, enter `doc_filtercutoffs` at the MATLAB command line.

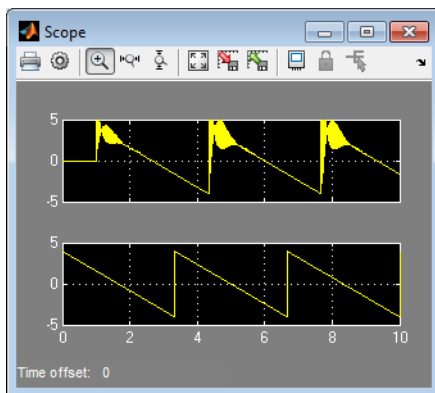
This example generates the following output:



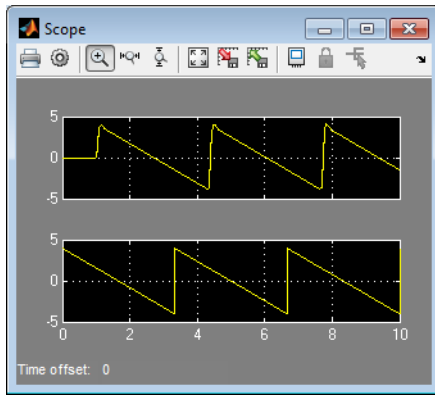
There is invariably a delay between a demodulated signal and the original received signal. Both the filter order and the filter parameters directly affect the length of this delay.

Other Filter Cutoffs

To see the effect of a lowpass filter with a *higher* cutoff frequency, set the **Cutoff frequency** of the SSB AM Demodulator Passband block to 49, and run the simulation again. The new result is shown below. The higher cutoff frequency allows the carrier signal to interfere with the demodulated signal.



To see the effect of a lowpass filter with a *lower* cutoff frequency, set the **Cutoff frequency** of the SSB AM Demodulator Passband block to 4, and run the simulation again. The new result is shown in the following figure. The lower cutoff frequency narrows the bandwidth of the demodulated signal.



Phase-Locked Loops

Phase-Locked Loop Features

The Components sublibrary contains voltage-controlled oscillator (VCO) models as well as phase-locked loop (PLL) models.

For details about phase-locked loops, see the works listed in “Selected Bibliography for Synchronization” on page 8-206.

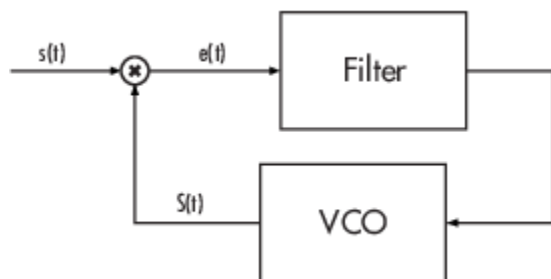
Voltage-Controlled Oscillator Blocks

A voltage-controlled oscillator is one part of a phase-locked loop. The Continuous-Time VCO and Discrete-Time VCO blocks implement voltage-controlled oscillators. These blocks produce continuous-time and discrete-time output signals, respectively. Each block's output signal is sinusoidal, and changes its frequency in response to the amplitude variations of the input signal.

Overview of PLL Simulation

A phase-locked loop (PLL), when used in conjunction with other components, helps synchronize the receiver. A PLL is an automatic control system that adjusts the phase of a local signal to match the phase of the received signal. The PLL design works best for narrowband signals.

A simple PLL consists of a phase detector, a loop filter, and a voltage-controlled oscillator (VCO). For example, the following figure shows how these components are arranged for an analog passband PLL. In this case, the phase detector is just a multiplier. The signal $e(t)$ is often called the error signal.



The following table indicates the supported types of PLLs and the blocks that implement them.

Supported PLLs in Components Library

Type of PLL	Block
Analog passband PLL	Phase-Locked Loop
Analog baseband PLL	Baseband PLL
Linearized analog baseband PLL	Linearized Baseband PLL
Digital PLL using a charge pump	Charge Pump PLL

Different PLLs use different phase detectors, filters, and VCO characteristics. Some of these attributes are built into the PLL blocks in this product, while others depend on parameters that you set in the block mask:

- You specify the filter's transfer function in the block mask using the **Lowpass filter numerator** and **Lowpass filter denominator** parameters. Each of these parameters is a vector that lists the coefficients of the respective polynomial in order of descending exponents of the variable s . To design a filter, you can use functions such as `butter`, `cheby1`, and `cheby2` in Signal Processing Toolbox.
- You specify the key VCO characteristics in the block mask. All four PLL blocks use a **VCO input sensitivity** parameter. Some blocks also use **VCO quiescent frequency**, **VCO initial phase**, and **VCO output amplitude** parameters.
- The phase detector for each of the PLL blocks is a feature that you cannot change from the block mask.

Implementing an Analog Baseband PLL

Unlike passband models for a phase-locked loop, a baseband model does not depend on a carrier frequency. This allows you to use a lower sampling rate in the simulation. Two blocks implement analog baseband PLLs:

- Baseband PLL
- Linearized Baseband PLL

The linearized model and the nonlinearized model differ in that the linearized model uses the approximation

$$\sin(\Delta\theta(t)) \cong \Delta\theta(t)$$

to simplify the computations. This approximation is close when $\Delta\theta(t)$ is near zero. Thus, instead of using the input signal and the VCO output signal directly, the linearized PLL model uses only their *phases*.

Implementing a Digital PLL

The charge pump PLL is a classical digital PLL. Unlike the analog PLLs mentioned above, the charge pump PLL uses a sequential logic phase detector, which is also known as a digital phase detector or a phase/frequency detector.

Selected Bibliography for Synchronization

- [1] Gardner, F.M., "Charge-pump Phase-lock Loops," *IEEE Trans. on Communications*, Vol. 28, November 1980, pp. 1849-1858.
- [2] Gardner, F.M., "Phase Accuracy of Charge Pump PLLs," *IEEE Trans. on Communications*, Vol. 30, October 1982, pp. 2362-2363.
- [3] Gupta, S.C., "Phase Locked Loops," *Proceedings of the IEEE*, Vol. 63, February 1975, pp. 291-306.
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Multiple-Input Multiple-Output (MIMO)

In this section...

“Orthogonal Space-Time Block Codes (OSTBC)” on page 8-207

“MIMO Fading Channel” on page 8-208

“Spherical Decoding” on page 8-208

“Selected Bibliography for MIMO systems” on page 8-208

The use of Multiple-Input Multiple-Output (MIMO) techniques has revolutionized wireless communications systems with potential gains in capacity when using multiple antennas at both transmitter and receiver ends of a communications system. New techniques, which account for the extra spatial dimension, have been adopted to realize these gains in new and previously existing systems.

MIMO technology has been adopted in multiple wireless systems, including Wi-Fi, WiMAX, LTE, and LTE-Advanced.

The Communications Toolbox product offers components to model:

- OSTBC (orthogonal space-time block coding technique)
- MIMO Fading Channels
- Spherical Decoding

and demos highlighting the use of these components in applications.

For background material on the subject of MIMO systems, see the works listed in Selected Bibliography for MIMO systems on page 8-208.

Orthogonal Space-Time Block Codes (OSTBC)

The Communications Toolbox product provides components to model Orthogonal Space Time Block Coding (OSTBC) - a MIMO technique which offers full spatial diversity gain with extremely simple single-symbol maximum likelihood decoding [4,6,8].

In Simulink, the OSTBC Encoder and OSTBC Combiner blocks, residing in the MIMO block library, implement the orthogonal space time block coding technique. These two blocks offer a variety of specific codes (with different rates) for up to 4 transmit and 8 receive antenna systems. The encoder block is used at the transmitter to map symbols to

multiple antennas while the combiner block is used at the receiver to extract the soft information per symbol using the received signal and the channel state information. You access the MIMO library by double clicking the icon in the main Communications Toolbox block library. Alternatively, you can type `commmimo` at the MATLAB command line.

The OSTBC technique is an attractive scheme because it can achieve the full (maximum) spatial diversity order and have symbol-wise maximum-likelihood (ML) decoding. For more information pertaining to the algorithmic details and the specific codes implemented, see OSTBC Combining Algorithms on the OSTBC Combiner block help page and OSTBC Encoding Algorithms on the OSTBC Encoder block help page. Similar functionality is available in MATLAB by using the `comm.OSTBCCombiner` and `comm.OSTBCEncoder` System objects.

MIMO Fading Channel

The Communications Toolbox software includes a MIMO fading channel object. You can use this object to model the fading channel characteristics of MIMO links. The object models both Rayleigh and Rician fading, and uses the Kronecker model for the spatial correlation between the links [1].

For more information, see the `comm.MIMOChannel`.

Spherical Decoding

The Communications Toolbox software includes a sphere decoder System object. You can use this object to find the maximum-likelihood solution for a set of received symbols over a MIMO channel with any number transmit antennas and receive antennas.

For more information, see the `comm.SphereDecoder` reference page.

Selected Bibliography for MIMO systems

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Huffman Coding

Huffman coding offers a way to compress data. The average length of a Huffman code depends on the statistical frequency with which the source produces each symbol from its alphabet. A Huffman code dictionary, which associates each data symbol with a codeword, has the property that no codeword in the dictionary is a prefix of any other codeword in the dictionary.

The `huffmandict`, `huffmanenco`, and `huffmandeco` functions support Huffman coding and decoding.

Note For long sequences from sources having skewed distributions and small alphabets, arithmetic coding compresses better than Huffman coding. To learn how to use arithmetic coding, see Arithmetic Coding on page 8-12.

Huffman coding requires statistical information about the source of the data being encoded. In particular, the `p` input argument in the `huffmandict` function lists the probability with which the source produces each symbol in its alphabet.

For example, consider a data source that produces 1s with probability 0.1, 2s with probability 0.1, and 3s with probability 0.8. The main computational step in encoding data from this source using a Huffman code is to create a dictionary that associates each data symbol with a codeword. The example below creates such a dictionary and then show the codeword vector associated with a particular value from the data source.

Create a Huffman Code Dictionary

This example shows how to create a Huffman code dictionary using the `huffmandict` function.

Create a vector of data symbols and assign a probability to each.

```
symbols = [1 2 3];  
prob = [0.1 0.1 0.8];
```

Create a Huffman code dictionary. The most probable data symbol, 3, is associated with a one-digit codeword, while less probable data symbols are associated with two-digit codewords.

```
dict = huffmandict(symbols,prob)
```



```
dict=3x2 cell
    {[1]}    {1x2 double}
    {[2]}    {1x2 double}
    {[3]}    {[          0]}
```

Display the second row of the dictionary. The output also shows that a Huffman encoder receiving the data symbol 2 substitutes the sequence 1 0.

```
dict{2,:}
ans = 2
ans = 1x2
    1    0
```

Create and Decode a Huffman Code

The example performs Huffman encoding and decoding using a source whose alphabet has three symbols. Notice that the `huffmanenco` and `huffmandeco` functions use the dictionary created by `huffmandict`.

Generate a data sequence to encode.

```
sig = repmat([3 3 1 3 3 3 3 2 3],1,50);
```

Define the set of data symbols and the probability associated with each element.

```
symbols = [1 2 3];
p = [0.1 0.1 0.8];
```

Create the Huffman code dictionary.

```
dict = huffmandict(symbols,p);
```

Encode and decode the data. Verify that the original data, `sig`, and the decoded data, `dhsig`, are identical.

```
hcode = huffmanenco(sig,dict);
dhsig = huffmandeco(hcode,dict);
isequal(sig,dhsig)
```

`ans = logical
1`

Differential Pulse Code Modulation

In this section...

“Section Overview” on page 8-213

“DPCM Terminology” on page 8-213

“Represent Predictors” on page 8-213

“Example: DPCM Encoding and Decoding” on page 8-214

“Optimize DPCM Parameters” on page 8-215

Section Overview

The quantization in the section Quantizing a Signal on page 8-13 requires no *a priori* knowledge about the transmitted signal. In practice, you can often make educated guesses about the present signal based on past signal transmissions. Using such educated guesses to help quantize a signal is known as *predictive quantization*. The most common predictive quantization method is differential pulse code modulation (DPCM).

The functions `dpcmenco`, `dpcmdeco`, and `dpcmopt` can help you implement a DPCM predictive quantizer with a linear predictor.

DPCM Terminology

To determine an encoder for such a quantizer, you must supply not only a partition and codebook as described in “Represent Partitions” on page 8-2 and “Represent Codebooks” on page 8-3, but also a *predictor*. The predictor is a function that the DPCM encoder uses to produce the educated guess at each step. A linear predictor has the form

$$y(k) = p(1)x(k-1) + p(2)x(k-2) + \dots + p(m-1)x(k-m+1) + p(m)x(k-m)$$

where x is the original signal, $y(k)$ attempts to predict the value of $x(k)$, and p is an m -tuple of real numbers. Instead of quantizing x itself, the DPCM encoder quantizes the *predictive error*, $x-y$. The integer m above is called the *predictive order*. The special case when $m = 1$ is called *delta modulation*.

Represent Predictors

If the guess for the k th value of the signal x , based on earlier values of x , is

$$y(k) = p(1)x(k-1) + p(2)x(k-2) + \dots + p(m-1)x(k-m+1) + p(m)x(k-m)$$

then the corresponding predictor vector for toolbox functions is

$$\text{predictor} = [0, p(1), p(2), p(3), \dots, p(m-1), p(m)]$$

Note The initial zero in the predictor vector makes sense if you view the vector as the polynomial transfer function of a finite impulse response (FIR) filter.

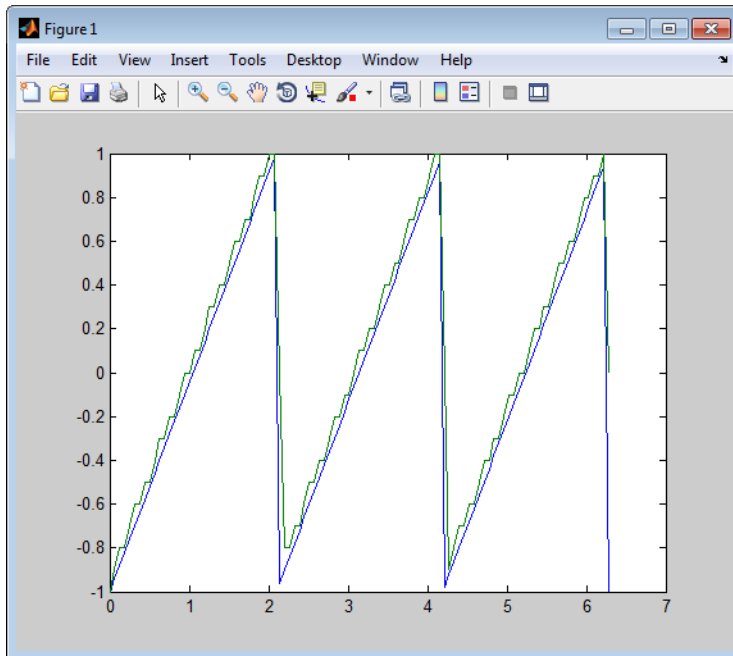
Example: DPCM Encoding and Decoding

A simple special case of DPCM quantizes the difference between the signal's current value and its value at the previous step. Thus the predictor is just $y(k) = x(k-1)$. The code below implements this scheme. It encodes a sawtooth signal, decodes it, and plots both the original and decoded signals. The solid line is the original signal, while the dashed line is the recovered signals. The example also computes the mean square error between the original and decoded signals.

```
predictor = [0 1]; % y(k)=x(k-1)
partition = [-1:.1:.9];
codebook = [-1:.1:1];
t = [0:pi/50:2*pi];
x = sawtooth(3*t); % Original signal
% Quantize x using DPCM.
encodedx = dpcmenco(x,codebook,partition,predictor);
% Try to recover x from the modulated signal.
decodedx = dpcmdeco(encodedx,codebook,predictor);
plot(t,x,t,decodedx,'--')
legend('Original signal','Decoded signal','Location','NorthOutside');
distor = sum((x-decodedx).^2)/length(x) % Mean square error
```

The output is

```
distor =
    0.0327
```



Optimize DPCM Parameters

- “Section Overview” on page 8-215
- “Example: Comparing Optimized and Nonoptimized DPCM Parameters” on page 8-216

Section Overview

The section “Optimize Quantization Parameters” on page 8-4 describes how to use training data with the `lloyd` function to help find quantization parameters that will minimize signal distortion.

This section describes similar procedures for using the `dpcmopt` function in conjunction with the two functions `dpcmenco` and `dpcmdeco`, which first appear in the previous section.

Note The training data you use with `dpcmopt` should be typical of the kinds of signals you will actually be quantizing with `dpcmenco`.

Example: Comparing Optimized and Nonoptimized DPCM Parameters

This example is similar to the one in the last section. However, where the last example created `predictor`, `partition`, and `codebook` in a straightforward but haphazard way, this example uses the same codebook (now called `initcodebook`) as an initial guess for a new *optimized* codebook parameter. This example also uses the predictive order, 1, as the desired order of the new optimized predictor. The `dpcmopt` function creates these optimized parameters, using the sawtooth signal `x` as training data. The example goes on to quantize the training data itself; in theory, the optimized parameters are suitable for quantizing other data that is similar to `x`. Notice that the mean square distortion here is much less than the distortion in the previous example.

```
t = [0:pi/50:2*pi];
x = sawtooth(3*t); % Original signal
initcodebook = [-1:.1:1]; % Initial guess at codebook
% Optimize parameters, using initial codebook and order 1.
[predictor,codebook,partition] = dpcmopt(x,1,initcodebook);
% Quantize x using DPCM.
encodedx = dpcmenco(x,codebook,partition,predictor);
% Try to recover x from the modulated signal.
decodedx = dpcmdeco(encodedx,codebook,predictor);
distor = sum((x-decodedx).^2)/length(x) % Mean square error
```

The output is

```
distor =
    0.0063
```

Compand a Signal

In certain applications, such as speech processing, it is common to use a logarithm computation, called a *compressor*, before quantizing. The inverse operation of a compressor is called an *expander*. The combination of a compressor and expander is called a *compander*.

The compand function supports two kinds of companders: μ -law and A-law companders. Its reference page lists both compressor laws.

Quantize and Compand an Exponential Signal

Quantize an exponential signal with and without companding and compare the mean square distortions.

Set the μ -law parameter μ .

```
Mu = 255;
```

Create an exponential signal and find its maximum value.

```
sig = exp(-4:0.1:4);
V = max(sig);
```

Quantize the signal using equal-length intervals. Set the partition and codebook arguments assuming six bit quantization.

```
partition = 0:2^6-1;
codebook = 0:2^6;
[~,~,distor] = quantiz(sig,partition,codebook);
```

Compress the signal using the compand function. Apply quantization and expand the quantized signal. Calculate the mean square distortion.

```
compsig = compand(sig,Mu,V,'mu/compressor');
[~,quants] = quantiz(compsig,partition,codebook);
newsig = compand(quants,Mu,max(quants),'mu/expander');
distor2 = sum((newsig-sig).^2)/length(sig);
```

Compare the mean square distortions. The output shows that the distortion is smaller when companding is used. This is because equal-length intervals are well suited to the logarithm of sig but not well suited to sig itself.

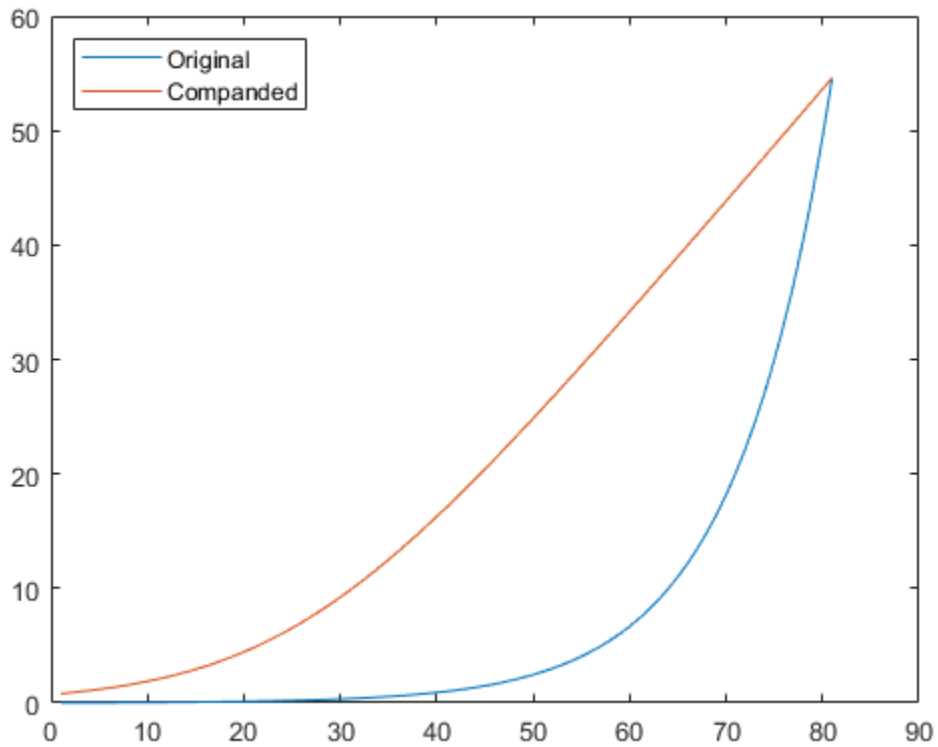
```
[distor, distor2]
```

```
ans = 1×2
```

```
0.5348    0.0397
```

Plot the signal and its companded version.

```
plot([sig' compsig'])  
legend('Original', 'Companded', 'location', 'nw')
```



Arithmetic Coding

Arithmetic coding offers a way to compress data and can be useful for data sources having a small alphabet. The length of an arithmetic code, instead of being fixed relative to the number of symbols being encoded, depends on the statistical frequency with which the source produces each symbol from its alphabet. For long sequences from sources having skewed distributions and small alphabets, arithmetic coding compresses better than Huffman coding.

The `arithenco` and `arithdeco` functions support arithmetic coding and decoding.

Represent Arithmetic Coding Parameters

Arithmetic coding requires statistical information about the source of the data being encoded. In particular, the `counts` input argument in the `arithenco` and `arithdeco` functions lists the frequency with which the source produces each symbol in its alphabet. You can determine the frequencies by studying a set of test data from the source. The set of test data can have any size you choose, as long as each symbol in the alphabet has a nonzero frequency.

For example, before encoding data from a source that produces 10 x's, 10 y's, and 80 z's in a typical 100-symbol set of test data, define

```
counts = [10 10 80];
```

Alternatively, if a larger set of test data from the source contains 22 x's, 23 y's, and 185 z's, then define

```
counts = [22 23 185];
```

Create and Decode an Arithmetic Code

Encode and decode a sequence from a source having three symbols.

Create a sequence vector containing symbols from the set of {1,2,3}.

```
seq = [3 3 1 3 3 3 3 2 3];
```

Set the `counts` vector to define an encoder that produces 10 ones, 20 twos, and 70 threes from a typical 100-symbol set of test data.

```
counts = [10 20 70];
```

Apply the arithmetic encoder and decoder functions.

```
code = arithenco(seq,counts);  
dseq = arithdeco(code,counts,length(seq));
```

Verify that the decoder output matches the original input sequence.

```
isequal(seq,dseq)
```

```
ans = logical  
     1
```

Quantization

In this section...

“Represent Partitions” on page 8-221

“Represent Codebooks” on page 8-221

“Determine Which Interval Each Input Is In” on page 8-222

“Optimize Quantization Parameters” on page 8-222

“Quantize a Signal” on page 8-224

Represent Partitions

Scalar quantization is a process that maps all inputs within a specified range to a common value. This process maps inputs in a different range of values to a different common value. In effect, scalar quantization digitizes an analog signal. Two parameters determine a quantization: a partition on page 8-2 and a codebook on page 8-3.

A quantization partition defines several contiguous, nonoverlapping ranges of values within the set of real numbers. To specify a partition in the MATLAB environment, list the distinct endpoints of the different ranges in a vector.

For example, if the partition separates the real number line into the four sets

- $\{x: x \leq 0\}$
- $\{x: 0 < x \leq 1\}$
- $\{x: 1 < x \leq 3\}$
- $\{x: 3 < x\}$

then you can represent the partition as the three-element vector

```
partition = [0,1,3];
```

The length of the partition vector is one less than the number of partition intervals.

Represent Codebooks

A codebook tells the quantizer which common value to assign to inputs that fall into each range of the partition. Represent a codebook as a vector whose length is the same as the number of partition intervals. For example, the vector

```
codebook = [-1, 0.5, 2, 3];
```

is one possible codebook for the partition [0, 1, 3].

Determine Which Interval Each Input Is In

The `quantiz` function also returns a vector that tells which interval each input is in. For example, the output below says that the input entries lie within the intervals labeled 0, 6, and 5, respectively. Here, the 0th interval consists of real numbers less than or equal to 3; the 6th interval consists of real numbers greater than 8 but less than or equal to 9; and the 5th interval consists of real numbers greater than 7 but less than or equal to 8.

```
partition = [3,4,5,6,7,8,9];  
index = quantiz([2 9 8],partition)
```

The output is

```
index =  
  
    0  
    6  
    5
```

If you continue this example by defining a codebook vector such as

```
codebook = [3,3,4,5,6,7,8,9];
```

then the equation below relates the vector `index` to the quantized signal `quants`.

```
quants = codebook(index+1);
```

This formula for `quants` is exactly what the `quantiz` function uses if you instead phrase the example more concisely as below.

```
partition = [3,4,5,6,7,8,9];  
codebook = [3,3,4,5,6,7,8,9];  
[index,quants] = quantiz([2 9 8],partition,codebook);
```

Optimize Quantization Parameters

- “Section Overview” on page 8-223
- “Example: Optimizing Quantization Parameters” on page 8-223

Section Overview

Quantization distorts a signal. You can reduce distortion by choosing appropriate partition and codebook parameters. However, testing and selecting parameters for large signal sets with a fine quantization scheme can be tedious. One way to produce partition and codebook parameters easily is to optimize them according to a set of so-called *training data*.

Note The training data you use should be typical of the kinds of signals you will actually be quantizing.

Example: Optimizing Quantization Parameters

The `lloyds` function optimizes the partition and codebook according to the Lloyd algorithm. The code below optimizes the partition and codebook for one period of a sinusoidal signal, starting from a rough initial guess. Then it uses these parameters to quantize the original signal using the initial guess parameters as well as the optimized parameters. The output shows that the mean square distortion after quantizing is much less for the optimized parameters. The `quantiz` function automatically computes the mean square distortion and returns it as the third output parameter.

```
% Start with the setup from 2nd example in "Quantizing a Signal."
t = [0:.1:2*pi];
sig = sin(t);
partition = [-1:.2:1];
codebook = [-1.2:.2:1];
% Now optimize, using codebook as an initial guess.
[partition2,codebook2] = lloyds(sig,codebook);
[index,quants,distor] = quantiz(sig,partition,codebook);
[index2,quant2,distor2] = quantiz(sig,partition2,codebook2);
% Compare mean square distortions from initial and optimized
[distor, distor2] % parameters.
```

The output is

```
ans =
    0.0148    0.0024
```

Quantize a Signal

- “Scalar Quantization Example 1” on page 8-224
- “Scalar Quantization Example 2” on page 8-224

Scalar Quantization Example 1

The code below shows how the `quantiz` function uses `partition` and `codebook` to map a real vector, `samp`, to a new vector, `quantized`, whose entries are either -1, 0.5, 2, or 3.

```
partition = [0,1,3];
codebook = [-1, 0.5, 2, 3];
samp = [-2.4, -1, -.2, 0, .2, 1, 1.2, 1.9, 2, 2.9, 3, 3.5, 5];
[index,quantized] = quantiz(samp,partition,codebook);
quantized
```

The output is below.

```
quantized =

Columns 1 through 6
-1.0000 -1.0000 -1.0000 -1.0000 0.5000 0.5000

Columns 7 through 12
2.0000 2.0000 2.0000 2.0000 2.0000 3.0000

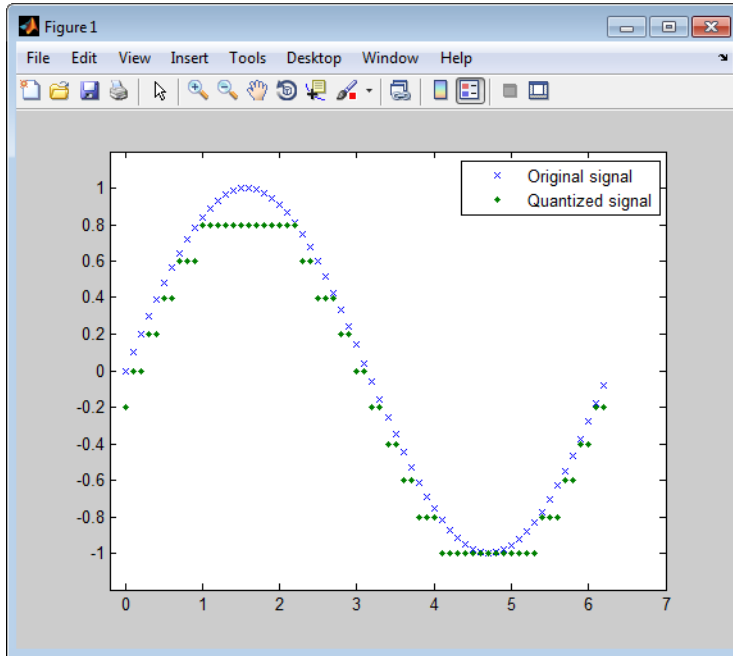
Column 13
3.0000
```

Scalar Quantization Example 2

This example illustrates the nature of scalar quantization more clearly. After quantizing a sampled sine wave, it plots the original and quantized signals. The plot contrasts the x's that make up the sine curve with the dots that make up the quantized signal. The vertical coordinate of each dot is a value in the vector `codebook`.

```
t = [0:.1:2*pi]; % Times at which to sample the sine function
sig = sin(t); % Original signal, a sine wave
partition = [-1:.2:1]; % Length 11, to represent 12 intervals
codebook = [-1.2:.2:1]; % Length 12, one entry for each interval
[index,quants] = quantiz(sig,partition,codebook); % Quantize.
```

```
plot(t,sig,'x',t,quants,'.')  
legend('Original signal','Quantized signal');  
axis([-0.2 7 -1.2 1.2])
```



OFDM Modulation

- “OFDM with User-Specified Pilot Indices” on page 9-2
- “SER Simulation for OFDM Link” on page 9-7
- “OFDM with MIMO Simulation” on page 9-10
- “Gray Coded 8-PSK” on page 9-15
- “CPM Phase Tree” on page 9-23
- “Filtered QPSK vs. MSK” on page 9-28
- “GMSK vs. MSK” on page 9-32
- “GMSK vs. MSK” on page 9-36
- “Gray Coded 8-PSK” on page 9-42
- “Soft Decision GMSK Demodulator” on page 9-49
- “16-PSK with Custom Symbol Mapping” on page 9-56
- “General QAM Modulation in AWGN Channel” on page 9-60
- “FM Modulate and Demodulate a Sinusoidal Signal” on page 9-63

OFDM with User-Specified Pilot Indices

This example shows how to construct an orthogonal frequency division modulation (OFDM) modulator/demodulator pair and to specify their pilot indices. The OFDM modulator System object enables you to specify pilot subcarrier indices consistent with the constraints described in `comm.OFDMModulator.info`. In this example, for OFDM transmission over a 3x2 channel, pilot indices are created for each of the three transmit antennas. Additionally, the pilot indices differ between odd and even symbols.

Create an OFDM modulator object having five symbols, three transmit antennas, and length six windowing.

```
ofdmMod = comm.OFDMModulator('FFTLength',256, ...  
    'NumGuardBandCarriers',[12; 11], ...  
    'NumSymbols', 5, ...  
    'NumTransmitAntennas', 3, ...  
    'PilotInputPort',true, ...  
    'Windowing', true, ...  
    'WindowLength', 6);
```

Specify pilot indices for even and odd symbols for the first transmit antenna.

```
pilotIndOdd = [20; 58; 96; 145; 182; 210];  
pilotIndEven = [35; 73; 111; 159; 197; 225];
```

```
pilotIndicesAnt1 = cat(2, pilotIndOdd, pilotIndEven, pilotIndOdd, ...  
    pilotIndEven, pilotIndOdd);
```

Generate pilot indices for the second and third antennas based on the indices specified for the first antenna. Concatenate the indices for the three antennas and assign them to the `PilotCarrierIndices` property.

```
pilotIndicesAnt2 = pilotIndicesAnt1 + 5;  
pilotIndicesAnt3 = pilotIndicesAnt1 - 5;
```

```
ofdmMod.PilotCarrierIndices = cat(3, pilotIndicesAnt1, pilotIndicesAnt2, pilotIndicesAnt3);
```

Create an OFDM demodulator with two receive antennas based on the existing OFDM modulator System object. Determine the data and pilot dimensions using the `info` function.

```
ofdmDemod = comm.OFDMDemodulator(ofdmMod);  
ofdmDemod.NumReceiveAntennas = 2;
```

```
dims = info(ofdmMod)

dims = struct with fields:
    DataInputSize: [215 5 3]
    PilotInputSize: [6 5 3]
    OutputSize: [1360 3]
```

Generate data and pilot symbols for the OFDM modulator given the array sizes specified in `modDim`.

```
dataIn = complex(randn(dims.DataInputSize), randn(dims.DataInputSize));
pilotIn = complex(randn(dims.PilotInputSize), randn(dims.PilotInputSize));
```

Apply OFDM modulation to the data and pilots.

```
modOut = ofdmMod(dataIn,pilotIn);
```

Pass the modulated data through a 3x2 random channel.

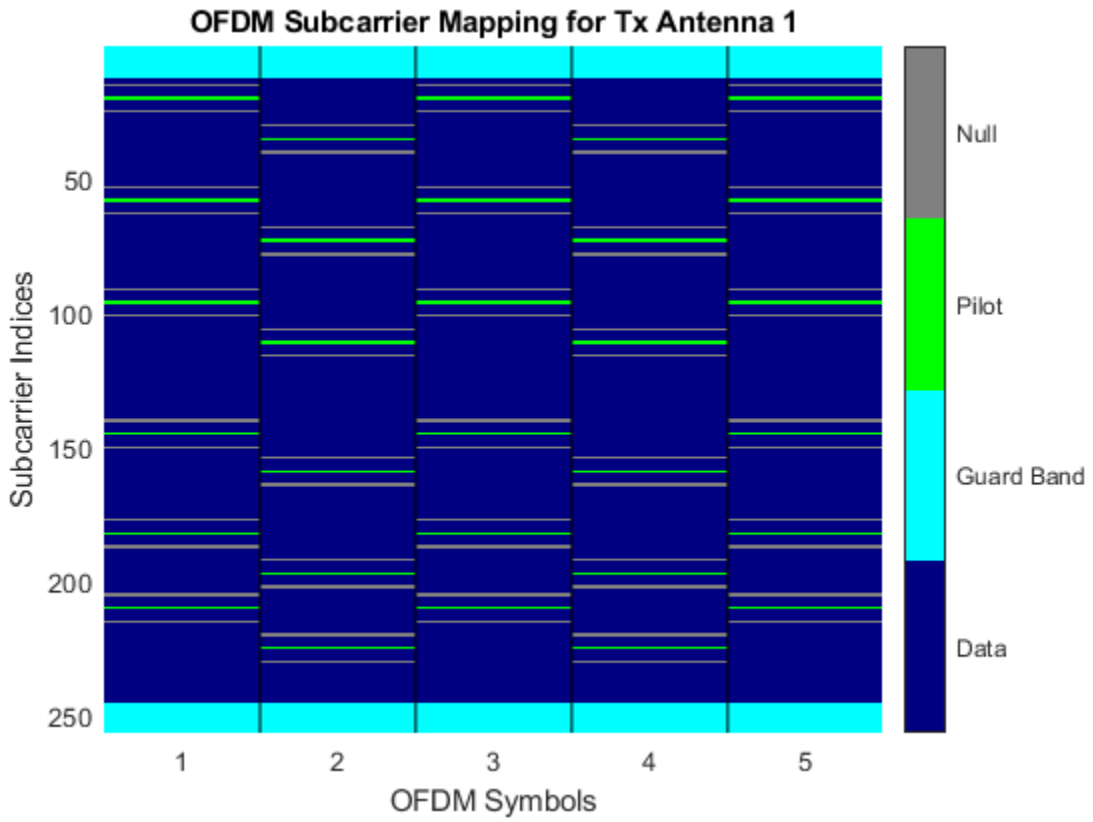
```
chanGain = complex(randn(3,2), randn(3,2));
chanOut = modOut * chanGain;
```

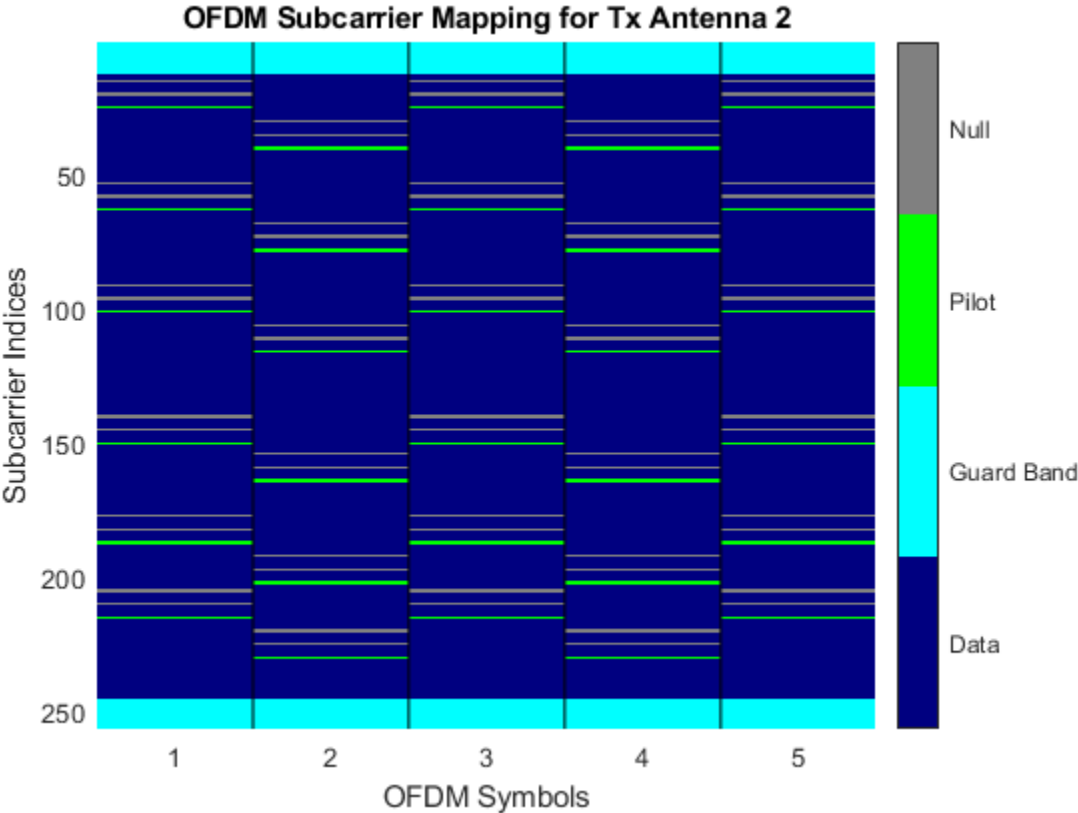
Demodulate the received data using the OFDM demodulator object.

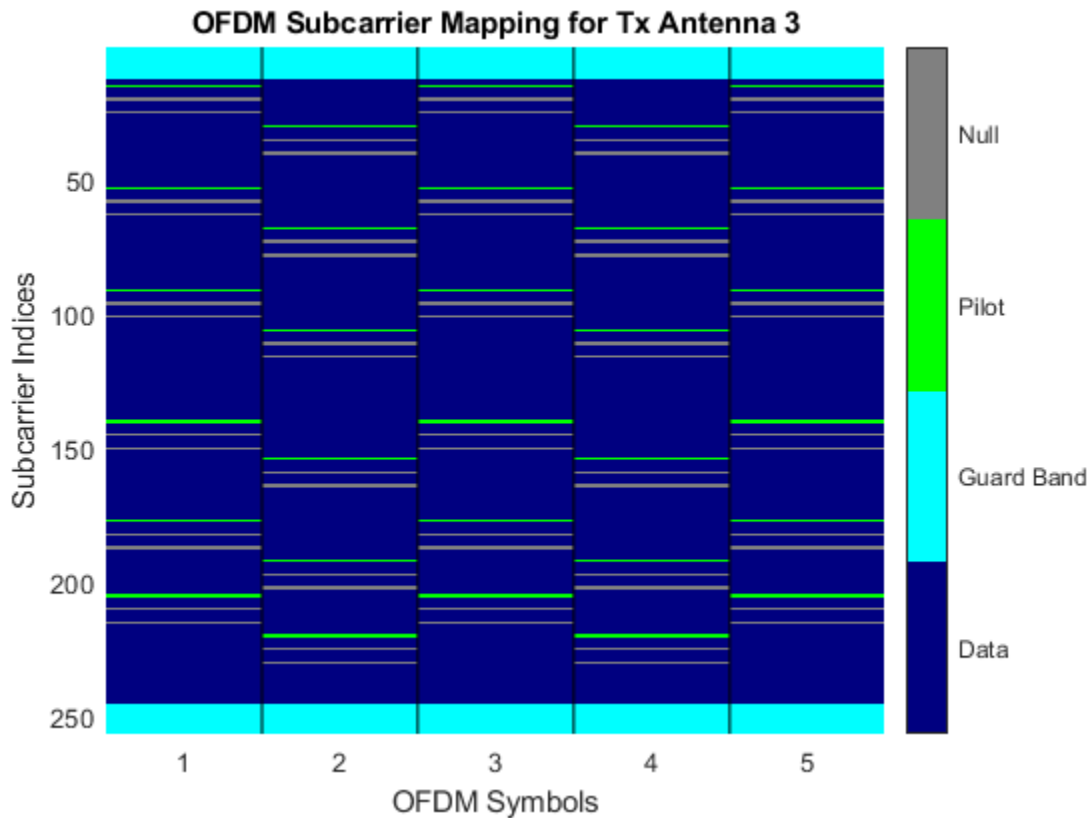
```
[dataOut,pilotOut] = ofdmDemod(chanOut);
```

Show the resource mapping for the three transmit antennas. The gray lines in the figure show the placement of custom nulls to avoid interference among antennas.

```
showResourceMapping(ofdmMod)
```







For the first transmit and first receive antenna pair, demonstrate that the input pilot signal matches the input pilot signal.

```
pilotCompare = abs(pilotIn(:,:,1)*chanGain(1,1)) - abs(pilotOut(:,:,1,1));
max(pilotCompare(:) < 1e-10)
```

```
ans = logical
     1
```

SER Simulation for OFDM Link

This example shows how to perform a symbol error rate (SER) simulation of an over-the-air OFDM communication link.

A basic communications link using OFDM modulation with QPSK symbols is simulated. There is a single transmit and a single receive antenna.

Create QPSK modulator and demodulator objects.

```
qpskMod = comm.QPSKModulator;
qpskDemod = comm.QPSKDemodulator;
```

Create a default OFDM modulator and demodulator pair.

```
ofdmMod = comm.OFDMModulator;
ofdmDemod = comm.OFDMDemodulator;
```

Use the `info` function to determine the required input dimensions for the OFDM modulator.

```
modDim = info(ofdmMod)

modDim = struct with fields:
    DataInputSize: [53 1]
    OutputSize: [80 1]
```

Set the number of frames. Determine the number of OFDM symbols per frame from the `modDim.DataInputSize` array.

```
nFrames = 10000;
nSymbolsPerFrame = modDim.DataInputSize(1);
```

Create an error rate counter with a reset input port. Initialize the symbol error rate vector, SER.

```
errRate = comm.ErrorRate('ResetInputPort',true);
SER = zeros(nFrames,1);
```

Run the simulation over 10000 OFDM frames (530000 symbols). During loop execution, generate a random data vector with length equal to the required number of symbols per frame, Apply QPSK modulation and then apply OFDM modulation. Pass the OFDM modulated data through the AWGN channel and then apply OFDM demodulation.

Demodulate the resultant QPSK data and compare it with the original data to determine the symbol error rate.

```
snr = [8 9 10 11];
meanSER = zeros(size(snr));
for j = 1:length(snr)
    for k = 1:nFrames
        % Generate random data for each OFDM frame
        data = randi([0 3],nSymbolsPerFrame,1);

        % Apply QPSK modulation
        txQPSK = qpskMod(data);

        % Apply OFDM modulation
        txSig = ofdmMod(txQPSK);

        % Pass OFDM signal through AWGN channel
        rxSig = awgn(txSig,snr(j),'measured');

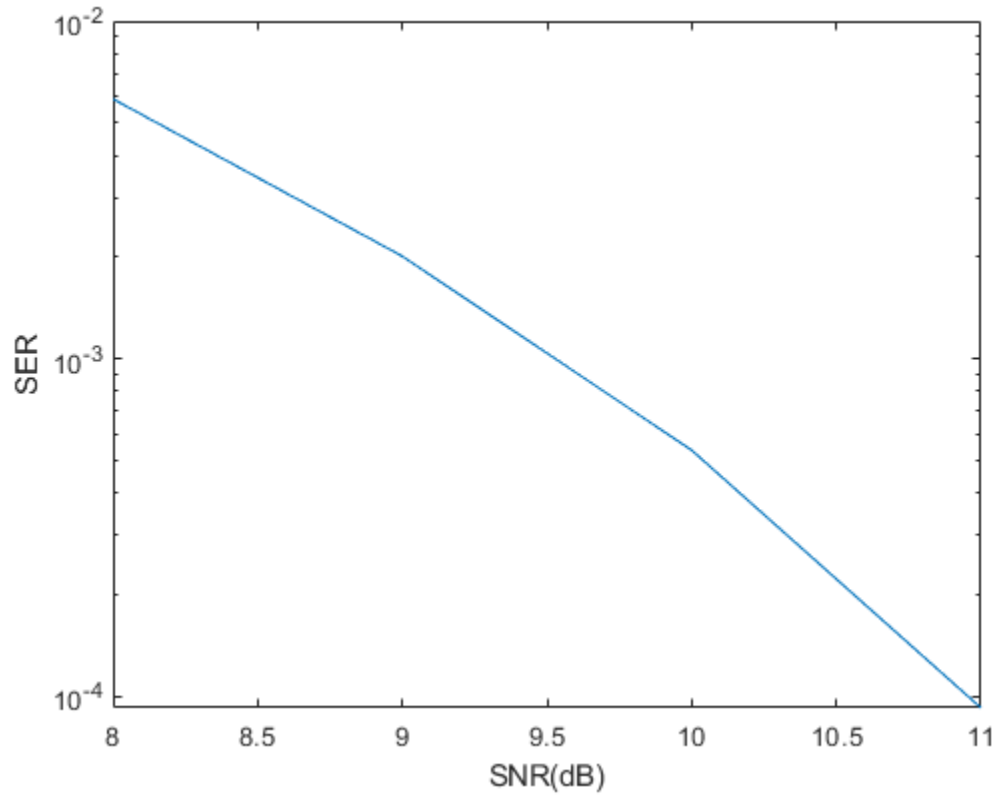
        % Demodulate OFDM data
        rxQPSK = ofdmDemod(rxSig);

        % Demodulate QPSK data
        rxData = qpskDemod(rxQPSK);

        % Compute BER
        errors = errRate(data,rxData,1);
        SER(k) = errors(1);
    end
    meanSER(j) = mean(SER);
end
```

Display the symbol error rate (SER) curve.

```
semilogy(snr,meanSER)
xlabel('SNR(dB)')
ylabel('SER')
```

OFDM with MIMO Simulation

This example shows how to use an OFDM modulator and demodulator in a simple, 2x2 MIMO error rate simulation. The OFDM parameters are based on the 802.11n standard.

Create a QPSK modulator and demodulator pair.

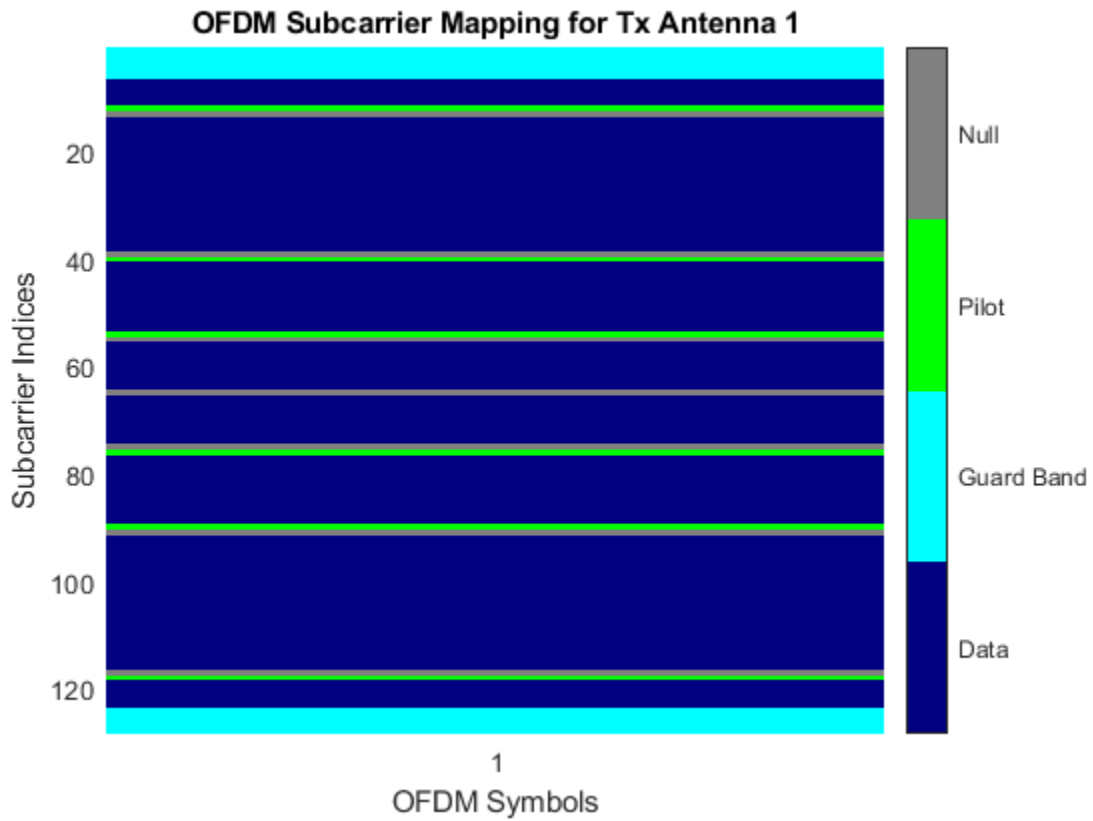
```
qpskMod = comm.QPSKModulator;  
qpskDemod = comm.QPSKDemodulator;
```

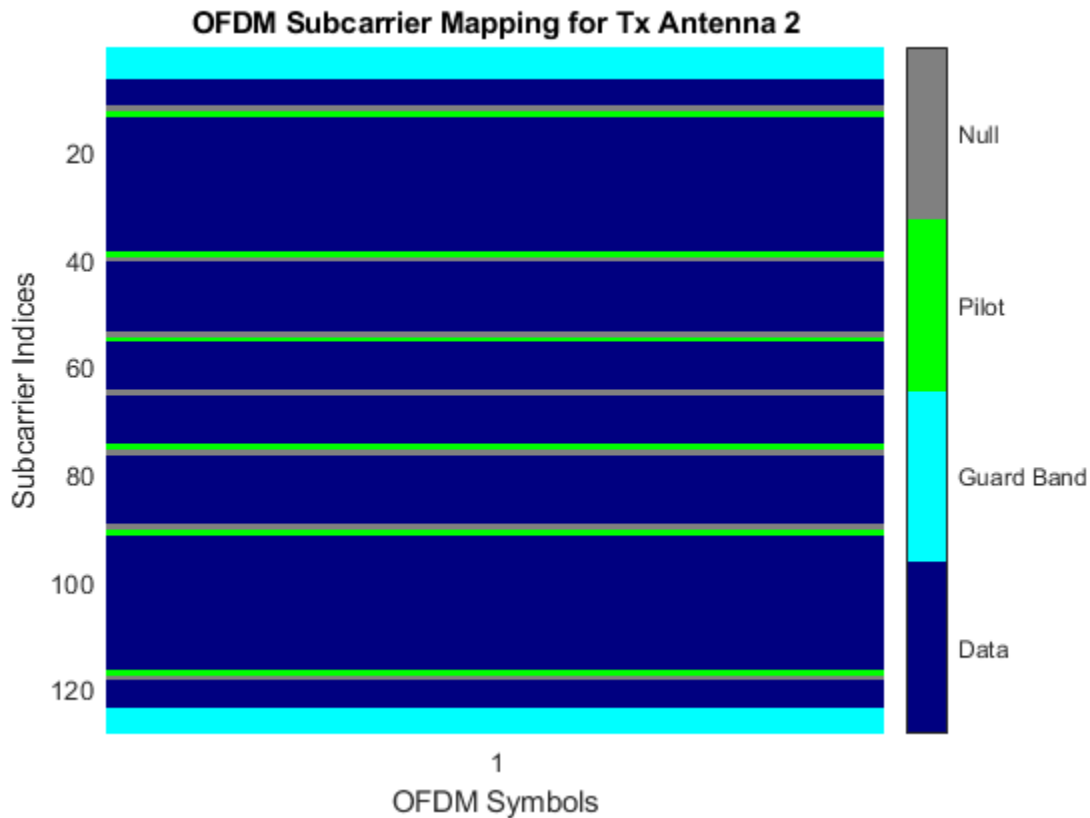
Create an OFDM modulator and demodulator pair with user-specified pilot indices, an inserted DC null, two transmit antennas, and two receive antennas. Specify pilot indices that vary across antennas.

```
ofdmMod = comm.OFDMModulator('FFTLength',128,'PilotInputPort',true,...  
    'PilotCarrierIndices',cat(3,[12; 40; 54; 76; 90; 118],...  
    [13; 39; 55; 75; 91; 117]),'InsertDCNull',true,...  
    'NumTransmitAntennas',2);  
ofdmDemod = comm.OFDMDemodulator(ofdmMod);  
ofdmDemod.NumReceiveAntennas = 2;
```

Show the resource mapping of pilot subcarriers for each transmit antenna. The gray lines in the figure denote the insertion of null subcarriers to minimize pilot signal interference.

```
showResourceMapping(ofdmMod)
```





Determine the dimensions of the OFDM modulator by using the `info` method.

```
ofdmModDim = info(ofdmMod);

numData = ofdmModDim.DataInputSize(1); % Number of data subcarriers
numSym = ofdmModDim.DataInputSize(2); % Number of OFDM symbols
numTxAnt = ofdmModDim.DataInputSize(3); % Number of transmit antennas
```

Generate data symbols to fill 100 OFDM frames.

```
nframes = 100;
data = randi([0 3], nframes*numData, numSym, numTxAnt);
```

Apply QPSK modulation to the random symbols and reshape the resulting column vector to match the OFDM modulator requirements.

```
modData = qpskMod(data(:));
modData = reshape(modData,nframes*numData,numSym,numTxAnt);
```

Create an error rate counter.

```
errorRate = comm.ErrorRate;
```

Simulate the OFDM system over 100 frames assuming a flat, 2x2, Rayleigh fading channel. Remove the effects of multipath fading using a simple, least squares solution, and demodulate the OFDM waveform and QPSK data. Generate error statistics by comparing the original data with the demodulated data.

```
for k = 1:nframes

    % Find row indices for kth OFDM frame
    indData = (k-1)*ofdmModDim.DataInputSize(1)+1:k*numData;

    % Generate random OFDM pilot symbols
    pilotData = complex(rand(ofdmModDim.PilotInputSize), ...
        rand(ofdmModDim.PilotInputSize));

    % Modulate QPSK symbols using OFDM
    dataOFDM = ofdmMod(modData(indData,:,:),pilotData);

    % Create flat, i.i.d., Rayleigh fading channel
    chGain = complex(randn(2,2),randn(2,2))/sqrt(2); % Random 2x2 channel

    % Pass OFDM signal through Rayleigh and AWGN channels
    receivedSignal = awgn(dataOFDM*chGain,30);

    % Apply least squares solution to remove effects of fading channel
    rxSigMF = chGain.' \ receivedSignal.';

    % Demodulate OFDM data
    receivedOFDMData = ofdmDemod(rxSigMF.');

    % Demodulate QPSK data
    receivedData = qpskDemod(receivedOFDMData(:));

    % Compute error statistics
    dataTmp = data(indData,:,:);
    errors = errorRate(dataTmp(:),receivedData);
end
```

Display the error statistics.

```
fprintf('\nSymbol error rate = %d from %d errors in %d symbols\n',errors)
Symbol error rate = 9.471154e-02 from 1970 errors in 20800 symbols
```

Gray Coded 8-PSK

This example shows a communications system with Gray-coded 8-ary phase shift keying (8-PSK) modulation using communications related System objects. Gray coding is a technique that multilevel modulation schemes often use to minimize the bit error rate. It consists of ordering modulation symbols so that the binary representations of adjacent symbols differ by only one bit.

In this section...

- “Introduction” on page 9-15
- “Initialization” on page 9-17
- “Stream Processing Loop” on page 9-19
- “Cleanup” on page 9-20
- “Conclusions” on page 9-20

Introduction

This example modulates data using the 8-PSK method. The data passes through an AWGN channel, and is demodulated using an 8-PSK demodulator. An error rate calculator System object measures the symbol and bit error rates.

In this communications system, the PSK Modulator System object:

- Accepts binary-valued inputs that represent integers between 0 and $M - 1$. M is the modulation order and is equal to 8 for 8-PSK modulation.
- Maps binary representations to constellation points using Gray-coded ordering.
- Produces unit-magnitude complex phasor outputs, with evenly spaced phases between 0 and $2\pi(M - 1)/M$.

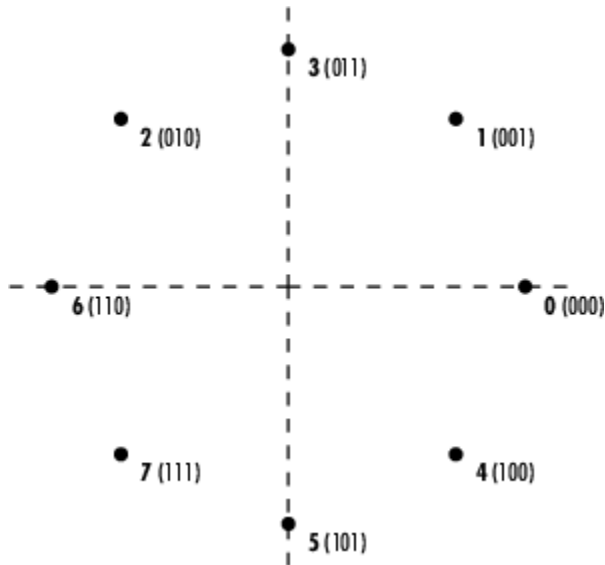
The following table indicates the relationship between binary representations in the input and phasors in the output. The second column of the table is an intermediate representation that the System object uses in its computations.

Modulator Input	Gray-Coded Ordering	Modulator Output
000	0	$\exp(0)$

Modulator Input	Gray-Coded Ordering	Modulator Output
001	1	$\exp(j\pi/4)$
010	3	$\exp(j3\pi/4)$
011	2	$\exp(j\pi/2) = \exp(j2\pi/4)$
100	7	$\exp(j7\pi/4)$
101	6	$\exp(j3\pi/2) = \exp(j6\pi/4)$
110	4	$\exp(j\pi) = \exp(j4\pi/4)$
111	5	$\exp(j5\pi/4)$

The table below sorts the first two columns from the previous table, according to the output values. This sorting makes it clearer that there is only a 1 bit difference between neighboring symbols. In the following figure, notice that the numbers in the second column of the table appear in counterclockwise order.

Modulator Output	Modulator Input
$\exp(0)$	000
$\exp(j\pi/4)$	001
$\exp(j\pi/2) = \exp(j2\pi/4)$	011
$\exp(j3\pi/4)$	010
$\exp(j\pi) = \exp(j4\pi/4)$	110
$\exp(j5\pi/4)$	111
$\exp(j3\pi/2) = \exp(j6\pi/4)$	101
$\exp(j7\pi/4)$	100



Initialization

This section of the code initializes the system variables. It also creates and configures the System objects used in this example.

Set the modulation order to 8 for 8-PSK modulation. Run the simulation until either the specified maximum number of bit errors (`maxNumErrs`) or the maximum number of bits (`maxNumBits`) is reached. This simulation iterates over a number of bit energy to noise power spectral density E_b/N_0 values.

```
M = 8; % Modulation order
SamplesPerFrame = 10000; % Symbols processed for each iteration of the
                          % stream processing loop

% Initialize variables used to determine when to stop processing bits
maxNumErrs=100;
maxNumBits=1e8;

% Since the AWGN Channel as well as the RANDI function uses the default
% random stream, the following commands are executed so that the results
% will be repeatable, i.e. same results will be obtained for every run of
% the example. The default stream will be restored at the end of the
% example.
```

```
prevState = rng;  
rng(529558);
```

Create an integer to bit converter (hInt2Bit) and a bit to integer converter (hBit2Int) System object to convert the randomly generated integer data to bits and the demodulated data bits back to integers

```
hInt2Bit = comm.IntegerToBit('BitsPerInteger',log2(M), ...  
                             'OutputDataType','uint8');  
hBit2Int = comm.BitToInteger('BitsPerInteger',log2(M), ...  
                             'OutputDataType','uint8');
```

Create and configure a PSK modulator (hMod) System object to map the binary input data to an 8-PSK gray coded constellation as well as a matching PSK demodulator (hDemod) System object

```
hMod = comm.PSKModulator('ModulationOrder',M, ...  
                         'SymbolMapping','gray', ...  
                         'PhaseOffset',0, ...  
                         'BitInput',true);  
hDemod = comm.PSKDemodulator('ModulationOrder',M, ...  
                             'SymbolMapping','gray', ...  
                             'PhaseOffset',0, ...  
                             'BitOutput',true, ...  
                             'OutputDataType','uint8', ...  
                             'DecisionMethod','Hard decision');
```

Create an AWGN channel System object to add additive white Gaussian noise to the modulated signal. The noise method is appropriately selected so it specifies the bit energy to noise power spectral density in the stream processing loop. Because the PSK modulator generates symbols with 1 Watt of power, the signal power property of the AWGN channel is also set to 1.

```
hChan = comm.AWGNChannel('NoiseMethod','Signal to noise ratio (Eb/No)', ...  
                        'BitsPerSymbol',log2(M), ...  
                        'SignalPower',1);
```

Create a symbol error rate calculator (hSymError) and a bit error rate calculator (hBitError) System object to compare the demodulated integer and bit data with the original source data. This comparison yields symbol error and bit error statistics. The output of the error rate calculator System object is a three-element vector containing the calculated error rate, the number of errors observed, and the amount of data processed. The simulation uses the three-element vector generated by hBitError to determine when to stop the simulation.

```
hSymError = comm.ErrorRate;
hBitError = comm.ErrorRate;
```

Stream Processing Loop

This section of the code calls the processing loop where data is gray coded, modulated, and demodulated using 8-PSK modulation. The loop simulates the communications system for Eb/No values in the range 0dB to 12dB in steps of 2dB.

```
% For each Eb/No value, simulation stops when either the maximum number of
% errors (maxNumErrs) or the maximum number of bits (maxNumBits) processed
% by the bit error rate calculator System object is reached.
EbNoVec = 0:2:12; % Eb/No values to simulate
SERVec = zeros(size(EbNoVec)); % Initialize SER history
BERVec = zeros(size(EbNoVec)); % Initialize BER history
for p = 1:length(EbNoVec)
    % Reset System objects
    reset(hSymError);
    reset(hBitError);
    hChan.EbNo = EbNoVec(p);
    % Reset SER / BER for the current Eb/No value
    SER = zeros(3,1); % Symbol Error Rate
    BER = zeros(3,1); % Bit Error Rate
    while (BER(2)<maxNumErrs) && (BER(3)<maxNumBits)
        % Generate random data
        txSym = randi([0 M-1], SamplesPerFrame, 1, 'uint8');
        txBits = step(hInt2Bit, txSym); % Convert symbols to bits
        tx = step(hMod, txBits); % Modulate
        rx = step(hChan, tx); % Add white Gaussian noise
        rxBits = step(hDemod, rx); % Demodulate
        rxSym = step(hBit2Int, rxBits); % Convert bits back to symbols

        % Calculate error rate
        SER = step(hSymError, txSym, rxSym); % Symbol Error Rate
        BER = step(hBitError, txBits, rxBits); % Bit Error Rate
    end
    % Save history of SER and BER values
    SERVec(p) = SER(1);
    BERVec(p) = BER(1);
end
```

Cleanup

Restore the default stream.

```
rng(prevState)
```

Conclusions

Analyze the data that the example produces and compare theoretical performance with simulation performance. The theoretical symbol error probability of MPSK is

$$P_E(M) = \operatorname{erfc}\left(\sqrt{\frac{E_s}{N_0}} \sin\left(\frac{\pi}{M}\right)\right)$$

where erfc is the complementary error function, E_s/N_0 is the ratio of energy in a symbol to noise power spectral density, and M is the number of symbols.

To determine the bit error probability, convert the symbol error probability, P_E , to its bit error equivalent. There is no general formula for the symbol to bit error conversion. Nevertheless, upper and lower limits are easy to establish. The actual bit error probability, P_b , can be shown to be bounded by

$$\frac{P_E(M)}{\log_2 M} \leq P_b \leq \frac{M/2}{M-1} P_E(M)$$

The lower limit corresponds to the case where the symbols have undergone Gray coding. The upper limit corresponds to the case of pure binary coding.

The following script plots the simulated symbol error rates (SERVec) and bit error rates (BERVec) together with the theoretical symbol error and bit error probabilities.

Calculate theoretical error probabilities.

```
[theorBER, theorSER] = berawgn(EbNoVec, 'psk', M, 'nondiff');
```

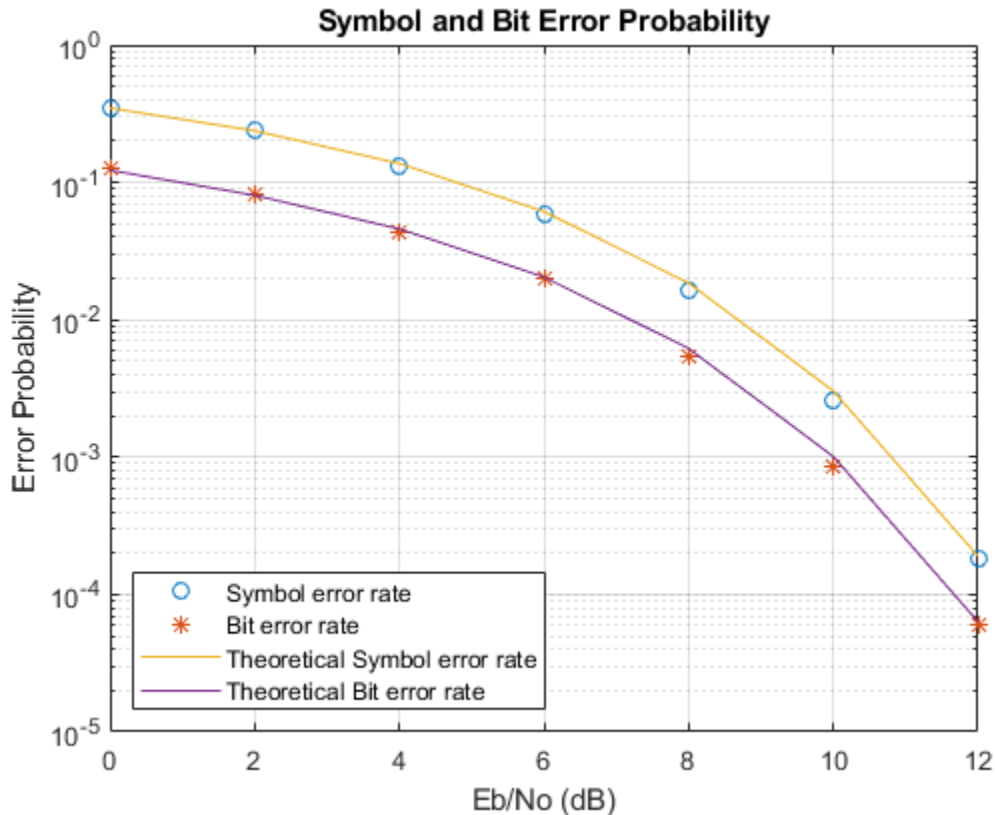
Plot the results.

```
figure;  
semilogy(EbNoVec, SERVec, 'o', EbNoVec, BERVec, '*', ...  
          EbNoVec, theorSER, '-', EbNoVec, theorBER, '-');  
legend('Symbol error rate', 'Bit error rate', ...  
       'Theoretical Symbol error rate', 'Theoretical Bit error rate', ...
```

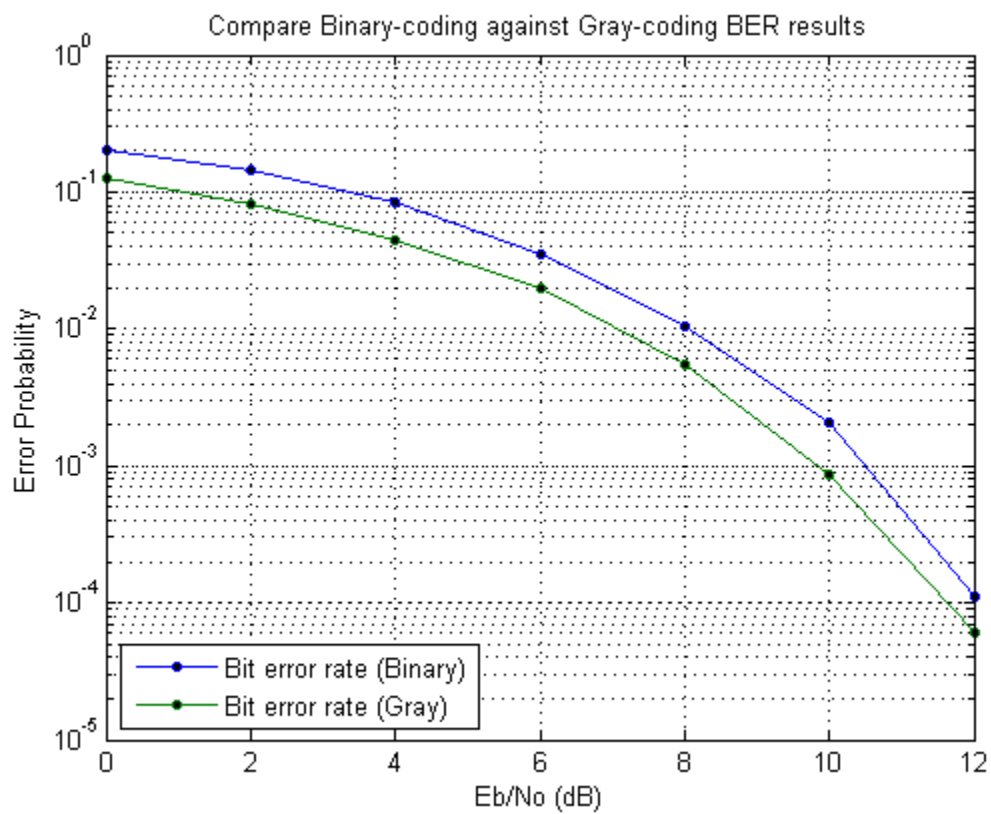
```

'Location','SouthWest');
xlabel ( 'Eb/No (dB)' ); ylabel( 'Error Probability' );
title ( 'Symbol and Bit Error Probability' ); grid on;

```



As a further exercise, you can compare Gray coding with pure binary coding by modifying the PSK modulator and PSK demodulator System objects so that their constellation ordering parameters are 'Binary' instead of 'Gray'. Setting this property and re-running the simulation should generate results similar to the following:



CPM Phase Tree

In this section...
“Structure of the Example” on page 9-23
“Results and Displays” on page 9-24
“Exploring the Example” on page 9-26

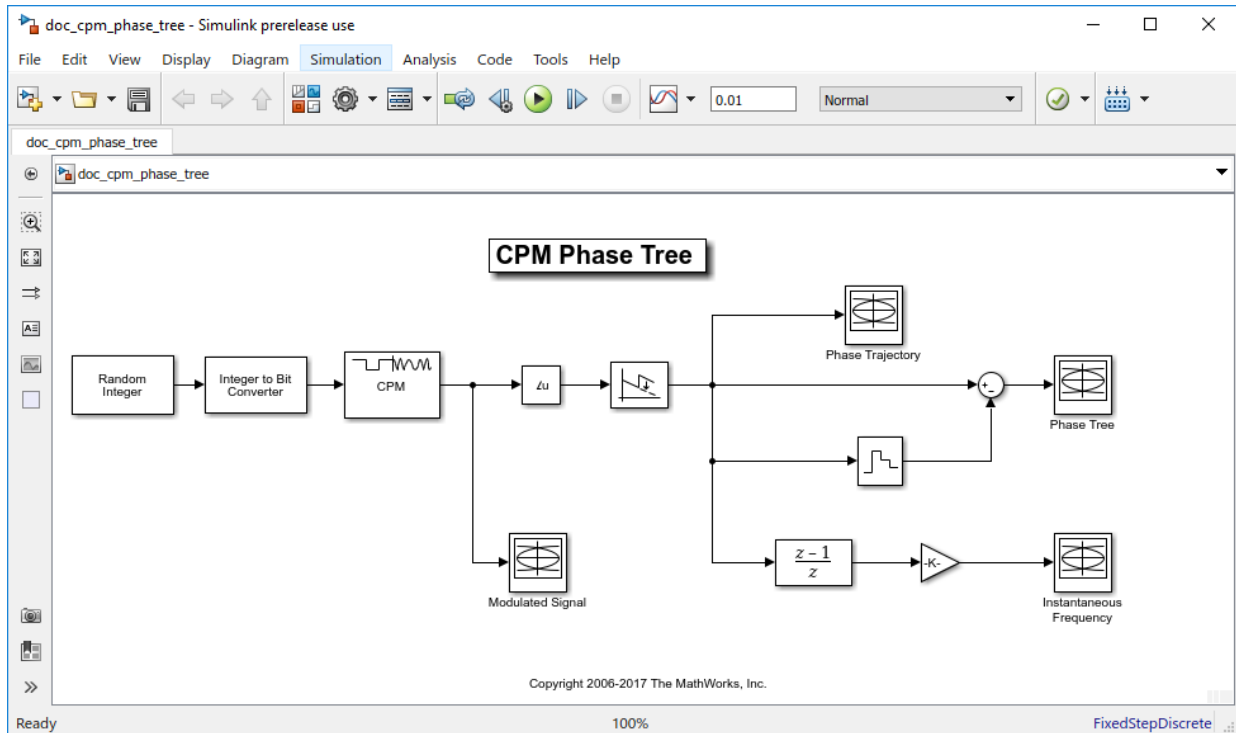
This model shows how to use the Eye Diagram block to view the phase trajectory, phase tree, and instantaneous frequency of a CPM modulated signal.

Structure of the Example

This example, `doc_cpm_phase_tree`, uses various Communications Toolbox, DSP System Toolbox, and Simulink blocks to model a baseband CPM signal.

In particular, the example model includes these blocks:

- Random Integer Generator block
- Integer to Bit Converter block
- CPM Modulator Baseband block
- Complex to Magnitude-Angle block
- Phase Unwrap block
- Zero-Order Hold block
- Discrete Transfer Fcn block
- Gain block
- Multiple copies of the Eye Diagram block

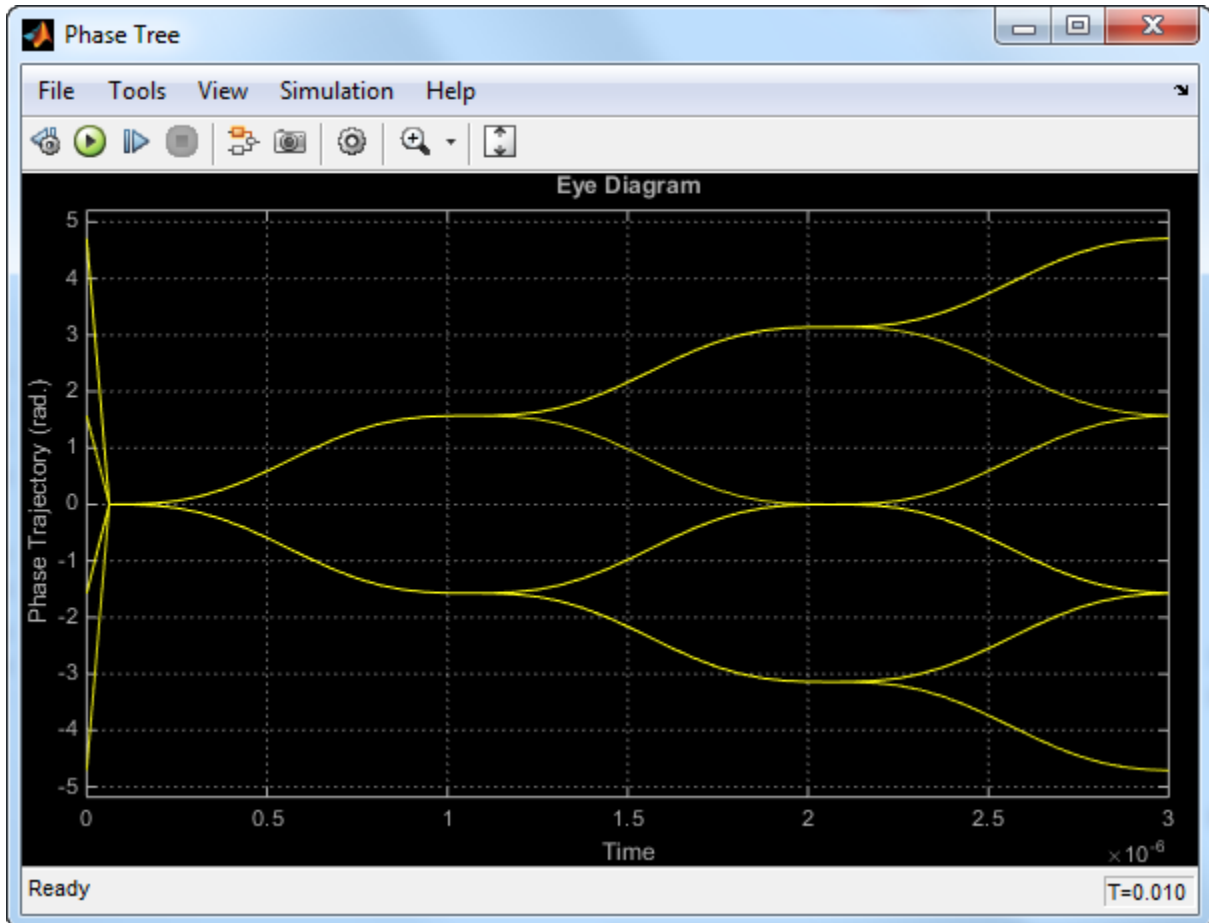


Results and Displays

When you run the example, several Eye Diagram blocks show how the CPM signal changes over time:

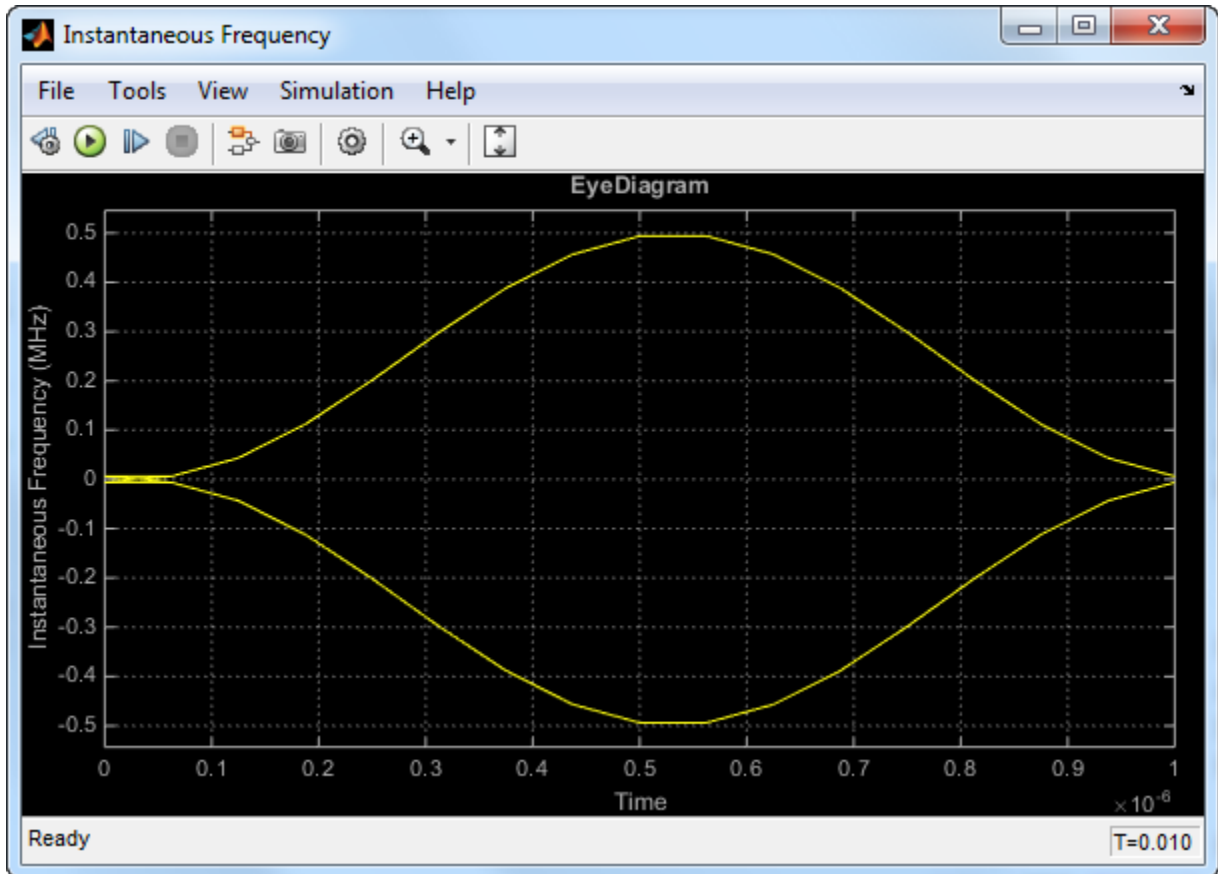
- The Modulated Signal block displays the in-phase and quadrature signals. Double-click the block to open the scope. The modulated signal is easy to see in the eye diagram only when the **Modulation index** parameter in the CPM Modulator Baseband block is set to 0.5. If you set the **Modulation index** to another value, for example 2/3, the features of the modulated signal are difficult to decipher for this more complex modulation. Unwrapping the phase and plotting it is another way to illustrate these more complex CPM modulated signals.
- The Phase Trajectory block displays the CPM phase. Double-click the block to open the scope. The Phase Trajectory block reveals that the signal phase is also difficult to view because it drifts with the data input to the modulator.

- The Phase Tree block displays the phase tree of the signal. The CPM phase is processed by a few simple blocks to make the CPM pulse shaping easier to view. This processing holds the phase at the beginning of the symbol interval and subtracts it from the signal. This resets the phase to zero every three symbols. The resulting plot shows the many phase trajectories that can be taken by the signal from any given symbol epoch.



- The Instantaneous Frequency block displays the instantaneous frequency of the signal. The CPM phase is differentiated to produce the frequency deviation of the signal. Viewing the CPM frequency signal enables you to observe the frequency deviation

qualitatively, as well as make quantitative observations, such as measuring peak frequency deviation.



Exploring the Example

To learn more about the example, try changing the following parameters in the CPM Modulator Baseband block:

- Change **Pulse length** to a value between 1 and 6.
- Change **Frequency pulse shape** to one of the other settings, such as Rectangular or Gaussian.

You can observe the effect of changing these parameters on the phase tree and instantaneous frequency of the modulated signal.

Filtered QPSK vs. MSK

In this section...
“Structure of the Example” on page 9-28
“Results and Displays” on page 9-29

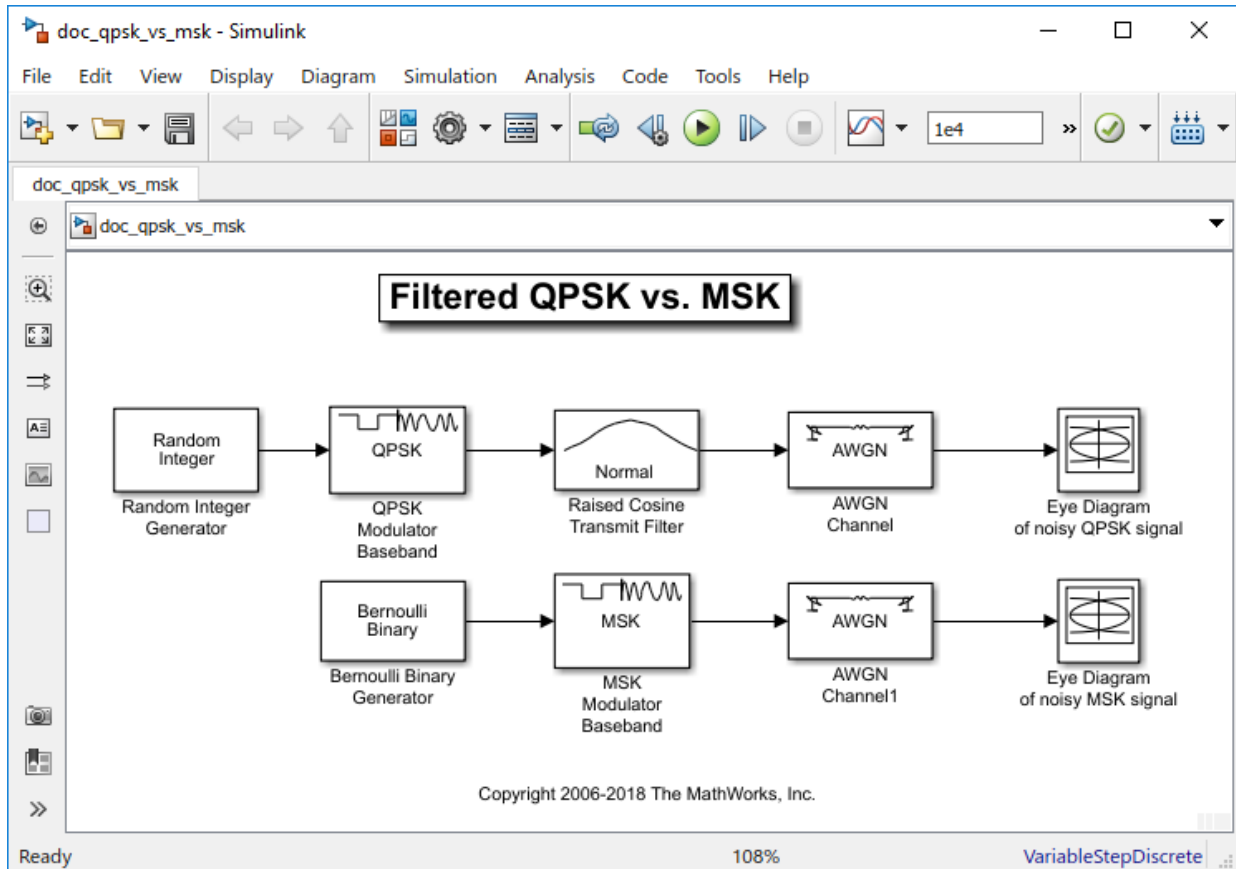
This model shows filtered quadrature phase shift keying (QPSK) and minimum shift keying (MSK) modulation schemes and visually compare them.

Structure of the Example

This example, `doc_gpsk_vs_msk`, uses Communications Toolbox blocks to model filtered QPSK and MSK modulation.

The example model includes these blocks:

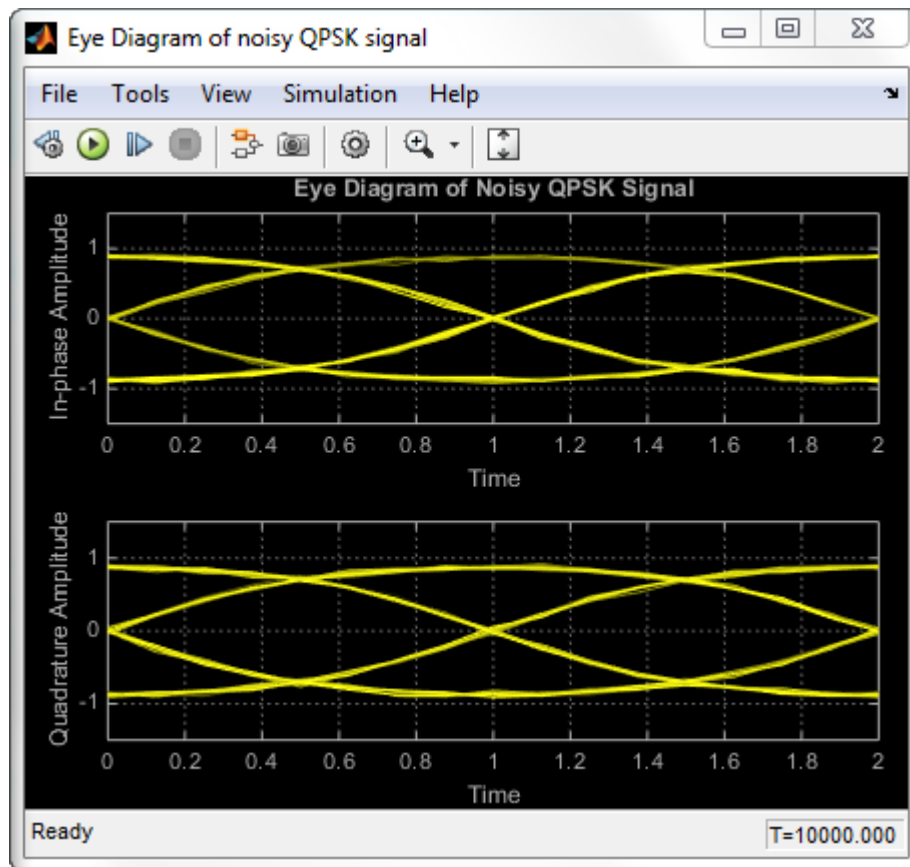
- Random Integer Generator block (for QPSK)
- Bernoulli Binary Generator block (for MSK)
- QPSK Modulator Baseband block
- MSK Modulator Baseband block
- Raised Cosine Transmit Filter block
- AWGN Channel block
- Eye Diagram block



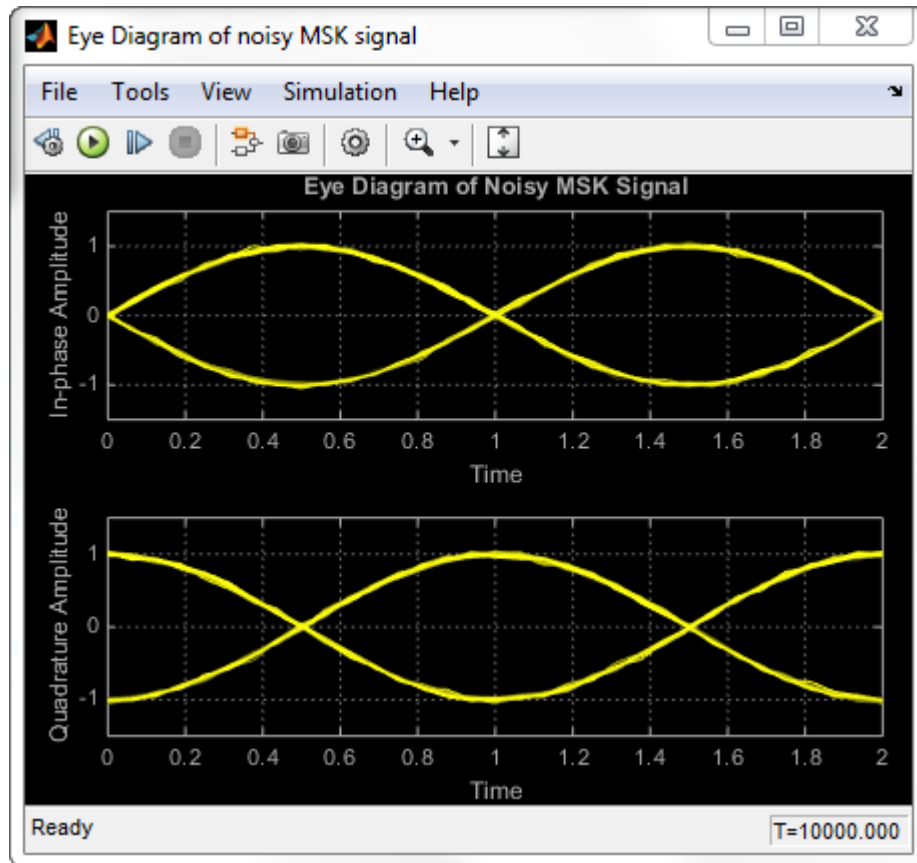
Results and Displays

The example uses eye diagram blocks to show the eye diagrams of filtered QPSK and MSK signals plus noise. You can observe that:

- 1 In filtered QPSK, the values of both the in-phase and quadrature components of the signal are permitted to change at any symbol interval.



- 2 However, for MSK, the symbol interval is half that for QPSK, but the in-phase and quadrature components change values in alternate symbol epochs. Therefore, the ideal sampling time for QPSK is 0.5, 1.5, 2.5,..., while the ideal sampling period for MSK is 0.5, 1.5, 2.5,... for the in-phase signal and 1, 2, 3,... for the quadrature signal.



GMSK vs. MSK

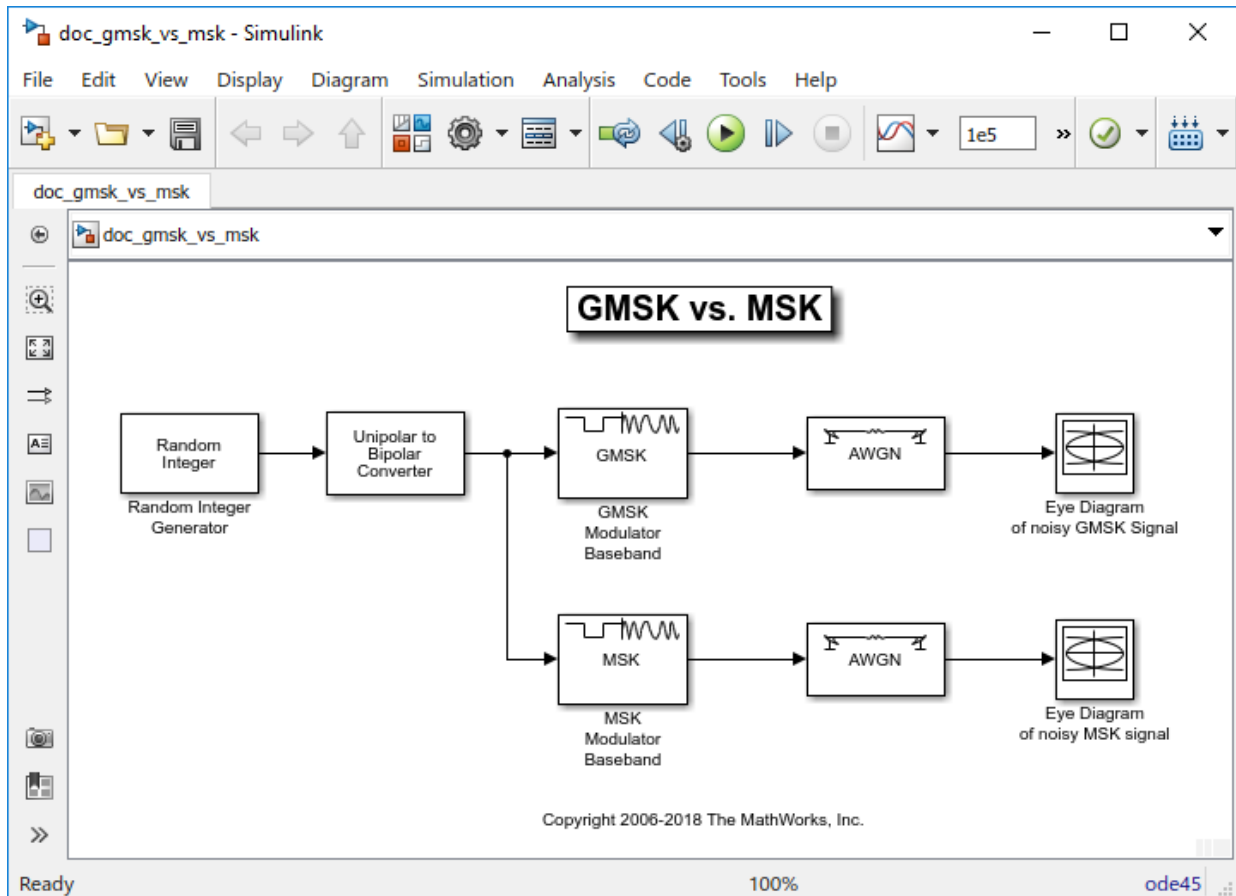
In this section...
“Structure of the Example” on page 9-32
“Results and Displays” on page 9-33

This model shows how to visually compare Gaussian minimum shift keying (GMSK) and minimum shift keying (MSK) modulation schemes.

Structure of the Example

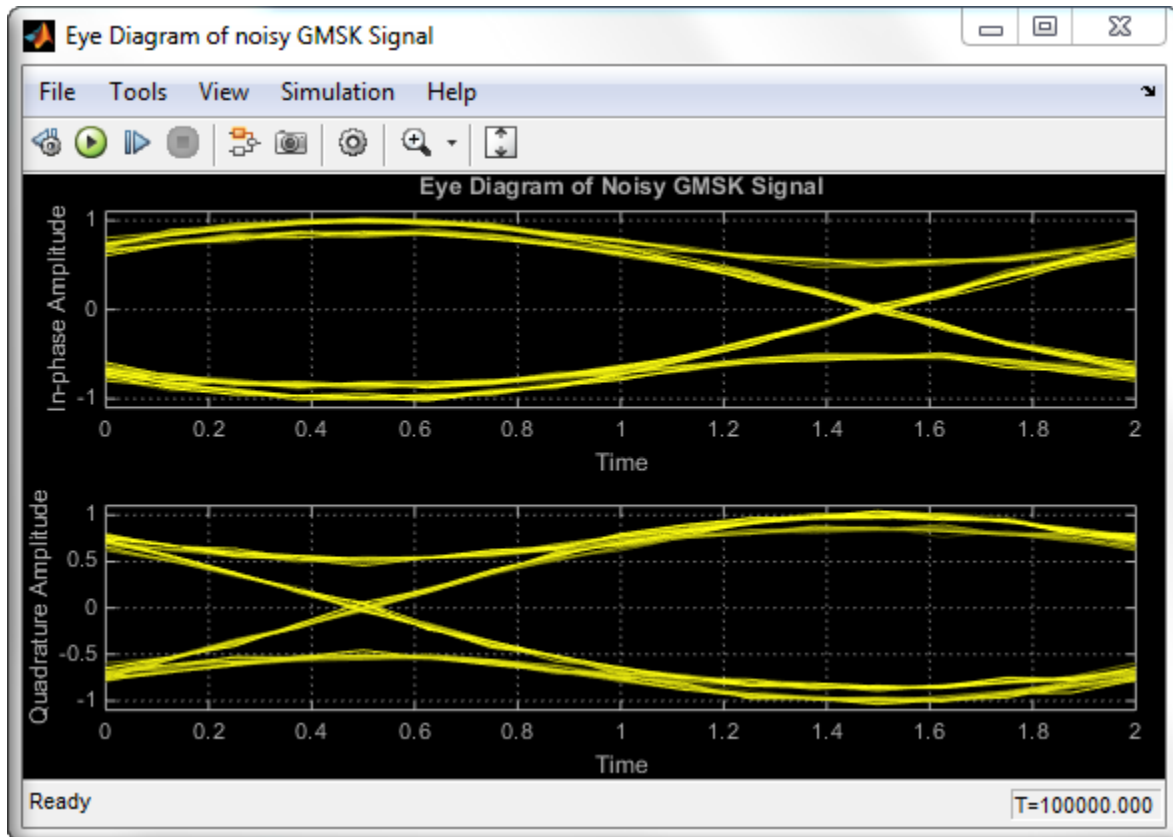
The example model, `doc_gmsk_vs_msk`, includes these blocks:

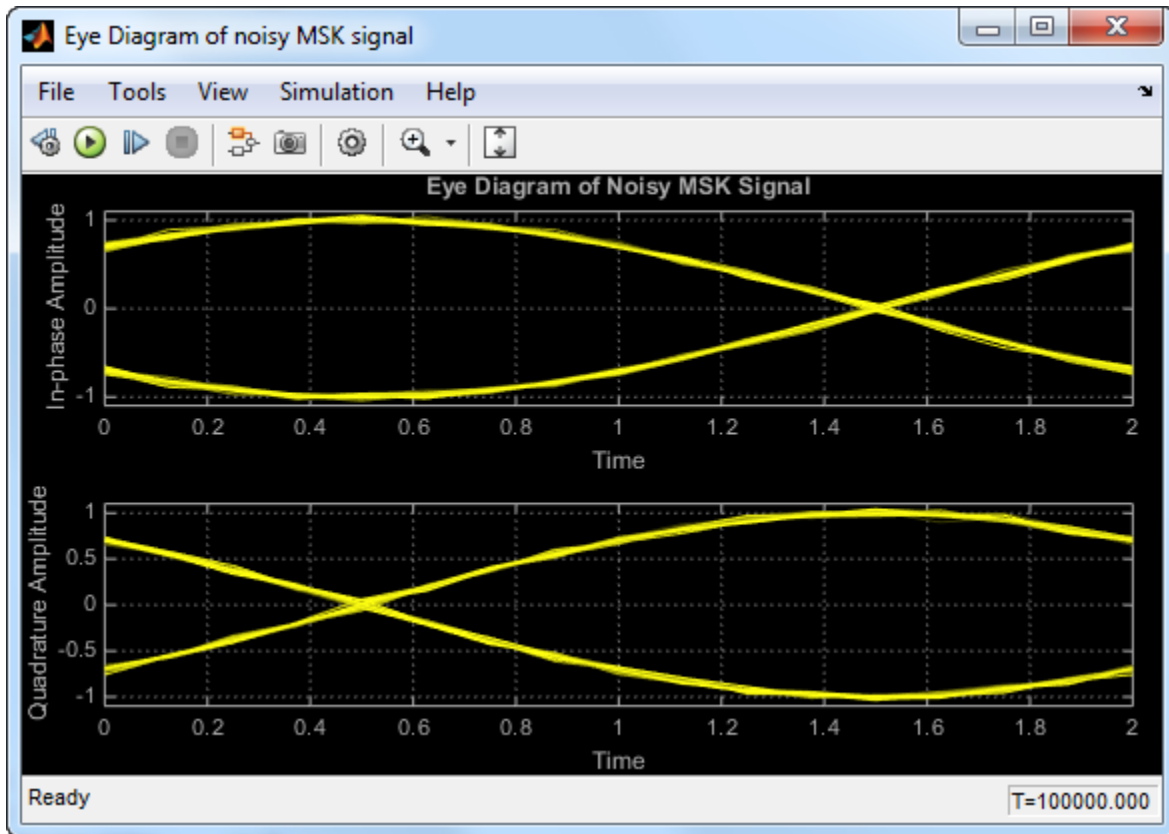
- Random Integer Generator block, which provides a source of uniformly distributed random integers in the range $[0, M-1]$, where M is the constellation size of the GMSK or MSK signal
- Unipolar to Bipolar Converter block
- GMSK Modulator Baseband block
- MSK Modulator Baseband block
- AWGN Channel block
- Eye Diagram block



Results and Displays

The example illustrates the difference between the two modulation schemes. The Eye Diagram blocks show the eye diagrams of the noisy GMSK and MSK signals.





The eye diagrams show the similarity between the GMSK and MSK signals when you set the **Pulse length** of the GMSK Modulator Baseband block to 1. Setting the **Pulse length** to 3 or 5 enables you to view the difference that a partial response modulation can have on the eye diagram. The number of paths increases, showing that the CPM waveform depends on values of the previous symbols as well as the present symbol. You can change the pulse length to 2 or 4, but you should change the **Phase offset** to $\pi/4$ for a better view of the modulated signal. In order to more clearly view the Gaussian pulse shape, you must use scopes that enable you to view the phase of the signal, as described in the “CPM Phase Tree” on page 9-23 example.

GMSK vs. MSK

Compare, using eye diagrams, Gaussian minimum shift keying (GMSK) and minimum shift keying (MSK) modulation schemes.

Set the samples per symbol variable.

```
sps = 8;
```

Generate random binary data.

```
data = randi([0 1],1000,1);
```

Create GMSK and MSK modulators that accept binary inputs. Set the `PulseLength` property of the GMSK modulator to 1.

```
gmskMod = comm.GMSKModulator('BitInput',true,'PulseLength',1, ...  
    'SamplesPerSymbol',sps);  
mskMod = comm.MSKModulator('BitInput',true,'SamplesPerSymbol',sps);
```

Modulate the data using the GMSK and MSK modulators.

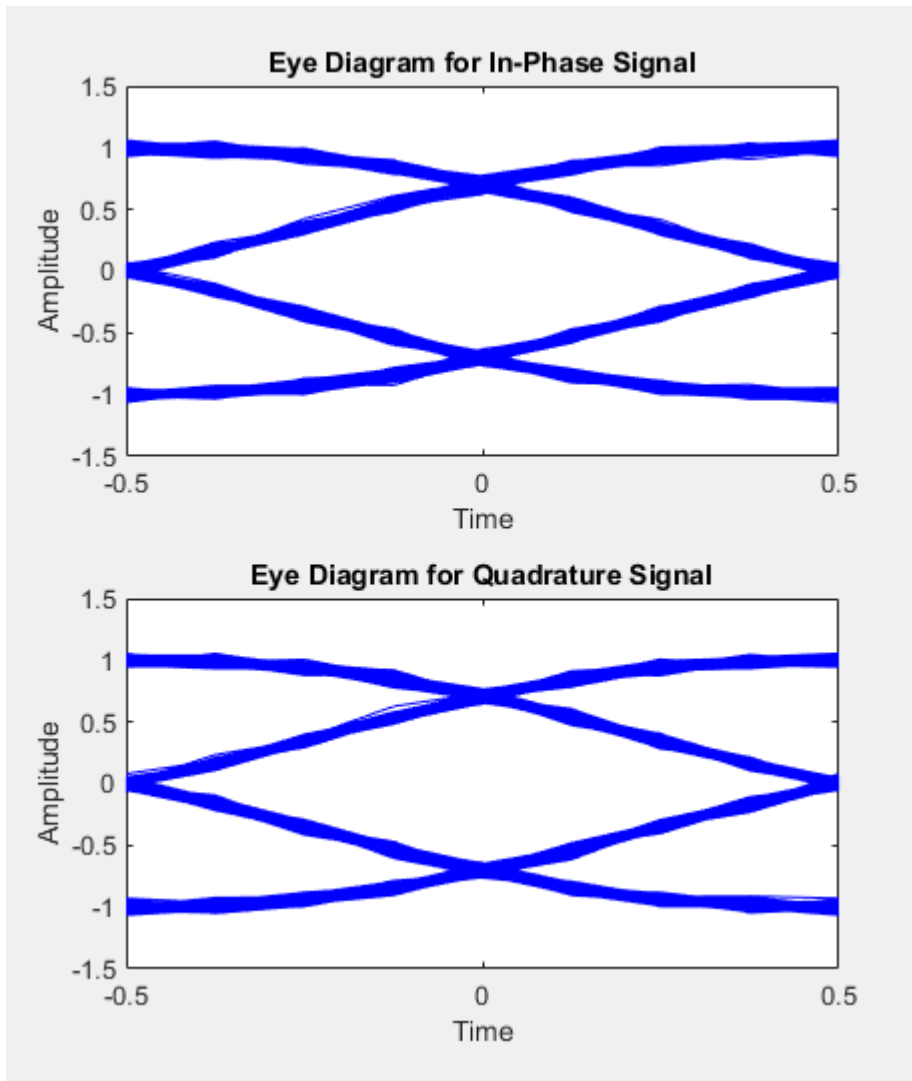
```
modSigGMSK = gmskMod(data);  
modSigMSK = mskMod(data);
```

Pass the modulated signals through an AWGN channel having an SNR of 30 dB.

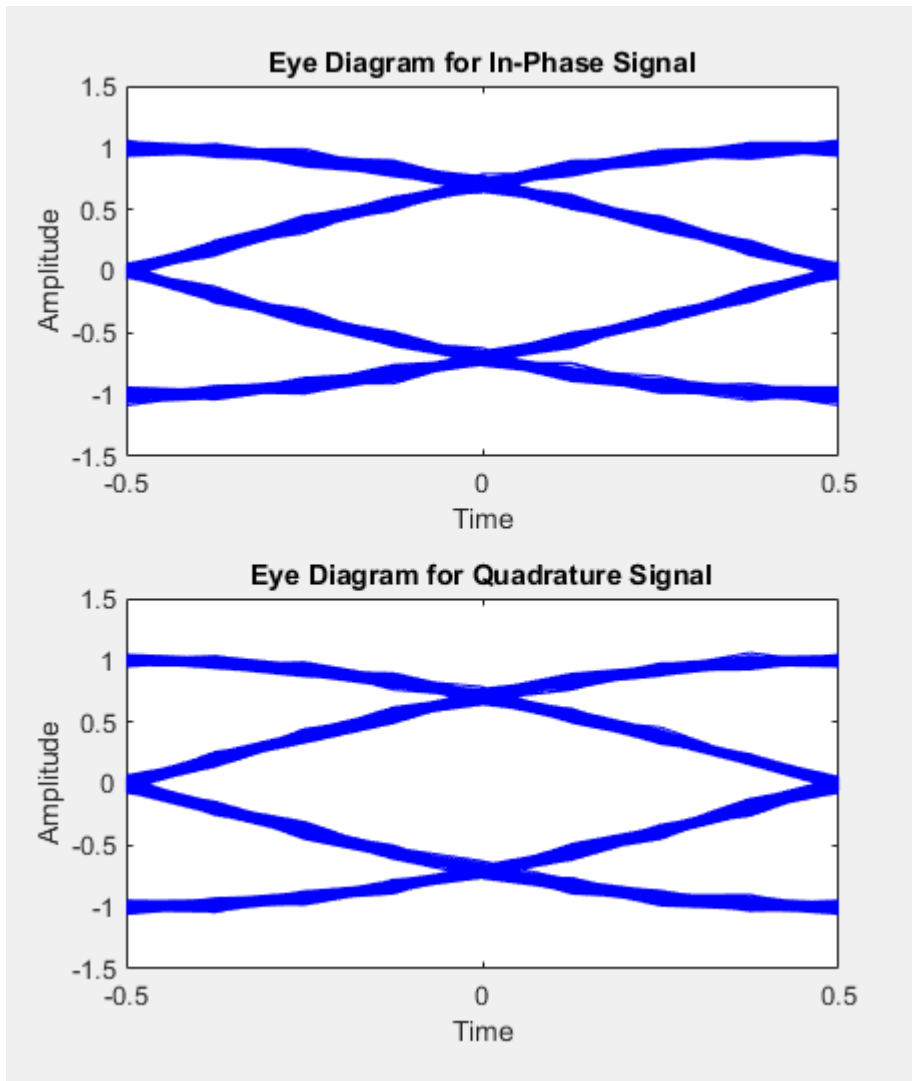
```
rxSigGMSK = awgn(modSigGMSK,30);  
rxSigMSK = awgn(modSigMSK,30);
```

Plot the eye diagrams of the noisy signals. With the GMSK pulse length set to 1, the eye diagrams are nearly identical.

```
eyediagram(rxSigGMSK,sps,1,sps/2)
```



```
eyediagram(rxSigMSK,sps,1,sps/2)
```



Set the `PulseLength` property of `hgmsk` to 3. Because the property is nontunable, the object must be released first.

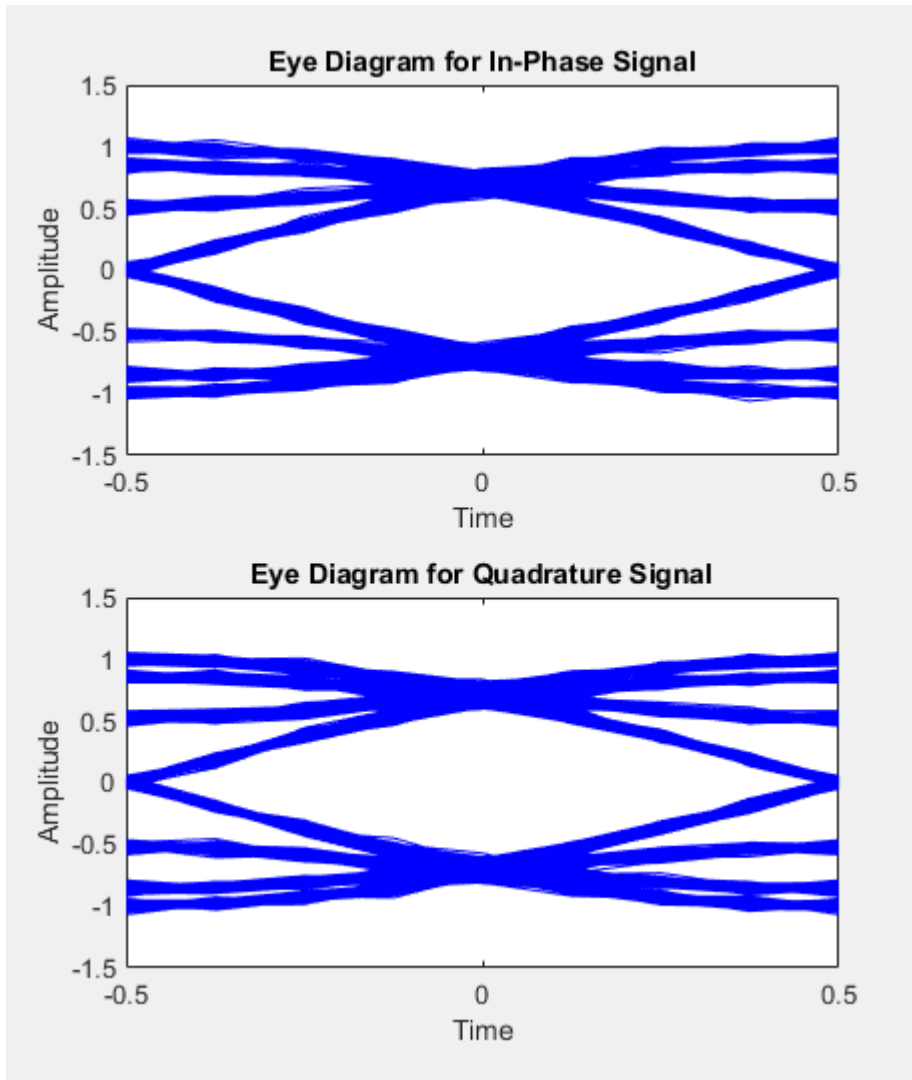
```
release(gmskMod)
gmskMod.PulseLength = 3;
```

Generate a modulated signal using the updated GMSK modulator object and pass it through the AWGN channel.

```
modSigGMSK = gmskMod(data);  
rxSigGMSK = awgn(modSigGMSK,30);
```

Plot the eye diagram of the GMSK signal. The increased pulse length results in an increase in the number of paths showing that the CPM waveform depends on values of the previous symbols as well as the present symbol.

```
eyediagram(rxSigGMSK,sps,1,sps/2)
```



Experiment by changing the `PulseLength` parameter of `hGMSK` to other values. If you set the property to an even number, you should set `hGMSK.InitialPhaseOffset` to $\pi/4$ and the offset argument of the `eyediagram` function from `sps/2` to 0 for a better view of the modulated signal. In order to more clearly view the Gaussian pulse shape, you must

use scopes that display the phase of the signal, as described in the “CPM Phase Tree” on page 9-23 example.

Gray Coded 8-PSK

In this section...

“Structure of the Example” on page 9-42

“Gray-Coded M-PSK Modulation” on page 9-43

“Exploring the Example” on page 9-46

“Simulation Results” on page 9-47

“Comparison with Pure Binary Coding and Theory” on page 9-48

This model, `doc_gray_code`, shows a communications link using Gray-coded 8-PSK modulation. Gray coding is a technique often used in multilevel modulation schemes to minimize the bit error rate by ordering modulation symbols so that the binary representations of adjacent symbols differ by only one bit.

Structure of the Example

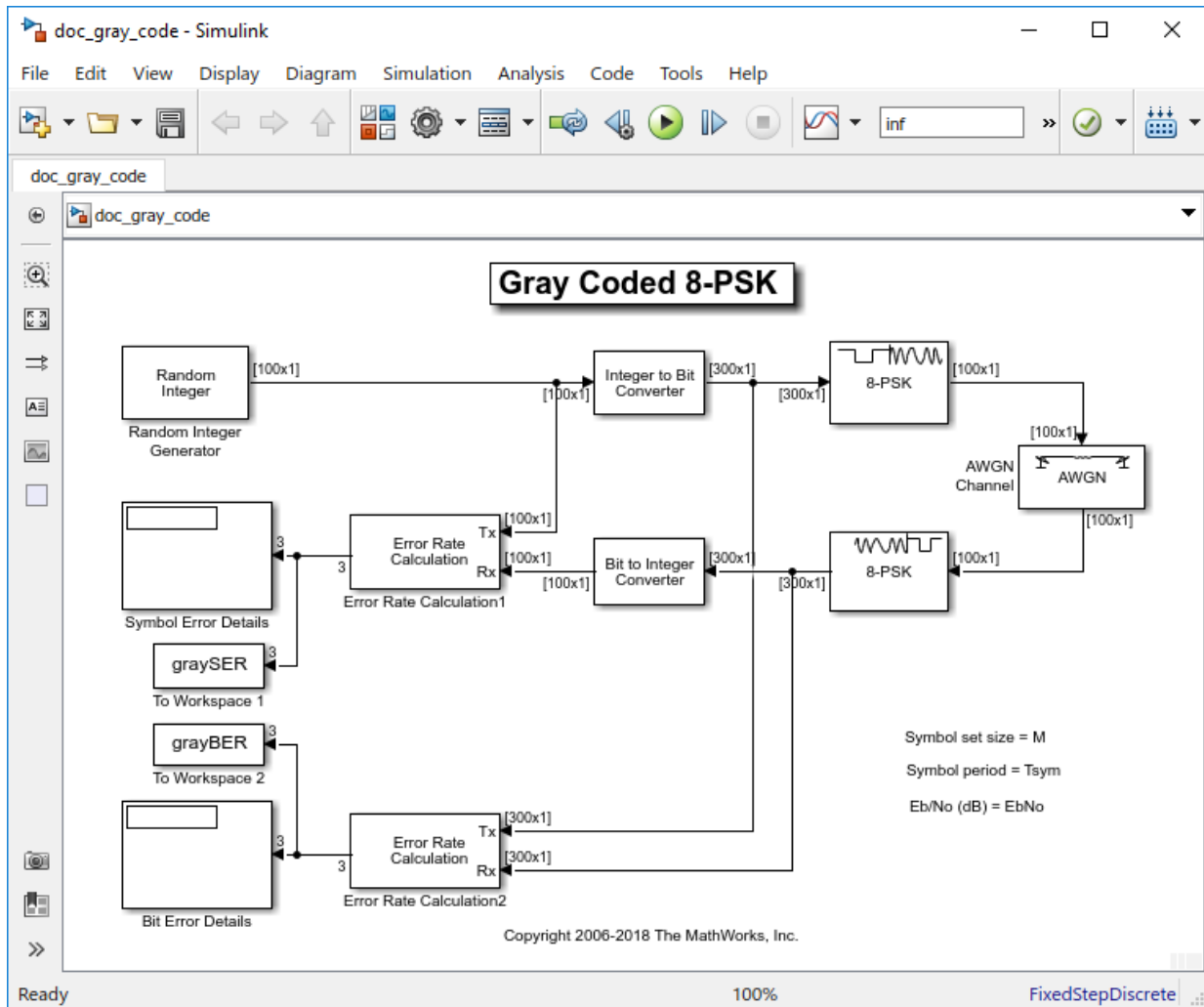
The example model includes these blocks:

- The Random Integer Generator block serves as the source, producing a sequence of integers.
- The Integer to Bit Converter block converts each integer into a corresponding binary representation.
- The AWGN Channel block adds white Gaussian noise to the modulated data.
- The M-PSK Demodulator Baseband block demodulates the corrupted data.
- The Bit to Integer Converter block converts each binary representation to a corresponding integer.
- One copy of the Error Rate Calculation block (labeled `Error Rate Calculation1` in this model) compares the demodulated integer data with the original source data, yielding symbol error statistics. The output of the Error Rate Calculation block is a three-element vector containing the calculated error rate, the number of errors observed, and the amount of data processed.
- Another copy of the Error Rate Calculation library block (labeled `Error Rate Calculation2` in this model) compares the demodulated binary data with the binary representations of the source data, yielding bit error statistics.

Gray-Coded M-PSK Modulation

In this model, the M-PSK Modulator Baseband block:

- Accepts binary-valued inputs that represent integers between 0 and $M - 1$, where M is the alphabet size
- Maps binary representations to constellation points using a Gray-coded ordering
- Produces unit-magnitude complex phasor outputs, with evenly spaced phases between 0 and $2\pi(M - 1)/M$



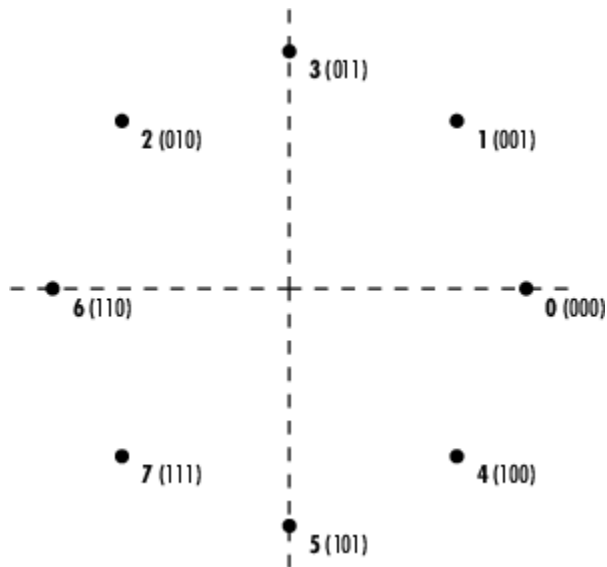
The table indicates which binary representations in the input correspond to which phasors in the output. The second column of the table is an intermediate representation that the block uses in its computations.

Modulator Input	Gray-Coded Ordering	Modulator Output
000	0	$\exp(0) = 1$

Modulator Input	Gray-Coded Ordering	Modulator Output
001	1	$\exp(j\pi/4)$
010	3	$\exp(j3\pi/4)$
011	2	$\exp(j2\pi/4) = \exp(j\pi/2)$
100	7	$\exp(j7\pi/4)$
101	6	$\exp(j6\pi/4) = \exp(j3\pi/2)$
110	4	$\exp(j4\pi/4) = \exp(j\pi)$
111	5	$\exp(j5\pi/4)$

The table below sorts the first two columns of the table above, according to the output values. This sorting makes it clearer that the overall effect of this subsystem is a Gray code mapping, as shown in the figure below. Notice that the numbers in the second column of the table below appear in counterclockwise order in the figure.

Modulator Output	Modulator Input
$\exp(0)$	000
$\exp(j\pi/4)$	001
$\exp(j2\pi/4) = \exp(j\pi/2)$	011
$\exp(j3\pi/4)$	010
$\exp(j4\pi/4) = \exp(j\pi)$	110
$\exp(j5\pi/4)$	111
$\exp(j6\pi/4) = \exp(j3\pi/2)$	101
$\exp(j7\pi/4)$	100



Exploring the Example

You can analyze the data that the example produces to compare theoretical performance with simulation performance.

The theoretical symbol error probability of MPSK is

$$P_E(M) = \text{erfc} \left(\sqrt{\frac{E_s}{N_0}} \sin\left(\frac{\pi}{M}\right) \right)$$

where erfc is the complementary error function, E_s/N_0 is the ratio of energy in a symbol to noise power spectral density, and M is the number of symbols.

To determine the bit error probability, the symbol error probability, P_E , needs to be converted to its bit error equivalent. There is no general formula for the symbol to bit error conversion. Upper and lower limits are nevertheless easy to establish. The actual bit error probability, P_b , can be shown to be bounded by

$$\frac{P_E(M)}{\log_2 M} \leq P_b \leq \frac{M/2}{M-1} P_E(M)$$

The lower limit corresponds to the case where the symbols have undergone Gray coding. The upper limit corresponds to the case of pure binary coding.

Simulation Results

To test the Gray code modulation scheme in this model, simulate the graycode model for a range of E_b/N_0 values. If you want to study bit error rates but not symbol error rates, then you can use the `bertool` graphical user interface as described in “BERTool” on page 15-37.

The rest of this section studies both the bit and symbol error rates and hence does not use `bertool`.

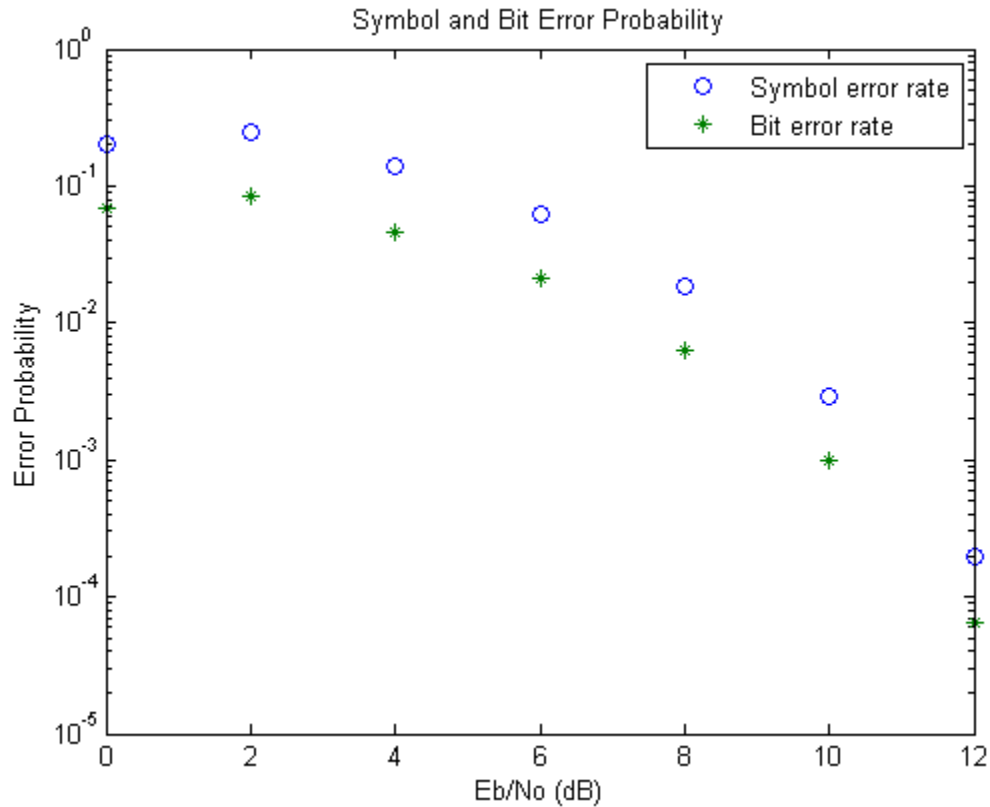
Because increasing the value of E_b/N_0 lowers the number of errors produced, the length of each simulation must be increased to ensure that the statistics of the errors remain stable.

Using the `sim` command to run a Simulink simulation from the MATLAB command window, the following code generates data for symbol error rate and bit error rate curves. It considers E_b/N_0 values in the range 0 dB to 12 dB, in steps of 2 dB.

```
M      = 8;
Tsym   = 0.2;
BERVec = [];
SERVec = [];
EbNoVec = [0:2:12];
for n   = 1:length(EbNoVec);
    EbNo = EbNoVec(n);
    sim('doc_gray_code') ;
    SERVec(n,:) = graySER;
    BERVec(n,:) = grayBER;
end;
```

After simulating for the full set of E_b/N_0 values, you can plot the results using these commands:

```
semilogy( EbNoVec, SERVec(:,1), 'o', EbNoVec, BERVec(:,1), '*' );
legend ( 'Symbol error rate', 'Bit error rate' );
xlabel ( 'Eb/No (dB)' ); ylabel( 'Error Probability' );
title ( 'Symbol and Bit Error Probability' );
```



Comparison with Pure Binary Coding and Theory

As a further exercise, using data obtained from `berawgn`, you can plot the theoretical curves on the same axes with the simulation results. You can also compare Gray coding with pure binary coding, by modifying the M-PSK Modulator Baseband and M-PSK Demodulator Baseband blocks so that their **Constellation ordering** parameters are Binary instead of Gray.

Soft Decision GMSK Demodulator

In this section...

“Structure of the Example” on page 9-49

“The Serial GMSK Receiver” on page 9-50

“Results and Displays” on page 9-51

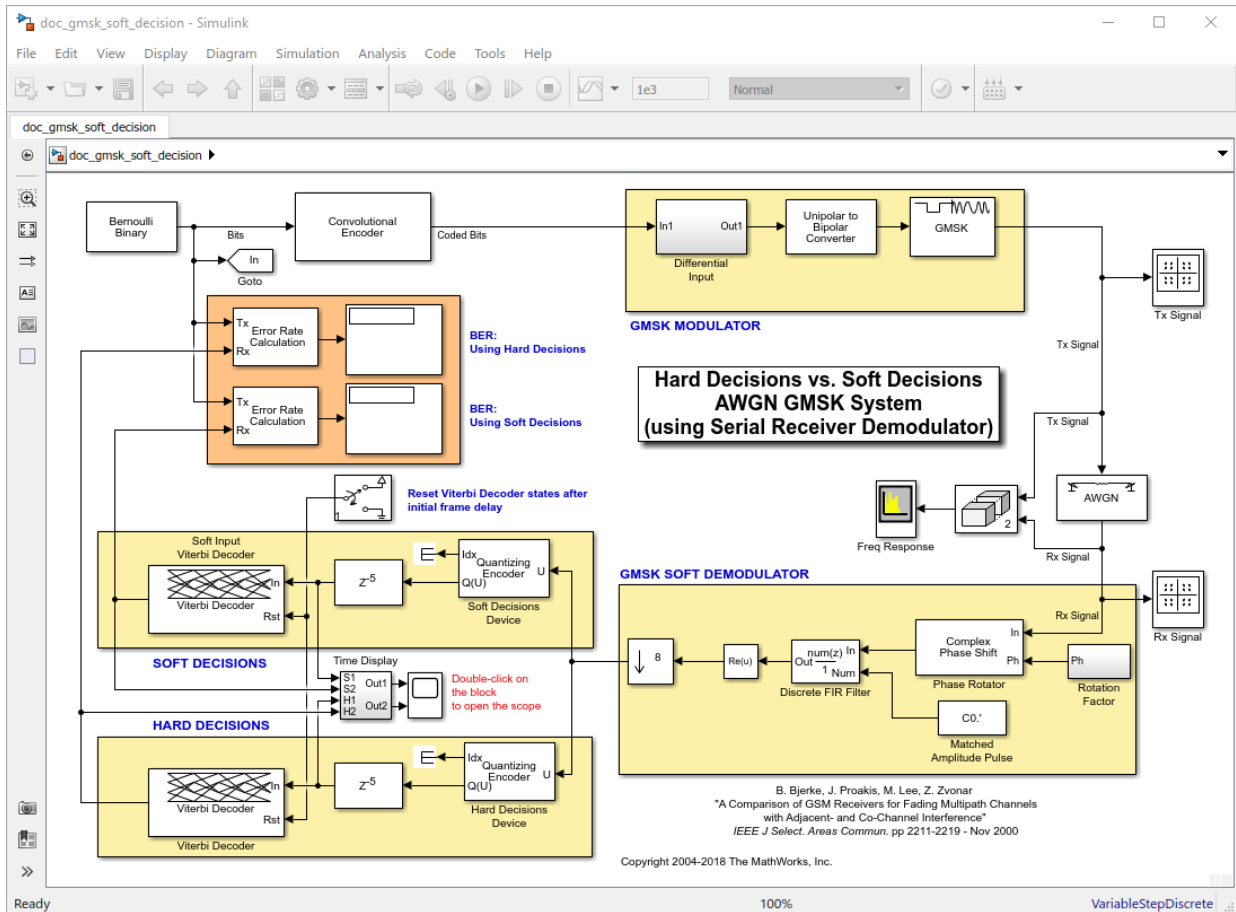
This model shows a system that includes convolutional coding and GMSK modulation. The receiver in this model includes two parallel paths, one that uses soft decisions and another that uses hard decisions. The model uses the bit error rates for the two paths to illustrate that the soft decision receiver performs better. This is to be expected, because soft decisions enable the system to retain more information from the demodulation operation to use in the decoding operation.

Structure of the Example

The example model, `doc_gmsk_soft_decision`, transmits and receives a coded GMSK signal.

The key components are:

- A Bernoulli Binary Generator block, which generates binary numbers.
- A Convolutional Encoder block, which encodes the binary numbers using a rate 1/2 convolutional code.
- A GMSK modulator section, which computes the logical difference between successive bits and modulates the result using the GMSK Modulator Baseband block.
- A **GMSK soft demodulator** section that implements the detector design proposed in [1], called a serial receiver. This section of the model produces a noisy bipolar signal. The section labeled **Soft Decisions** uses an eight-region partition in the Quantizing Encoder block to prepare for 3-bit soft-decision decoding using the Viterbi Decoder block. The section labeled **Hard Decisions** uses a two-region partition to prepare for hard-decision Viterbi decoding. Using a two-region partition here is equivalent to having the demodulator make hard decisions. In each decoding section, a Delay block aligns codeword boundaries with frame boundaries so that the Viterbi Decoder block can decode properly. This is necessary because the combined delay of other blocks in the system is not an integer multiple of the length of a codeword.
- A pair of Error Rate Calculation blocks, as well as Display blocks that show the BER for the system with each type of decision.



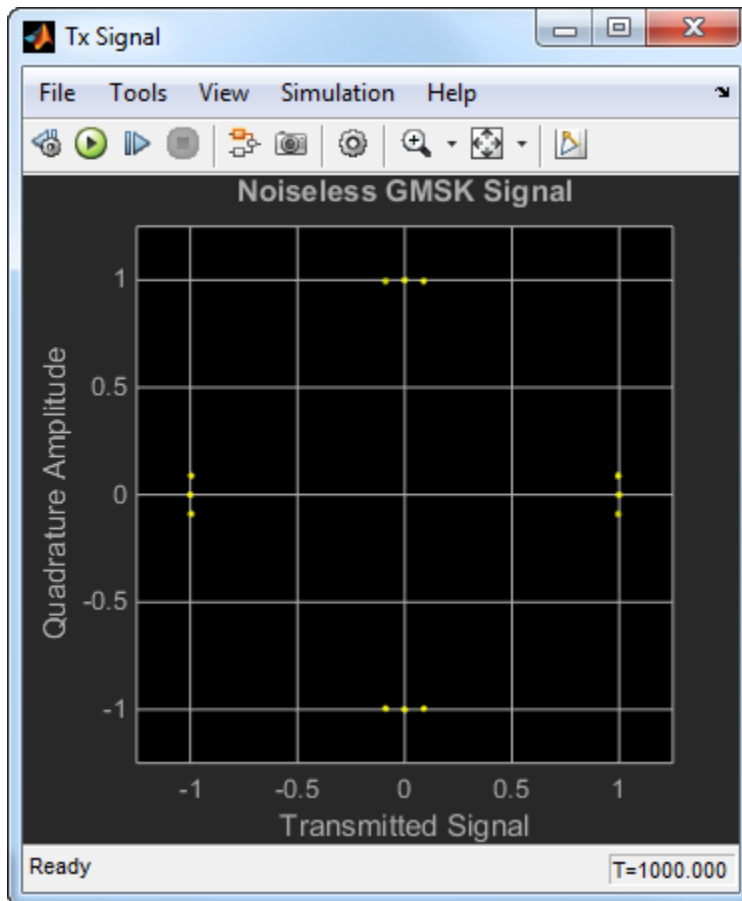
The Serial GMSK Receiver

The serial GMSK receiver is based on the fact that GMSK can be represented as a combination of amplitude pulses [2] - [3], and can, therefore, be demodulated with a matched filter. The GMSK waveform used in this model has a BT product of 0.3 and a frequency pulse length of 4 symbols. As such, it can be represented by eight different amplitude pulses, which are shown in Figure 2 of [3]. The matched filter in this model uses only the largest pulse of the eight, because of its simplicity of implementation. That same simplicity, however, yields BER performance that is inferior to the more traditional Viterbi-based demodulator.

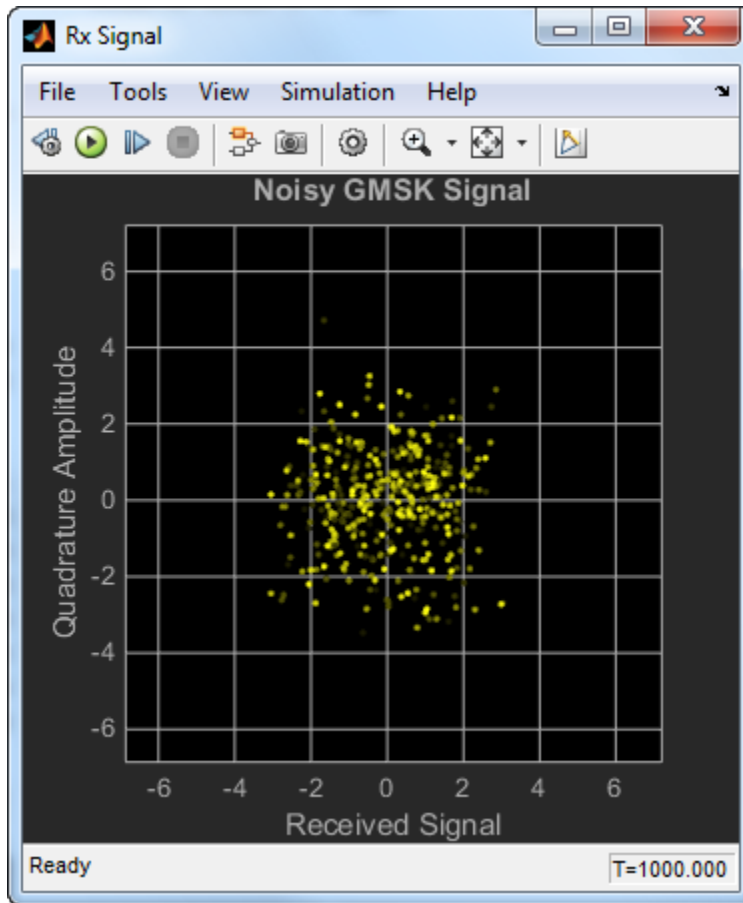
Results and Displays

The example model includes these visualizations to illustrate its performance:

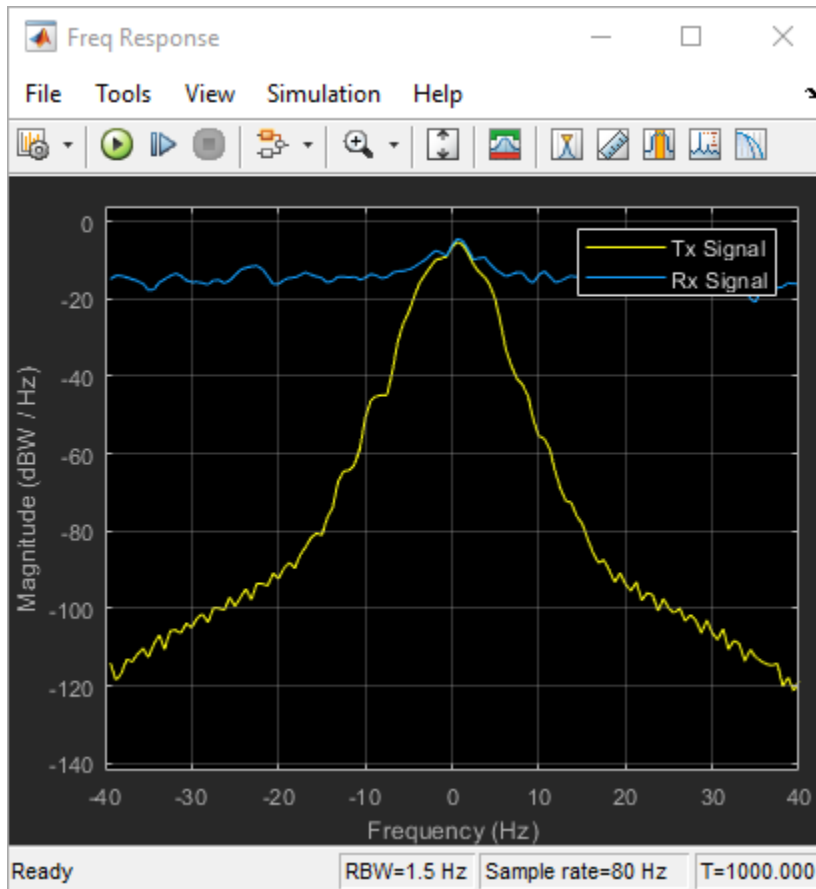
- The Display blocks illustrate that the soft decision receiver performs better (that is, has a smaller BER) than the hard decision receiver.
- The Tx Signal window shows the scatter plot of the signal before the AWGN channel.



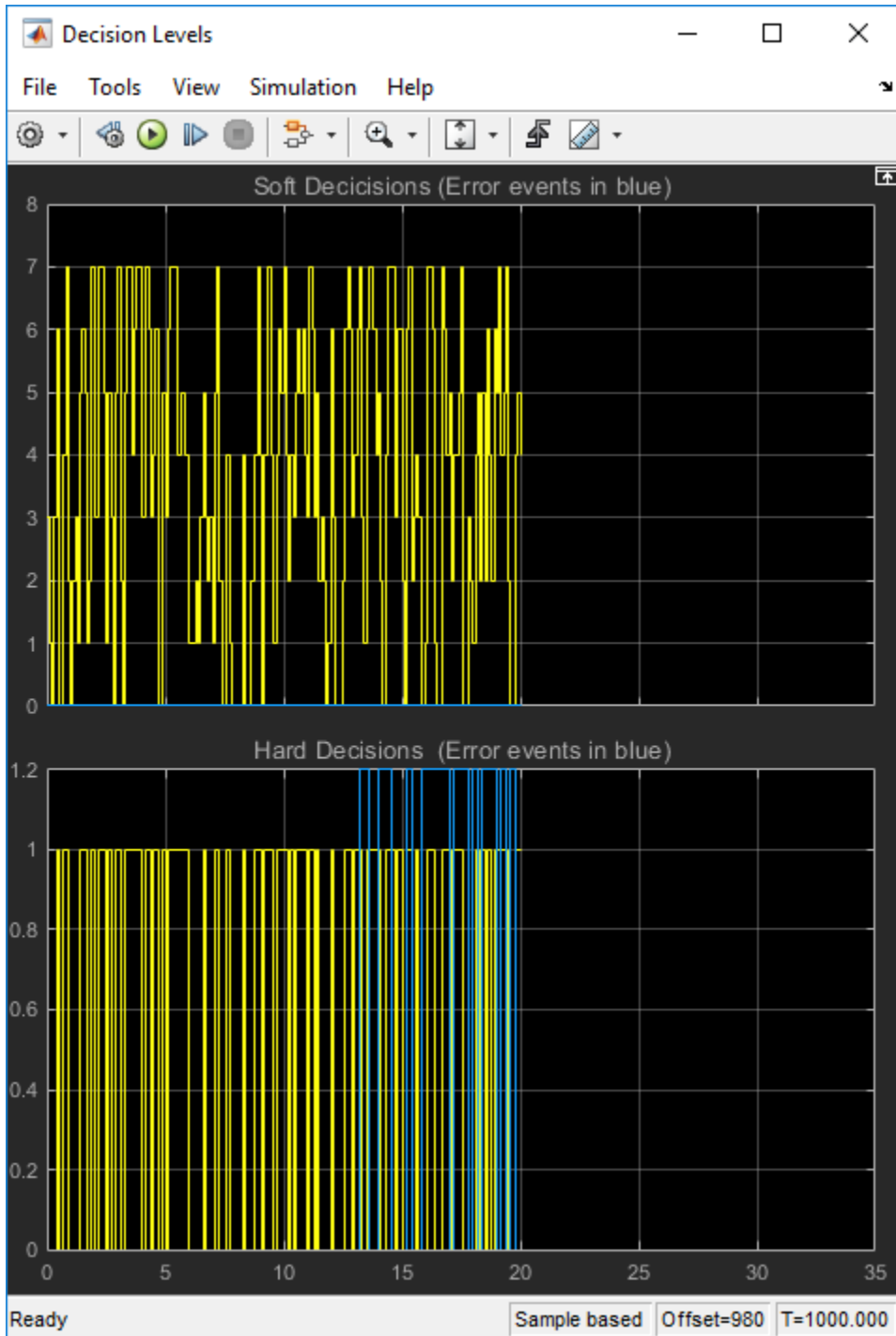
- The Rx Signal window shows the scatter plot of the signal after the AWGN channel.



- The Freq Response window shows the frequency response of the GSMK signal before and after the AWGN channel.



- The Decision Levels window shows, in yellow, the various soft decision levels in the top plot and the binary hard decisions in the bottom plot. This window also indicates, in blue, when errors occur.



References

- [1] Bjerke, B., J. Proakis, M. Lee, and Z. Zvonar, "A Comparison of GSM Receivers for Fading Multipath Channels with Adjacent- and Co-Channel Interference," IEEE J. Select. Areas Commun., Nov. 2000, pp. 2211-2219.
- [2] Laurent, Pierre, "Exact and Approximate Construction of Digital Phase Modulations by Superposition of Amplitude Modulated Pulses (AMP)," IEEE Trans. Comm., Vol. COM-34, No. 2, Feb. 1986, pp. 150-160.
- [3] Jung, Peter, "Laurent's Representation of Binary Digital Continuous Phase Modulated Signals with Modulation index 1/2 Revisited", IEEE Trans. Comm., Vol. COM-42, No. 2/3/4, Feb./Mar./Apr. 1994, pp. 221-224.

16-PSK with Custom Symbol Mapping

Create 16-PSK modulator and demodulator System objects™ in which custom symbol mapping is used. Estimate the BER in an AWGN channel and compare the performance with that of a theoretical Gray-coded PSK system.

Create a custom symbol mapping for the 16-PSK modulation scheme. The 16 integer symbols must have values which fall between 0 and 15.

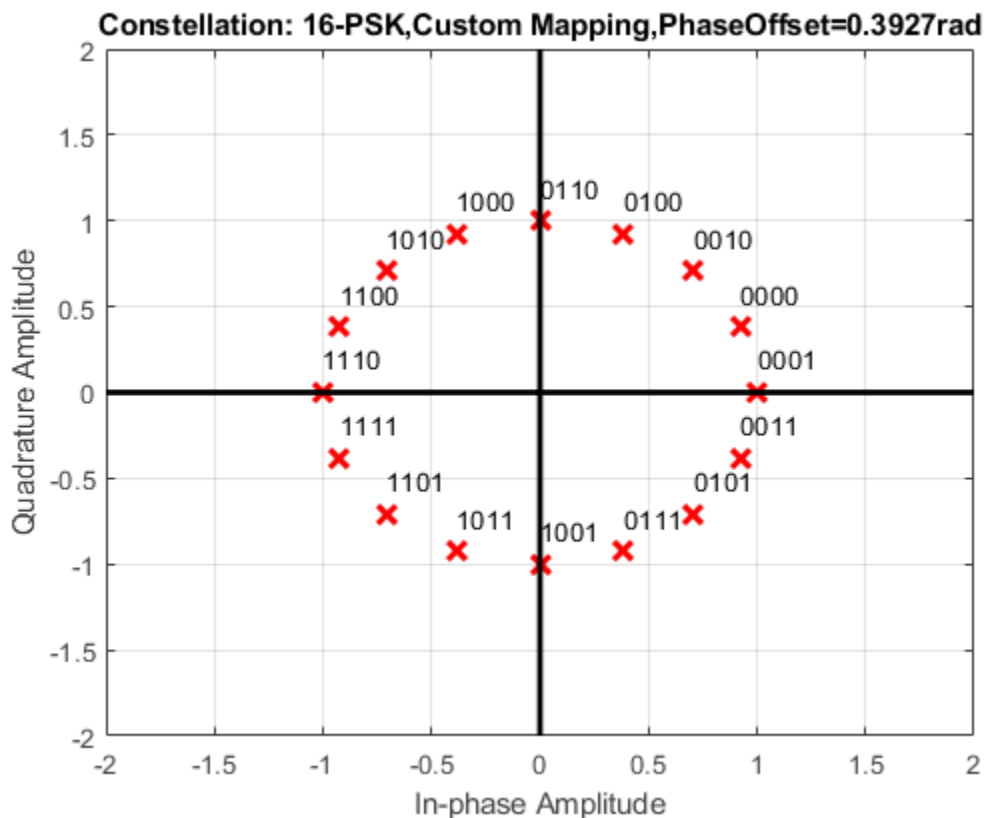
```
custMap = [0 2 4 6 8 10 12 14 15 13 11 9 7 5 3 1];
```

Create a 16-PSK modulator and demodulator pair having custom symbol mapping defined by the array, `custMap`.

```
pskModulator = comm.PSKModulator(16, 'BitInput', true, ...  
    'SymbolMapping', 'Custom', ...  
    'CustomSymbolMapping', custMap);  
pskDemodulator = comm.PSKDemodulator(16, 'BitOutput', true, ...  
    'SymbolMapping', 'Custom', ...  
    'CustomSymbolMapping', custMap);
```

Display the modulator constellation.

```
constellation(pskModulator)
```

Create an AWGN channel System object for use with 16-ary data.

```
awgnChannel = comm.AWGNChannel('BitsPerSymbol', log2(16));
```

Create an error rate object to track the BER statistics.

```
errorRate = comm.ErrorRate;
```

Initialize the simulation vectors. The Eb/No is varied from 6 to 18 dB in 1 dB steps.

```
ebnoVec = 6:18;
ber = zeros(size(ebnoVec));
```

Estimate the BER by modulating binary data, passing it through an AWGN channel, demodulating the received signal, and collecting the error statistics.

```
for k = 1:length(ebnoVec)

    % Reset the error counter for each Eb/No value
    reset(errorRate)
    % Reset the array used to collect the error statistics
    errVec = [0 0 0];
    % Set the channel Eb/No
    awgnChannel.EbNo = ebnoVec(k);

    while errVec(2) < 200 && errVec(3) < 1e7
        % Generate a 1000-symbol frame
        data = randi([0 1],4000,1);
        % Modulate the binary data
        modData = pskModulator(data);
        % Pass the modulated data through the AWGN channel
        rxSig = awgnChannel(modData);
        % Demodulate the received signal
        rxData = pskDemodulator(rxSig);
        % Collect the error statistics
        errVec = errorRate(data,rxData);
    end

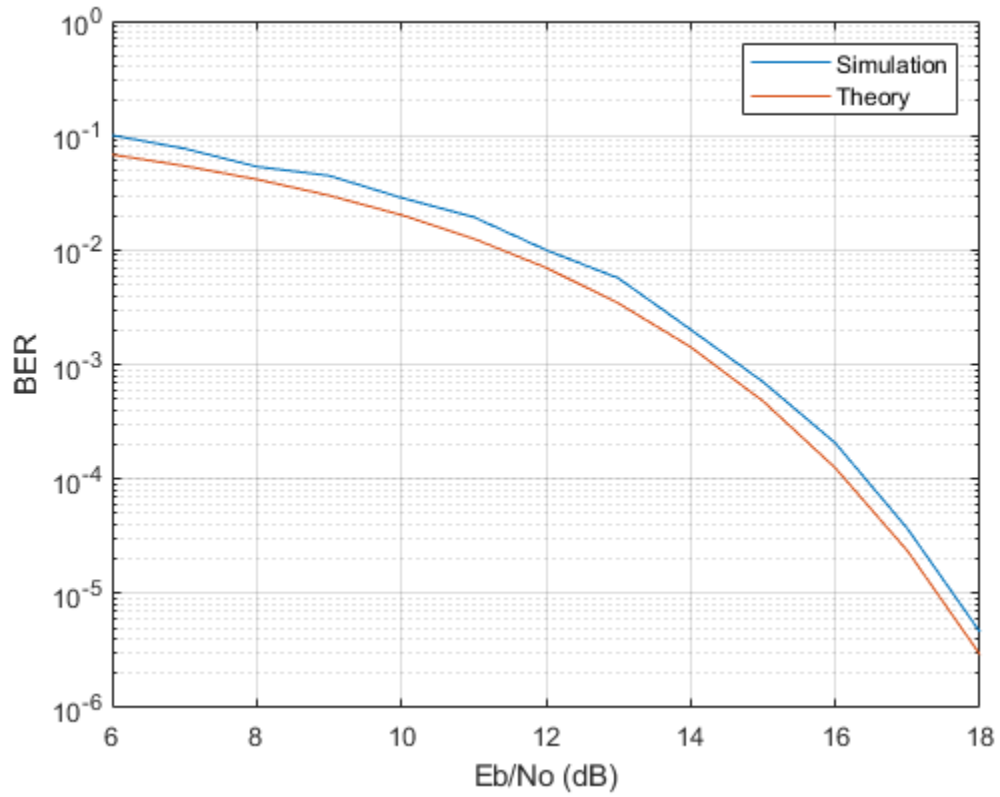
    % Save the BER data
    ber(k) = errVec(1);
end
```

Generate theoretical BER data for an AWGN channel using `berawgn`.

```
berTheory = berawgn(ebnoVec, 'psk', 16, 'nondiff');
```

Plot the simulated and theoretical results. Because the simulated results rely on 16-PSK modulation that does not use Gray codes, the performance is not as good as that predicted by theory.

```
figure
semilogy(ebnoVec, [ber; berTheory])
xlabel('Eb/No (dB)')
ylabel('BER')
grid
legend('Simulation', 'Theory', 'location', 'ne')
```



General QAM Modulation in AWGN Channel

Transmit and receive data using a nonrectangular 16-ary constellation in the presence of Gaussian noise. Show the scatter plot of the noisy constellation and estimate the symbol error rate (SER) for two different signal-to-noise ratios.

Create a 16-QAM constellation based on the V.29 standard for telephone-line modems.

```
c = [-5 -5i 5 5i -3 -3-3i -3i 3-3i 3 3+3i 3i -3+3i -1 -1i 1 1i];  
M = length(c);
```

Generate random symbols.

```
data = randi([0 M-1],2000,1);
```

Modulate the data by using the `genqammod` function. General QAM modulation is necessary because the custom constellation is not rectangular.

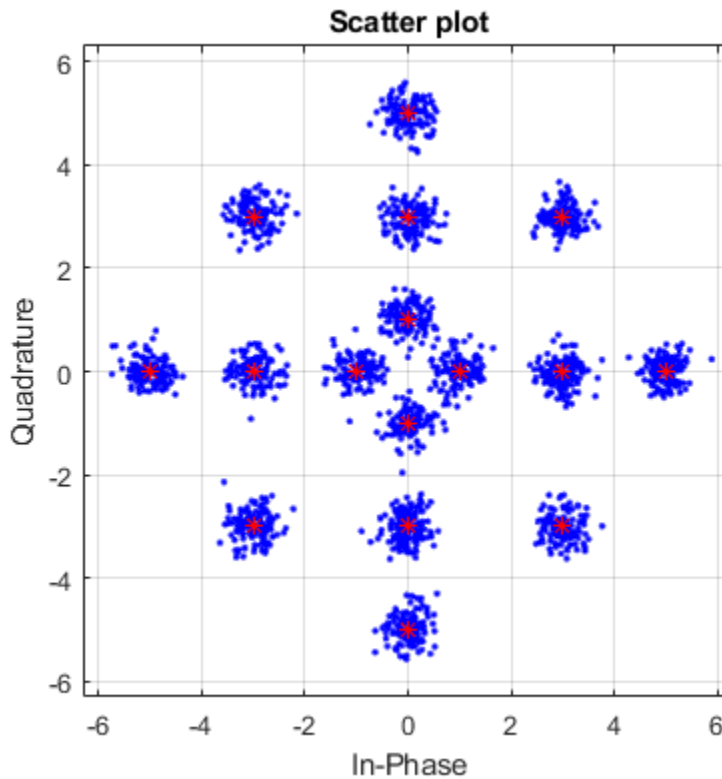
```
modData = genqammod(data,c);
```

Pass the signal through an AWGN channel having a 20 dB signal-to-noise ratio (SNR).

```
rxSig = awgn(modData,20,'measured');
```

Display a scatter plot of the received signal and the reference constellation, `c`.

```
h = scatterplot(rxSig);  
hold on  
scatterplot(c,[],[],'r*',h)  
grid  
hold off
```



Demodulate the received signal by using the `genqamdemod` function. Determine the number of symbol errors and the symbol error ratio.

```
demodData = genqamdemod(rxSig,c);
[numErrors,ser] = symerr(data,demodData)
```

```
numErrors = 1
```

```
ser = 5.0000e-04
```

Repeat the transmission and demodulation process with an AWGN channel having a 10 dB SNR. Determine the symbol error rate for the reduced SNR. As expected, the performance degrades when the SNR is decreased.

```
rxSig = awgn(modData,10,'measured');  
demodData = genqamdemod(rxSig,c);  
[numErrors,ser] = symerr(data,demodData)  
  
numErrors = 462  
ser = 0.2310
```

FM Modulate and Demodulate a Sinusoidal Signal

Modulate and demodulate a sinusoidal signal. Plot the demodulated signal and compare it to the original signal.

Set the example parameters.

```
fs = 100;           % Sample rate (Hz)
ts = 1/fs;         % Sample period (s)
fd = 25;           % Frequency deviation (Hz)
```

Create a sinusoidal input signal with duration 0.5s and frequency 4 Hz.

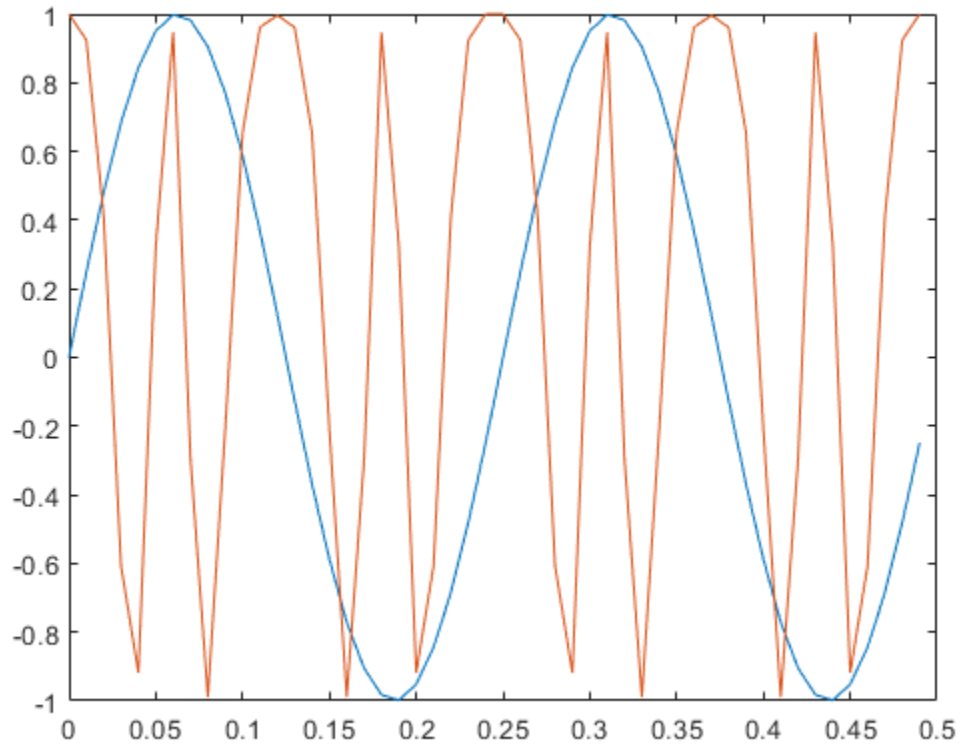
```
t = (0:ts:0.5-ts)';
x = sin(2*pi*4*t);
```

Create an FM modulator System object™.

```
MOD = comm.FMModulator('SampleRate',fs,'FrequencyDeviation',fd);
```

FM modulate the input signal and plot its real part. You can see that the frequency of the modulated signal changes with the amplitude of the input signal.

```
y = step(MOD,x);
plot(t,[x real(y)])
```

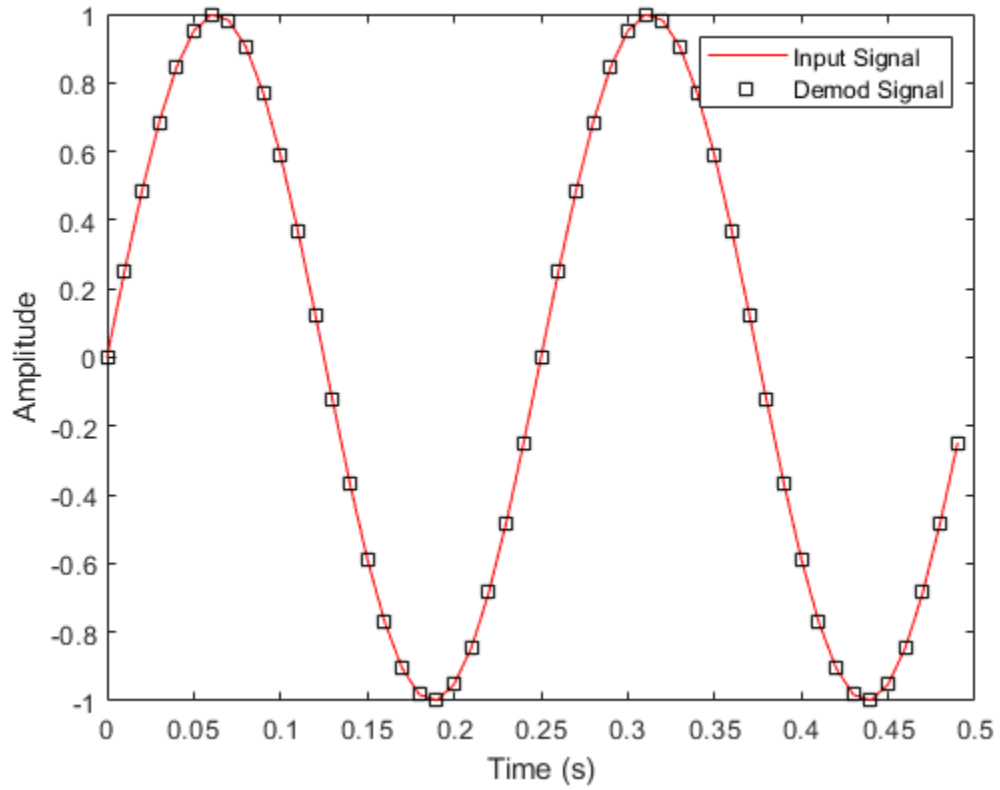


Demodulate the FM modulated signal.

```
DEM0D = comm.FMDemodulator('SampleRate', fs, 'FrequencyDeviation', fd);  
z = step(DEM0D, y);
```

Plot the input and demodulated signals. The demodulator output signal exactly aligns with the input signal.

```
plot(t, x, 'r', t, z, 'ks')  
legend('Input Signal', 'Demod Signal')  
xlabel('Time (s)')  
ylabel('Amplitude')
```

MSK

- “MSK Signal Recovery” on page 10-2
- “MSK Signal Recovery” on page 10-10

MSK Signal Recovery

Model channel impairments such as timing phase offset, carrier frequency offset, and carrier phase offset for a minimum shift keying (MSK) signal. Use `comm.MSKTimingSynchronizer` and `comm.CarrierSynchronizer` System objects to synchronize such signals at the receiver. The MSK timing synchronizer recovers the timing offset, while a carrier synchronizer recovers the carrier frequency and phase offsets.

Initialize system variables by running the MATLAB script `configureMSKSignalRecoveryEx`. Define the logical control variable `recoverTimingPhase` to enable timing phase recovery, and `recoverCarrier` to enable carrier frequency and phase recovery.

```
configureMSKSignalRecoveryEx;  
recoverTimingPhase = true;  
recoverCarrier = true;
```

Modeling Channel Impairments

Specify the sample delay, `timingOffset`, that the channel model applies. Create a variable fractional delay object to introduce the timing delay to the transmitted signal.

```
timingOffset = 0.2;  
varDelay = dsp.VariableFractionalDelay;
```

Create a `comm.PhaseFrequencyOffset` System object to introduce carrier phase and frequency offsets to a modulated signal. Because the MSK modulator upsamples the transmitted symbols, set the `SampleRate` property to the ratio of the `samplesPerSymbol` and the sample time, `Ts`.

```
freqOffset = 50;  
phaseOffset = 30;  
pfo = comm.PhaseFrequencyOffset(...  
    'FrequencyOffset', freqOffset, ...  
    'PhaseOffset', phaseOffset, ...  
    'SampleRate', samplesPerSymbol/Ts);
```

Create a `comm.AWGNChannel` System object to add white Gaussian noise to the modulated signal. The noise power is determined by the `EbNo` property, that is the bit energy to noise power spectral density ratio. Because the MSK modulator generates symbols with 1 Watt of power, set the `signalPower` property of the AWGN channel System object to 1.

```

EbNo = 20 + 10*log10(samplesPerSymbol);
chAWGN = comm.AWGNChannel(...
    'NoiseMethod','Signal to noise ratio (Eb/No)', ...
    'EbNo',EbNo,...
    'SignalPower',1, ...
    'SamplesPerSymbol',samplesPerSymbol);

```

Timing Phase, Carrier Frequency, and Carrier Phase Synchronization

Create an MSK timing synchronizer to recover symbol timing phase using a fourth-order nonlinearity method.

```

timeSync = comm.MSKTimingSynchronizer(...
    'SamplesPerSymbol',samplesPerSymbol, ...
    'ErrorUpdateGain',0.02);

```

Create a carrier synchronizer to recover both carrier frequency and phase. Because the MSK constellation is QPSK with a 0-degree phase offset, set the `comm.CarrierSynchronizer` accordingly.

```

phaseSync = comm.CarrierSynchronizer(...
    'Modulation','QPSK', ...
    'ModulationPhaseOffset','Custom', ...
    'CustomPhaseOffset',0, ...
    'SamplesPerSymbol',1);

```

Stream Processing Loop

The simulation modulates data using MSK modulation. The modulated symbols pass through the channel model, which applies timing delay, carrier frequency and phase shift, and additive white Gaussian noise. The receiver performs timing phase and carrier frequency and phase recovery. Finally, the signal symbols are demodulated and the bit error rate is calculated. The `plotResultsMSKSignalRecoveryEx` script generates scatter plots in this order to show these effects:

- 1 Channel impairments
- 2 Timing synchronization
- 3 Carrier synchronization

At the end of the simulation, the example displays the timing phase, frequency, and phase estimates as a function of simulation time.

```

for p = 1:numFrames
    %-----

```

```

% Generate and modulate data
%-----
txBits = randi([0 1],samplesPerFrame,1);
txSym = modem(txBits);
%-----
% Transmit through channel
%-----
%
% Add timing offset
rxSigTimingOff = varDelay(txSym,timingOffset*samplesPerSymbol);
%
% Add carrier frequency and phase offset
rxSigCFO = pfo(rxSigTimingOff);
%
% Pass the signal through an AWGN channel
rxSig = chAWGN(rxSigCFO);
%
% Save the transmitted signal for plotting
plot_rx = rxSig;
%
%-----
% Timing recovery
%-----
if recoverTimingPhase
    % Recover symbol timing phase using fourth-order nonlinearity
    % method
    [rxSym,timEst] = timeSync(rxSig);
    % Calculate the timing delay estimate for each sample
    timEst = timEst(1)/samplesPerSymbol;
else
    % Do not apply timing recovery and simply downsample the received
    % signal
    rxSym = downsample(rxSig,samplesPerSymbol);
    timEst = 0;
end

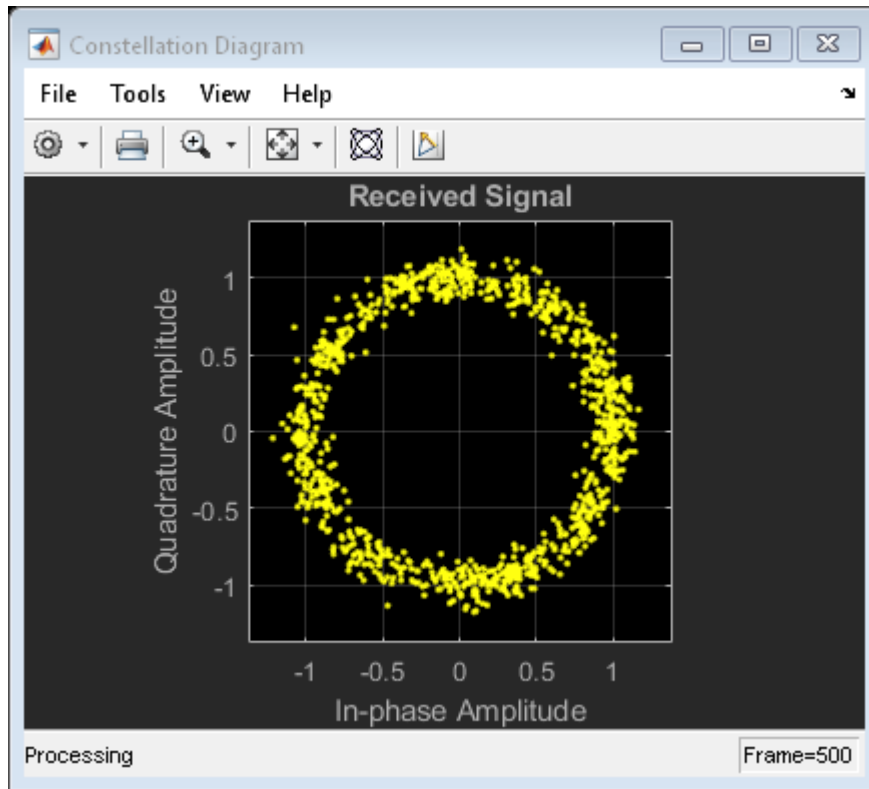
% Save the timing synchronized received signal for plotting
plot_rxTimeSync = rxSym;

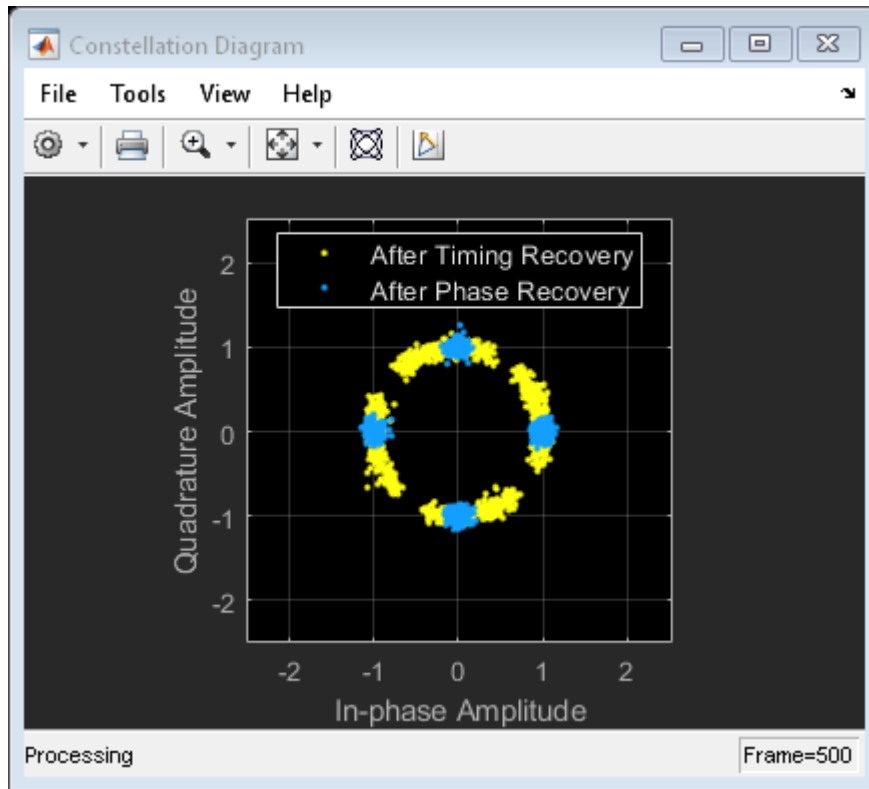
%-----
% Carrier frequency and phase recovery
%-----
if recoverCarrier
    % The following script applies carrier frequency and phase recovery

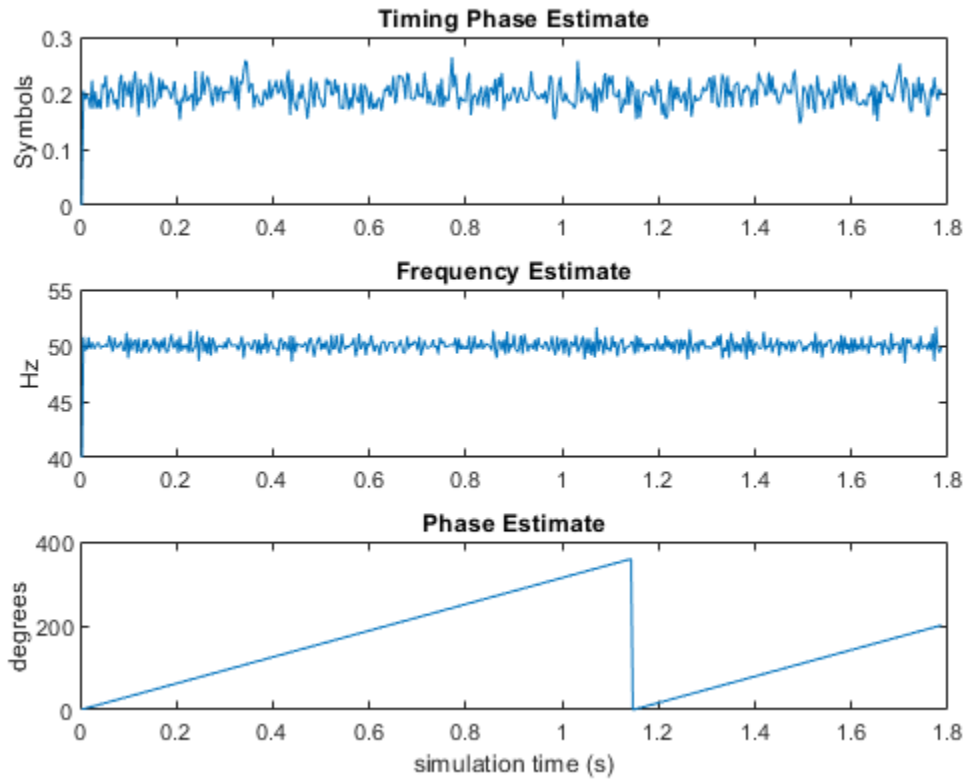
```

```
        % using a second order phase-locked loop (PLL), and removes phase ambiguity
        [rxSym,phEst] = phaseSync(rxSym);
        removePhaseAmbiguityMSKSignalRecoveryEx;
        freqShiftEst = mean(diff(phEst)/(Ts*2*pi));
        phEst = mod(mean(phEst),360); % in degrees
    else
        freqShiftEst = 0;
        phEst = 0;
    end

    % Save the phase synchronized received signal for plotting
    plot_rxPhSync = rxSym;
    %-----
    % Demodulate the received symbols
    %-----
    rxBits = demod(rxSym);
    %-----
    % Calculate the bit error rate
    %-----
    errorStats = BERCalc(txBits,rxBits);
    %-----
    % Plot results
    %-----
    plotResultsMSKSignalRecoveryEx;
end
```







Display the bit error rate and the total number of symbols processed by the error rate calculator.

```
BitErrorRate = errorStats(1)  
TotalNumberOfSymbols = errorStats(3)
```

```
BitErrorRate =  
    4.0001e-06
```

```
TotalNumberOfSymbols =
```

499982

Conclusion and Further Experimentation

The recovery algorithms are demonstrated by using constellation plots taken after timing, carrier frequency, and carrier phase synchronization.

Open the script to create a writable copy of this example and its supporting files. Then, to show the effects of the recovery algorithms, you can enable and disable the logical control variables `recoverTimingPhase` and `recoverCarrier` and rerun the simulation.

Appendix

This example uses these scripts:

- `configureMSKSignalRecoveryEx`
- `plotResultsMSKSignalRecoveryEx`
- `removePhaseAmbiguityMSKSignalRecoveryEx`

MSK Signal Recovery

In this section...
“Exploring the Model” on page 10-10
“Results and Displays” on page 10-11
“Experimenting with the Example” on page 10-14

This model shows how channel impairments such as timing phase offset, carrier frequency offset, and phase offset for a minimum shift keying (MSK) signal are modeled. The model uses blocks from the Synchronization library to recover the signal. Open the model, `doc_commmsksync`.

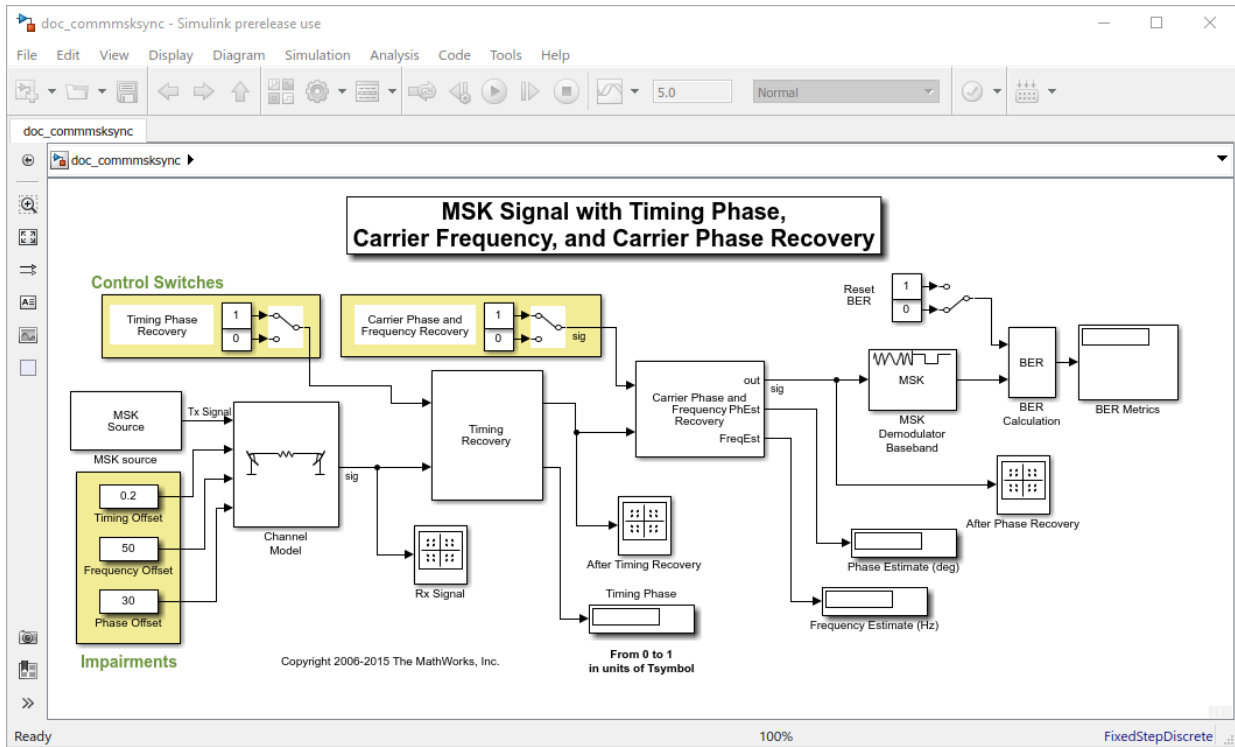
`doc_commmsksync`

Exploring the Model

The example models an MSK transmitted signal undergoing channel impairments, including these components:

- 1 An MSK signal source that uses the Bernoulli Binary Generator block to output equiprobable symbols and modulates the symbols using an MSK Modulator Baseband block
- 2 A channel model that incorporates independently variable offsets in the timing phase, frequency, and phase. The channel model also includes the AWGN Channel block
- 3 Signal recovery, consisting of:
 - Timing recovery using the MSK-Type Signal Timing Recovery block
 - Carrier frequency and phase recovery using the Carrier Synchronizer block
- 4 An MSK Demodulator Baseband block
- 5 Blocks that compute and display the system's bit error rate (BER)

When you load the model, it also initializes some parameters that several blocks share.

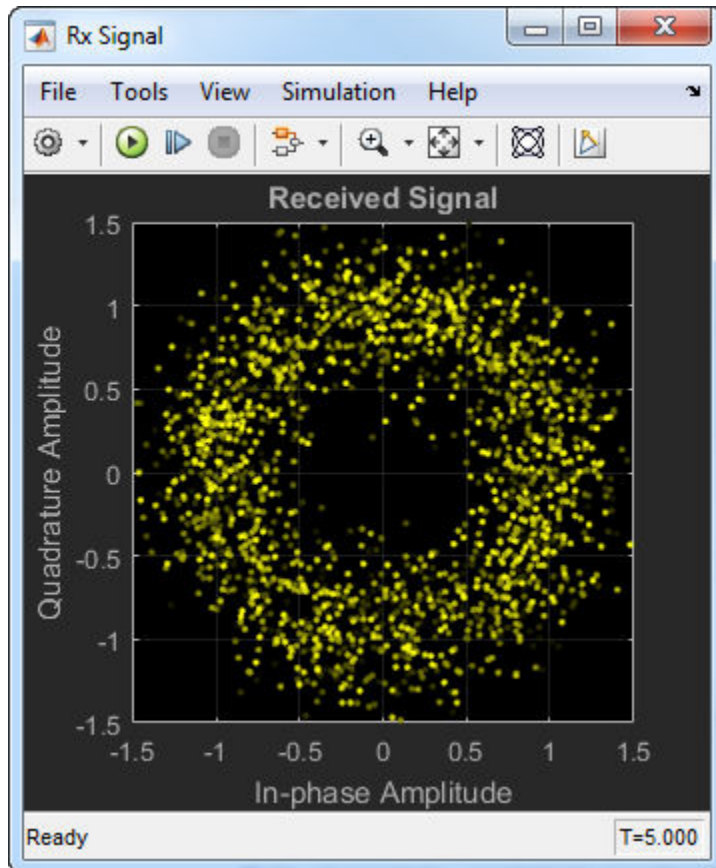


Results and Displays

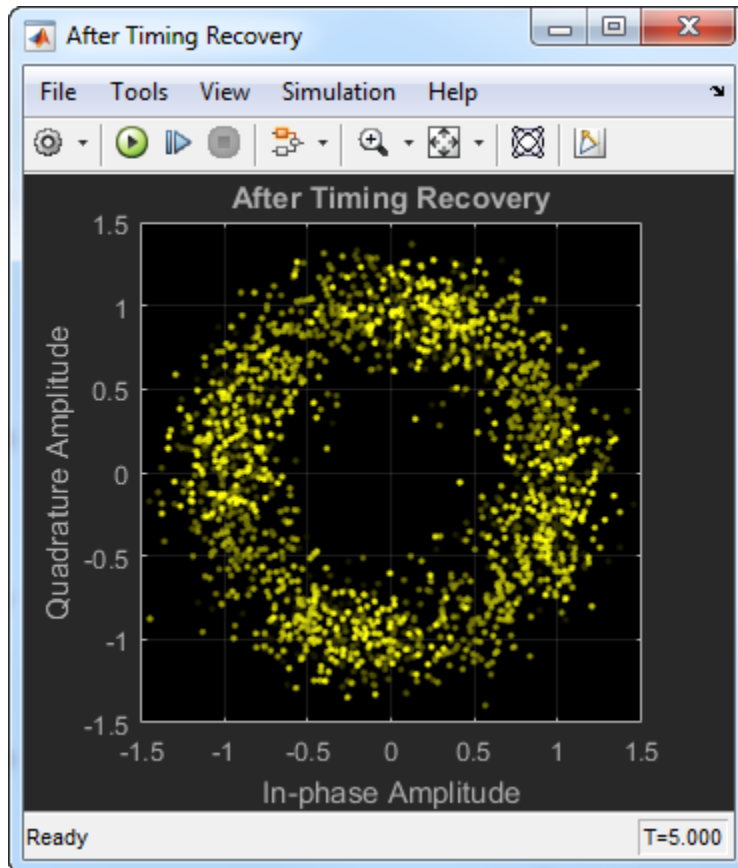
When you run the simulation, the displays show the estimated values for the impairments as well as the BER metrics. Because the Carrier Synchronizer block performs both frequency and phase correction, the display of estimated phase offset may fluctuate rapidly. The display labeled BER Metrics shows a three-element vector containing the calculated bit error rate (BER), the number of errors observed, and the number of bits processed.

You can view the MSK signal via the Constellation Diagram blocks at the different stages. This provides a compelling visual rendition of the recovery algorithms in action, especially as you turn the algorithms on and off using the two control switches.

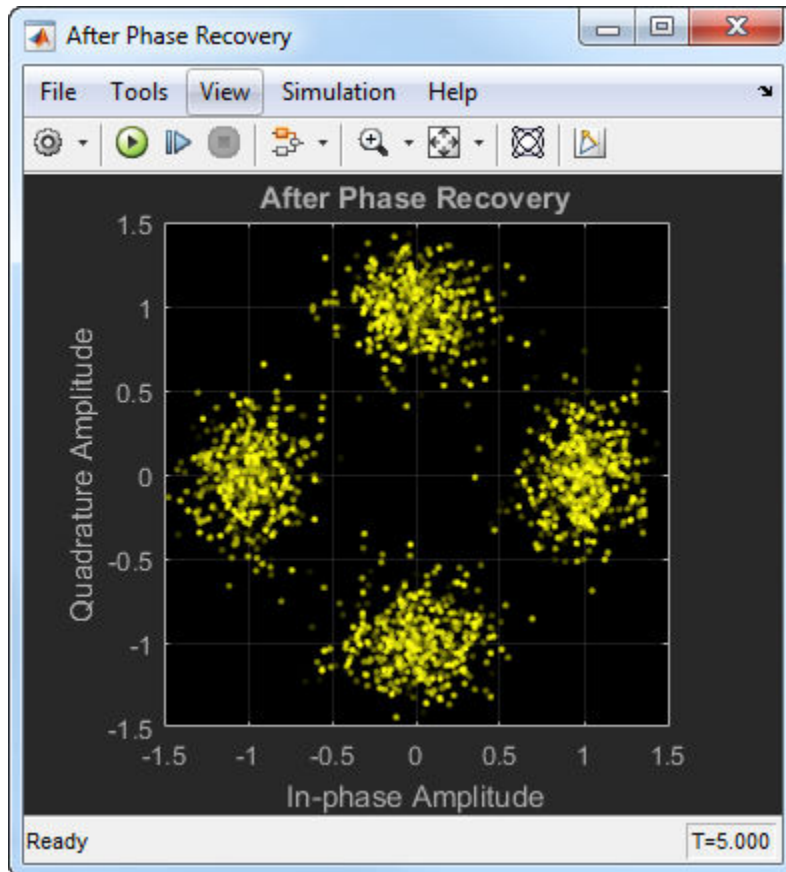
Scatter plot of received signal:



Scatter plot of signal after timing recovery:



Scatter plot of signal after carrier frequency and phase recovery:



You can also reset the BER computation after the signal has reached a steady state.

Experimenting with the Example

The example is designed so that you can vary the impairments independently while the simulation is running. You can also use the toggle switches to turn the recovery schemes on and off while the simulation is running, and then see the effects on the scatter plots.

Further items to investigate include:

- Set the frequency offset to θ and observe the displayed signal constellations and estimated phase offset.

- Observe that the Carrier Synchronizer block is set for a QPSK constellation with a phase offset of 0° .
- To see how the timing offset is tracked, replace the Constant block with a Sine Wave block. Vary the offset between 0 and 1 over the duration of the simulation.
- Vary the error update gain of the MSK-Type Signal Timing Recovery block to assess its ability to track constant and time-varying offsets. To access the block, open the *Timing Recovery* subsystem and then open the *Timing Recovery Algorithm* subsystem.

Reed-Solomon Coding

- “Reed-Solomon Coding Part I - Erasures” on page 11-2
- “Reed-Solomon Coding Part II - Punctures” on page 11-8
- “Reed-Solomon Coding Part III - Shortening” on page 11-14
- “Reed-Solomon Coding with Erasures, Punctures, and Shortening” on page 11-20
- “Estimate LDPC Performance in AWGN” on page 11-29
- “Character Representation of Polynomials” on page 11-31
- “Estimate BER of 8-PSK in AWGN with Reed-Solomon Coding” on page 11-32
- “Transmit and Receive Shortened Reed-Solomon Codes” on page 11-35

Reed-Solomon Coding Part I - Erasures

This example shows how to configure the RSEncoder and RSDecoder System objects to perform Reed-Solomon (RS) block coding with erasures when simulating a communications system. RS decoders can correct both errors and erasures. A receiver that identifies the most unreliable symbols in a given codeword can generate erasures. When a receiver erases a symbol, it replaces that symbol with a zero. The receiver then passes a flag to the decoder, indicating that the symbol is an erasure, not a valid code symbol. In addition, an encoder can generate punctures for which specific parity symbols are always removed from its output. The decoder, which knows the puncture pattern, inserts zeros in the puncture positions and treats those symbols as erasures. The decoder treats encoder-generated punctures and receiver-generated erasures the exact same way when it decodes a symbol. Puncturing also has the added benefit of making the code rate more flexible, at the expense of some error correction capability. Shortened codes achieve the same code rate flexibility without degrading the error correction performance, given the same demodulator input energy per bit to noise power spectral density ratio (E_b/N_0). Note that puncturing is the removal of parity symbols from a codeword, and shortening is the removal of message symbols from a codeword. In addition to this example, the examples “Reed-Solomon Coding Part II - Punctures” and “Reed-Solomon Coding Part III - Shortening” show RS block coding with punctures and shortened codes, respectively.

Introduction

This example shows the simulation of a communication system consisting of a random source, an RS encoder, a rectangular 64-QAM modulator, an AWGN channel, a rectangular 64-QAM demodulator, and an RS decoder. It includes analysis of RS coding with erasures by comparing the channel bit error rate (BER) performance versus the coded BER performance. This example obtains Channel BER by comparing inputs for the rectangular QAM modulator to outputs from the rectangular QAM demodulator. This example obtains Coded BER by comparing inputs for the RS encoder to outputs from the RS decoder. THIS IS FLAWED!

Initialization

The script file RSCodingConfigExample configures the rectangular 64-QAM modulator and demodulator, the AWGN channel, and the error rate measurement System objects used to simulate the communications system. The script also sets an uncoded E_b/N_0 ratio to $E_b/N_0_{\text{Uncoded}} = 15$ dB, and sets the simulation stop criteria by defining the target number of errors and the maximum number of bit transmissions to 500 and 5×10^6 respectively.

RSCodingConfigExample

Configuring the RS Encoder/Decoder

This example shows a (63,53) RS code operating with a 64-QAM modulation scheme. This code can correct $(63-53)/2 = 5$ errors, or it can alternatively correct $(63-53) = 10$ erasures. For each codeword at the output of the 64-QAM demodulator, the receiver determines the six least reliable symbols using the `RSCodingGetErasuresExample` function. The indices that point to the location of these unreliable symbols are passed as an input to the RS decoder. The RS decoder treats these symbols as erasures resulting in an error correction capability of $(10-6)/2 = 2$ errors per codeword.

Create a (63,53) `RSEncoder` System object and set the `BitInput` property to `false` to specify that the encoder inputs and outputs are integer symbols.

```
N = 63; % Codeword length
K = 53; % Message length
rsEncoder = comm.RSEncoder(N,K, 'BitInput', false);
numErasures = 6;
```

Create an `RSDecoder` System object using the same settings as in the encoder. Request an additional input for specifying erasures as an input to the object. This is done by setting the `ErasuresInputPort` property to `true`.

```
rsDecoder = comm.RSDecoder(N,K, 'BitInput', false, 'ErasuresInputPort', true);
```

Set the `NumCorrectedErrorsOutputPort` property to `true` so that the decoder outputs the number of corrected errors. A non negative value in the error output denotes the number of corrected errors in the input codeword. A value of -1 in the error output indicates a decoding error. A decoding error occurs when the input codeword has more errors than the error correction capability of the RS code.

```
rsDecoder.NumCorrectedErrorsOutputPort = true;
```

Stream Processing Loop

Simulate the communications system for an uncoded E_b/N_0 ratio of 15 dB. The uncoded E_b/N_0 is the ratio that would be measured at the input of the channel if there was no coding in the system.

The signal going into the AWGN channel is the encoded signal, so you must convert the uncoded E_b/N_0 values so that they correspond to the energy ratio at the encoder output. This ratio is the coded E_b/N_0 ratio. If you input K symbols to the encoder and obtain N

output symbols, then the energy relation is given by the K/N rate. Set the `EbNo` property of the AWGN channel object to the computed coded E_b/N_0 value.

```
EbNoCoded = EbNoUncoded + 10*log10(K/N);  
channel.EbNo = EbNoCoded;
```

Loop until the simulation reaches the target number of errors or the maximum number of transmissions.

```
chanErrorStats = zeros(3,1);  
codedErrorStats = zeros(3,1);  
correctedErrors = 0;  
while (codedErrorStats(2) < targetErrors) && ...  
    (codedErrorStats(3) < maxNumTransmissions)  
  
    % Data symbols - transmit 1 message word at a time. Each message word has  
    % K symbols in the [0 N] range.  
    data = randi([0 N],K,1);  
  
    % Encode the message word. The encoded word encData is N symbols long.  
    encData = rsEncoder(data);  
  
    % Modulate encoded data.  
    modData = qamModulator(encData);  
  
    % Add noise.  
    chanOutput = channel(modData);  
  
    % Demodulate channel output.  
    demodData = qamDemodulator(chanOutput);  
  
    % Find the 6 least reliable symbols and generate an erasures vector using  
    % the RSCodingGetErasuresExample function. The length of the erasures vector  
    % must be equal to the number of symbols in the demodulated codeword. A  
    % one in the ith element of the vector erases the ith symbol in the  
    % codeword. Zeros in the vector indicate no erasures.  
    erasuresVec = RSCodingGetErasuresExample(chanOutput, numErasures);  
  
    % Decode data.  
    [estData, errs] = rsDecoder(demodData,erasuresVec);  
  
    % If a decoding error did not occur, accumulate the number of corrected  
    % errors using the cumulative sum object.  
    if errs >= 0  
        correctedErrors = cumulativeSum(errs);
```

```

end

% Convert integers to bits and compute the channel BER.
chanErrorStats(:,1) = ...
    chanBERCalc(intToBit1(encData),intToBit1(demodData));

% Convert integers to bits and compute the coded BER.
codedErrorStats(:,1) = ...
    codedBERCalc(intToBit2(data),intToBit2(estData));
end

```

The error rate measurement objects, `chanBERCalc` and `codedBERCalc`, output 3-by-1 vectors containing BER measurement updates, the number of errors, and the total number of bit transmissions. Display the coded BER and the total number of errors corrected by the RS decoder.

```

codedBitErrorRate = codedErrorStats(1)
totalCorrectedErrors = correctedErrors

```

```

codedBitErrorRate =

```

```

    0

```

```

totalCorrectedErrors =

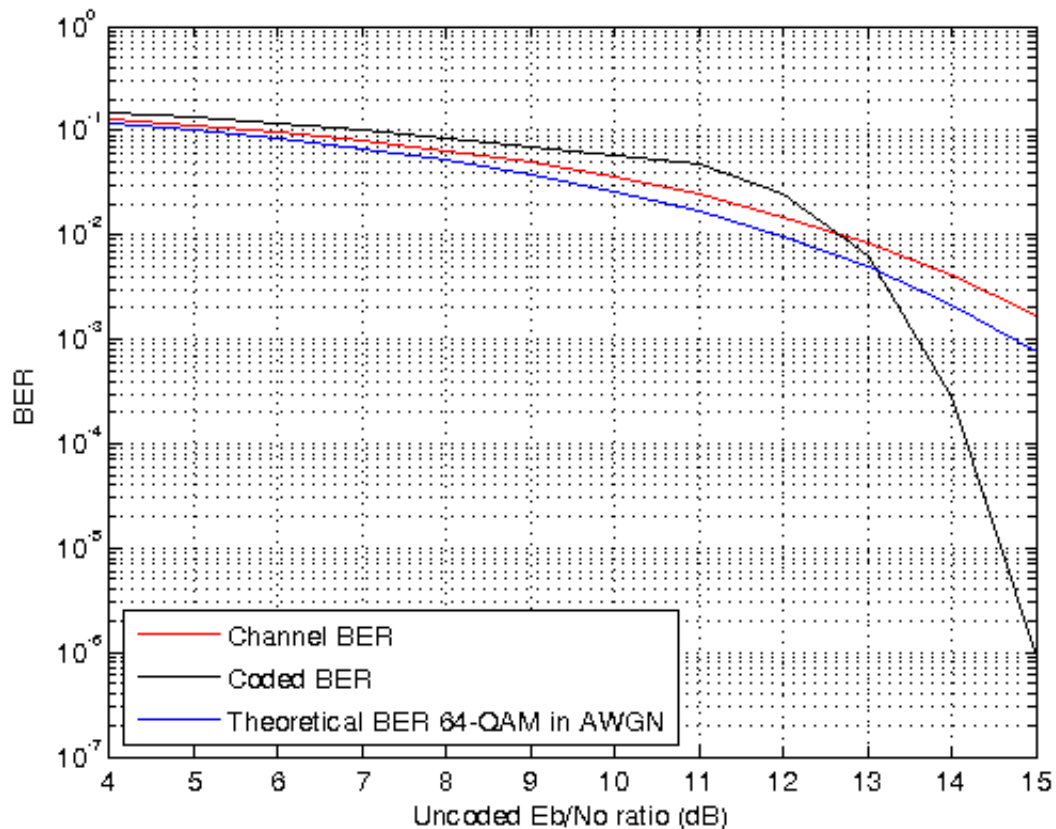
```

```

    882

```

You can add a for loop around the processing loop above to run simulations for a set of E_b/N_0 values. Simulations were run offline for uncoded E_b/N_0 values in 4:15 dB, target number of errors equal to 5000, and maximum number of transmissions equal to 50×10^6 . The results from the simulation are shown. The channel BER is worse than the theoretical 64-QAM BER because E_b/N_0 is reduced by the code rate.



Summary

This example utilized several System objects to simulate a rectangular 64-QAM communications system over an AWGN channel with RS block coding. It showed how to configure the RS decoder to decode symbols with erasures. System performance was measured using channel and coded BER curves obtained using error rate measurement System objects.

The examples “Reed-Solomon Coding Part II – Punctures” and “Reed-Solomon Coding Part III – Shortening” show how to perform RS block coding with punctured and shortened codes, respectively.

Appendix

This example uses the following script and helper function:

- RSCodingConfigExample
- RSCodingGetErasuresExample

Selected Bibliography

[1] G. C. Clark, Jr., J. B. Cain, *Error-Correction Coding for Digital Communications*, Plenum Press, New York, 1981.

Reed-Solomon Coding Part II - Punctures

This example shows how to set up the Reed-Solomon (RS) encoder/decoder to use punctured codes. An encoder can generate punctures for which specific parity symbols are always removed from its output. The decoder, which knows the puncture pattern, inserts zeros in the puncture positions and treats those symbols as erasures. The decoder treats encoder-generated punctures and receiver-generated erasures in exactly the same way when it decodes. Puncturing has the added benefit of making the code rate more flexible, at the expense of some error correction capability.

In addition to this example, the example “Reed-Solomon Coding Part I - Erasures” shows a rectangular 64-QAM communications system with a (63,53) RS block code with erasures, and the example “Reed-Solomon Coding Part III - Shortening” shows RS block coding with shortened codes.

Introduction

This example shows the simulation of a communication system consisting of a random source, an RS encoder, a rectangular 64-QAM modulator, an AWGN channel, a rectangular 64-QAM demodulator, and an RS decoder. It includes analysis of RS coding with erasures and puncturing by comparing the channel bit error rate (BER) performance versus the coded BER performance. This example obtains Channel BER by comparing inputs for the rectangular QAM modulator to outputs from the rectangular QAM demodulator. This example obtains Coded BER by comparing inputs for the RS encoder to outputs from the RS decoder.

Initialization

The script file `RSCodingConfigExample` configures the rectangular 64-QAM modulator and demodulator, the AWGN channel, and the error rate measurement System objects used to simulate the communications system. The script also sets an uncoded E_b/N_0 ratio to $E_b/N_0 = 15$ dB, and sets the simulation stop criteria by defining the target number of errors and the maximum number of bit transmissions to 500 and 5×10^6 , respectively.

`RSCodingConfigExample`

Configuring the RS Encoder/Decoder

Consider the same (63,53) RS code operating in concert with a 64-QAM modulation scheme that was used in the example “Reed-Solomon Coding Part I – Erasures” to

showcase how to decode erasures. This example shows how to set up the RS encoder/decoder to use a punctured code. In addition to decoding receiver-generated erasures, the RS decoder can correct encoder-generated punctures. The decoding algorithm is identical for the two cases. For each codeword, the sum of the punctures and erasures cannot exceed twice the error-correcting capability of the code.

```
N = 63; % Codeword length
K = 53; % Message length
numErasures = 6;
rsEncoder = comm.RSEncoder(N,K, 'BitInput', false);
rsDecoder = comm.RSDecoder(N,K, 'BitInput', false, 'ErasuresInputPort', true);
```

To enable code puncturing you set the `PuncturePatternSource` property to 'Property' and set the `PuncturePattern` property to the desired puncture pattern vector. The same puncture vector must be specified in both the encoder and decoder. This example punctures two symbols from each codeword. Values of one in the puncture pattern vector indicate nonpunctured symbols, while values of zero indicate punctured symbols.

```
numPuncs = 2;

rsEncoder.PuncturePatternSource = 'Property';
rsEncoder.PuncturePattern = [ones(N-K-numPuncs,1); zeros(numPuncs,1)];

rsDecoder.PuncturePatternSource = 'Property';
rsDecoder.PuncturePattern = rsEncoder.PuncturePattern;
```

Stream Processing Loop

Simulate the communications system for an uncoded E_b/N_0 ratio of 15 dB. The uncoded E_b/N_0 is the ratio that would be measured at the input of the channel if there was no coding in the system.

The length of the codewords generated by the RS encoder is reduced by the number of punctures specified in the puncture pattern vector. For this reason, the value of the coded E_b/N_0 ratio needs to be adjusted to account for these punctures. In this example, the uncoded E_b/N_0 ratio relates to the coded E_b/N_0 as shown below. Set the `EbNo` property of the AWGN channel object to the computed coded E_b/N_0 value.

```
EbNoCoded = EbNoUncoded + 10*log10(K/(N - numPuncs));
channel.EbNo = EbNoCoded;
```

Loop until the simulation reaches the target number of errors or the maximum number of transmissions.

```
chanErrorStats = zeros(3,1);
codedErrorStats = zeros(3,1);
correctedErrors = 0;
while (codedErrorStats(2) < targetErrors) && ...
    (codedErrorStats(3) < maxNumTransmissions)

    % Data symbols - transmit 1 message word at a time. Each message word
    % has K symbols in the [0 N] range.
    data = randi([0 N],K,1);

    % Encode the message word. The encoded word encData is N-numPuncs symbols
    % long.
    encData = rsEncoder(data);

    % Modulate encoded data.
    modData = qamModulator(encData);

    % Add noise.
    chanOutput = channel(modData);

    % Demodulate channel output.
    demodData = qamDemodulator(chanOutput);

    % Get erasures vector.
    erasuresVec = RSCodingGetErasuresExample(chanOutput,numErasures);

    % Decode data.
    [estData,errs] = rsDecoder(demodData,erasuresVec);

    % If a decoding error did not occur, accumulate the number of corrected
    % errors using the cumulative sum object.
    if errs >= 0
        correctedErrors = cumulativeSum(errs);
    end

    % Convert integers to bits and compute the channel BER.
    chanErrorStats(:,1) = ...
        chanBERCalc(intToBit1(encData),intToBit1(demodData));

    % Convert integers to bits and compute the coded BER.
    codedErrorStats(:,1) = ...
        codedBERCalc(intToBit2(data),intToBit2(estData));
end
```

The error rate measurement objects, `chanBERCalc` and `codedBERCalc`, output 3-by-1 vectors containing updates of the measured BER value, the number of errors, and the total number of bit transmissions. Display the coded BER and the total number of errors corrected by the RS decoder.

```
codedBitErrorRate = codedErrorStats(1)
totalCorrectedErrors = correctedErrors
```

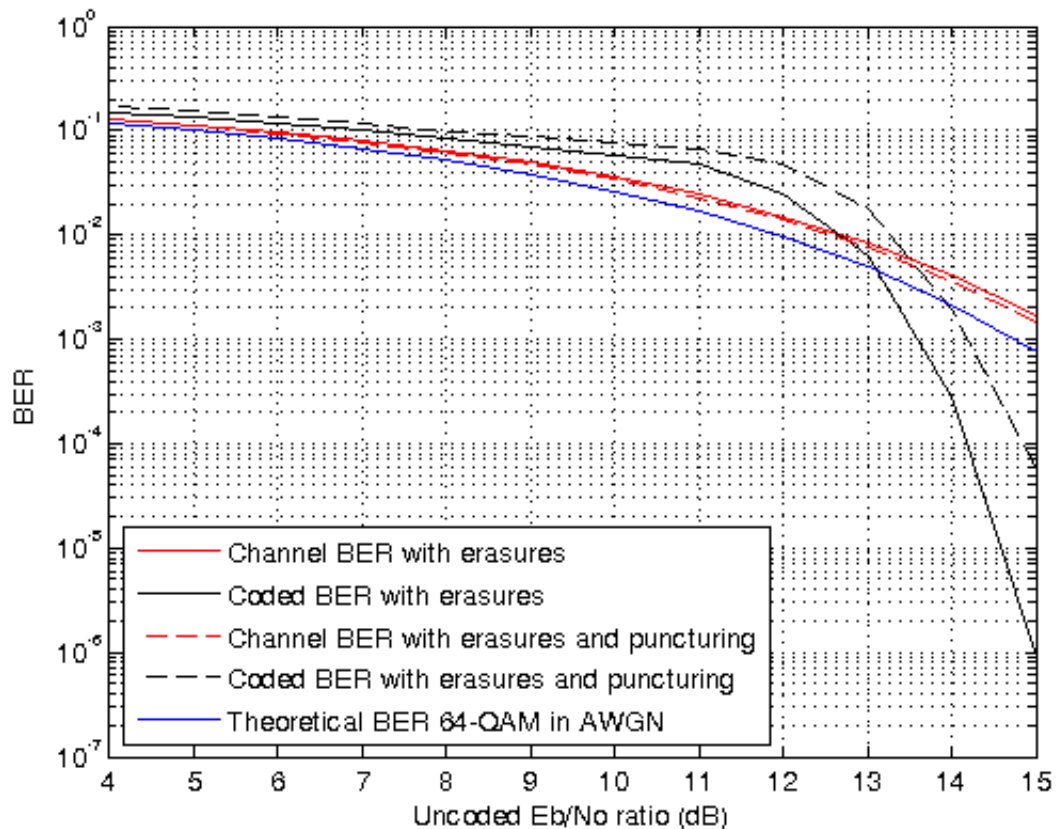
```
codedBitErrorRate =
```

```
    4.3198e-05
```

```
totalCorrectedErrors =
```

```
    578
```

You can add a for loop around the processing loop above to run simulations for a set of E_b/N_0 values. Simulations were run offline for uncoded E_b/N_0 values in 4:15 dB, target number of errors equal to 5000, and maximum number of transmissions equal to 50×10^6 . The results from the simulation are shown in the following figure. For comparison, the figure also shows the results obtained in the example “Reed-Solomon Coding Part I - Erasures”, which correspond to a system with erasures but no puncturing.



From the curves, observe that the channel BER is slightly better in the punctured case. The reason for this is that the coded E_b/N_0 is slightly higher. On the other hand, the coded BER is worse in the punctured case, because the two punctures reduce the error correcting capability of the code by one, leaving it able to correct only $(10-6-2)/2 = 1$ error per codeword.

Summary

The example utilized several System objects to simulate a rectangular 64-QAM communications system over an AWGN channel with RS block coding. It showed how to configure the RS encoder/decoder System objects to obtain punctured codes. System

performance was measured using channel and coded BER curves obtained using the error rate measurement System objects.

The example “Reed-Solomon Coding Part III - Shortening” shows how to perform RS block coding with shortened codes.

Appendix

This example uses the following script and helper function:

- RSCodingConfigExample
- RSCodingGetErasuresExample

Selected Bibliography

[1] G. C. Clark, Jr., J. B. Cain, *Error-Correction Coding for Digital Communications*, Plenum Press, New York, 1981.

Reed-Solomon Coding Part III - Shortening

This example shows how to set up the Reed-Solomon (RS) encoder/decoder to shorten the (63,53) code to a (28,18) code.

In addition to this example, the examples “Reed-Solomon Coding Part I - Erasures” and “Reed-Solomon Coding Part II - Punctures” show a rectangular 64-QAM communications system with a (63,53) RS block code with erasures and punctures respectively. As was mentioned in the two examples, puncturing has the benefit of making the code rate more flexible, at the expense of some error correction capability. Shortened codes achieve the same code rate flexibility without degrading the error correction performance, given the same demodulator input E_b/N_0 . Note that puncturing is the removal of parity symbols from a codeword, and shortening is the removal of message symbols from a codeword.

Introduction

This example shows the simulation of a communication system consisting of a random source, an RS encoder, a rectangular 64-QAM modulator, an AWGN channel, a rectangular 64-QAM demodulator, and an RS decoder. It analyzes the effects of RS coding with erasures, puncturing, and shortening, by comparing the channel bit error rate (BER) performance versus the coded BER performance. This example obtains Channel BER by comparing inputs for the rectangular QAM modulator to outputs from the rectangular QAM demodulator. This example obtains Coded BER by comparing inputs for the RS encoder to outputs from the RS decoder.

Initialization

The script file `RSCodingConfigExample` configures the rectangular 64-QAM modulator and demodulator, the AWGN channel, and the error rate measurement System objects used to simulate the communications system. The script also sets an uncoded E_b/N_0 ratio to `EbNoUncoded = 15 dB`, and sets the simulation stop criteria by defining the target number of errors and the maximum number of bit transmissions to 500 and 5×10^6 , respectively.

```
RSCodingConfigExample
```

Configuring the RS Encoder/Decoder

Consider the same (63,53) RS code operating in concert with a 64-QAM modulation scheme that was used in the examples Reed-Solomon Coding Part I - Erasures and Reed-Solomon Coding Part II - Punctures to showcase how to decode erasures and how to puncture the code. This example shows how to shorten the (63,53) code to a (28,18) code.

Shortening a block code removes symbols from its message portion, while puncturing removes symbols from its parity portion. You can incorporate both techniques with the RS encoder and decoder System objects.

For example, to shorten a (63,53) code to a (53,43) code, you can simply enter 53 and 43 for the `CodewordLength` and `MessageLength` properties, respectively (since $2^{\lceil \log_2(53+1) \rceil} - 1 = 63$). However, to shorten it by 35 symbols to a (28,18) code, you must explicitly specify that the symbols belong to the Galois field $GF(2^6)$. Otherwise, the RS blocks will assume that the code is shortened from a (31,21) code (since $2^{\lceil \log_2(28+1) \rceil} - 1 = 31$).

Set the RS encoder/decoder System objects so that they perform block coding with a (28,18) code shortened from a (63,53) code.

Create a (63,53) `RSEncoder` System object, and an `RSDecoder` object with an erasures input port, and two punctures.

```
N = 63; % Codeword length
K = 53; % Message length
S = 18;
numErasures = 6;
numPuncs = 2;
rsEncoder = comm.RSEncoder(N, K, 'BitInput', false);
rsDecoder = comm.RSDecoder(N, K, 'BitInput', false, 'ErasuresInputPort', true);
rsEncoder.PuncturePatternSource = 'Property';
rsEncoder.PuncturePattern = [ones(N-K-numPuncs,1); zeros(numPuncs,1)];
rsDecoder.PuncturePatternSource = 'Property';
rsDecoder.PuncturePattern = rsEncoder.PuncturePattern;

% Set the shortened message length values
rsEncoder.ShortMessageLength = S;
rsDecoder.ShortMessageLength = S;
```

Specify the field of $GF(2^6)$ in the RS encoder/decoder System objects, by setting the `PrimitivePolynomialSource` property to 'Property' and the `PrimitivePolynomial` property to a 6th degree primitive polynomial.

```
rsEncoder.PrimitivePolynomialSource = 'Property';
rsEncoder.PrimitivePolynomial = de2bi(primpoly(6, 'nodisplay'), 'left-msb');

rsDecoder.PrimitivePolynomialSource = 'Property';
rsDecoder.PrimitivePolynomial = de2bi(primpoly(6, 'nodisplay'), 'left-msb');
```

Stream Processing Loop

Simulate the communications system for an uncoded E_b/N_0 ratio of 15 dB. The uncoded E_b/N_0 is the ratio that would be measured at the input of the channel if there was no coding in the system.

Shortening alters the code rate much like puncturing does. You must adjust the value of the coded E_b/N_0 ratio to account for punctures and shortening. In this example, the uncoded E_b/N_0 ratio relates to the coded E_b/N_0 , as shown in the following syntax. Set the property of the AWGN channel object to the computed coded E_b/N_0 value.

```
EbNoCoded = EbNoUncoded + ...
    10*log10(S/(N - numPuncs - K + S));
```

```
channel.EbNo = EbNoCoded;
```

Loop until the simulation reaches the target number of errors or the maximum number of transmissions.

```
chanErrorStats = zeros(3,1);
codedErrorStats = zeros(3,1);
while (codedErrorStats(2) < targetErrors) && ...
    (codedErrorStats(3) < maxNumTransmissions)

    % Data symbols - transmit 1 message word at a time, each message word has
    % S symbols in the [0 2^P-1] range. P is the degree of the
    % primitive polynomial specified in the RS encoder/decoder, which in this
    % example equals 6.
    data = randi([0 2^6-1],S,1);

    % Encode the shortened message word. The encoded word encData is
    % N-numPuncs-K+S symbols long.
    encData = rsEncoder(data);

    % Modulate encoded data.
    modData = qamModulator(encData);

    % Add noise.
    chanOutput = channel(modData);

    % Demodulate channel output.
    demodData = qamDemodulator(chanOutput);

    % Get erasures vector.
    erasuresVec = RSCodingGetErasuresExample(chanOutput,numErasures);
```

```

% Decode data.
[estData,errs] = rsDecoder(demodData,erasuresVec);

% If a decoding error did not occur, accumulate the number of corrected
% errors using the cumulative sum object.
if errs >= 0
    correctedErrors = cumulativeSum(errs);
end

% Convert integers to bits and compute the channel BER.
chanErrorStats(:,1) = ...
    chanBERCalc(intToBit1(encData),intToBit1(demodData));

% Convert integers to bits and compute the coded BER.
codedErrorStats(:,1) = ...
    codedBERCalc(intToBit2(data),intToBit2(estData));
end

```

The error rate measurement objects, `chanBERCalc` and `codedBERCalc`, output 3-by-1 vectors containing updates of the measured BER value, the number of errors, and the total number of bit transmissions. Display the coded BER and the total number of errors corrected by the RS decoder.

```

codedBitErrorRate = codedErrorStats(1)
totalCorrectedErrors = correctedErrors

```

```

codedBitErrorRate =

```

```

    9.6599e-05

```

```

totalCorrectedErrors =

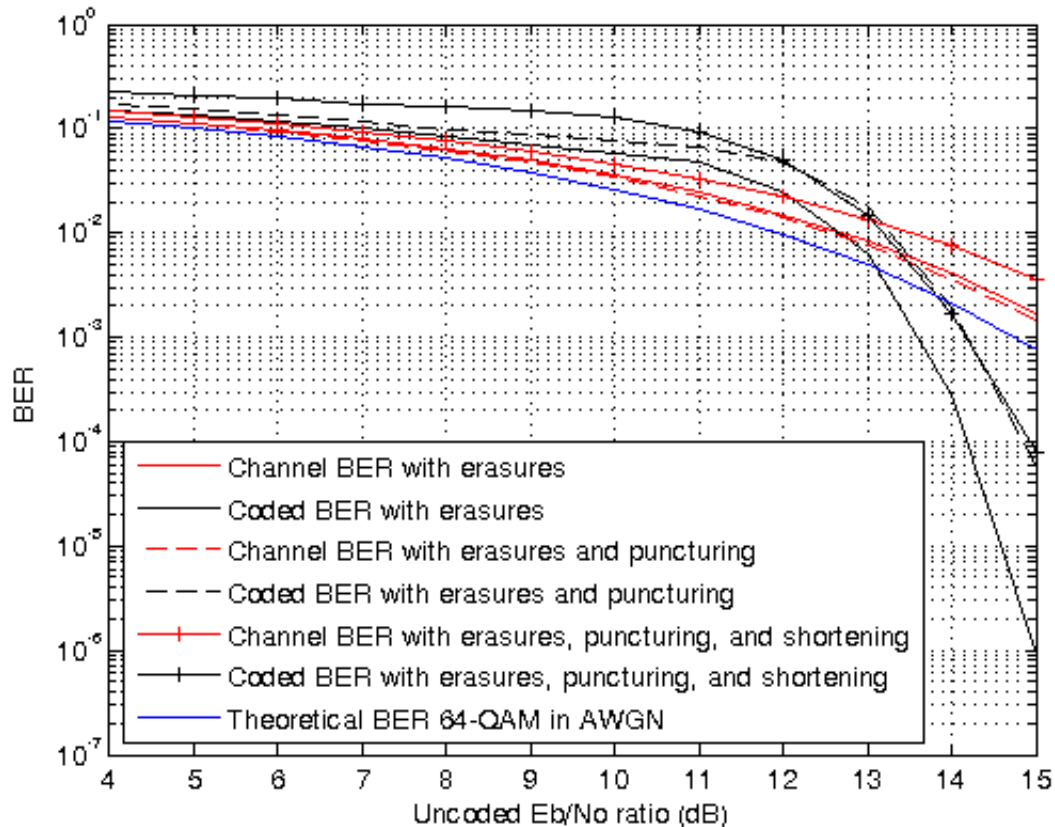
```

```

    1436

```

You can add a for loop around the processing loop above to run simulations for a set of E_b/N_0 values. Simulations were run offline for uncoded E_b/N_0 values in 4:15 dB, target number of errors equal to 5000, and maximum number of transmissions equal to 50×10^6 . The results from the simulation are shown in the following figure. For comparison, the figure also shows the results obtained in the examples “Reed-Solomon Coding Part I - Erasures” and “Reed-Solomon Coding Part II - Punctures”. The results of “Reed-Solomon Coding Part I - Erasures” correspond to a system with erasures but no puncturing. The results of “Reed-Solomon Coding Part II - Punctures” correspond to a system with erasures and punctures.



From the curves, observe that the channel BER is worse with shortening because the coded E_b/N_0 is worse. This degraded coded E_b/N_0 occurs because the code rate of the shortened code is much lower than that of the nonshortened code. As a result, the coded BER is also worse with shortening than without, especially at lower E_b/N_0 values.

Summary

This example utilized several System objects to simulate a rectangular 64-QAM communications system over an AWGN channel with a shortened RS block code. It showed how to configure the RS encoder/decoder to shorten a (63,53) code to a (28,18)

code. System performance was measured using channel and coded BER curves obtained with the help of error rate measurement System objects.

The examples “Reed-Solomon Coding Part I - Erasures”, “Reed-Solomon Coding Part II - Punctures”, and “Reed-Solomon Coding Part III - Shortening” show how to configure RS encoders and decoders to perform block coding with erasures, puncturing, and shortening.

Appendix

This example uses the following script and helper function:

- RSCodingConfigExample
- RSCodingGetErasuresExample

Selected Bibliography

[1] G. C. Clark, Jr., J. B. Cain, *Error-Correction Coding for Digital Communications*, Plenum Press, New York, 1981.

Reed-Solomon Coding with Erasures, Punctures, and Shortening

In this section...

“Decoding with Receiver Generated Erasures” on page 11-21

“Simulation and Visualization with Erasures Only” on page 11-21

“BER Performance with Erasures Only” on page 11-24

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“Simulation with Erasures, Punctures, and Shortening” on page 11-27

“BER Performance with Erasures, Punctures, and Shortening” on page 11-28

“Further Exploration” on page 11-28

This model shows how to configure Reed-Solomon (RS) codes to perform block coding with erasures, punctures, and shortening.

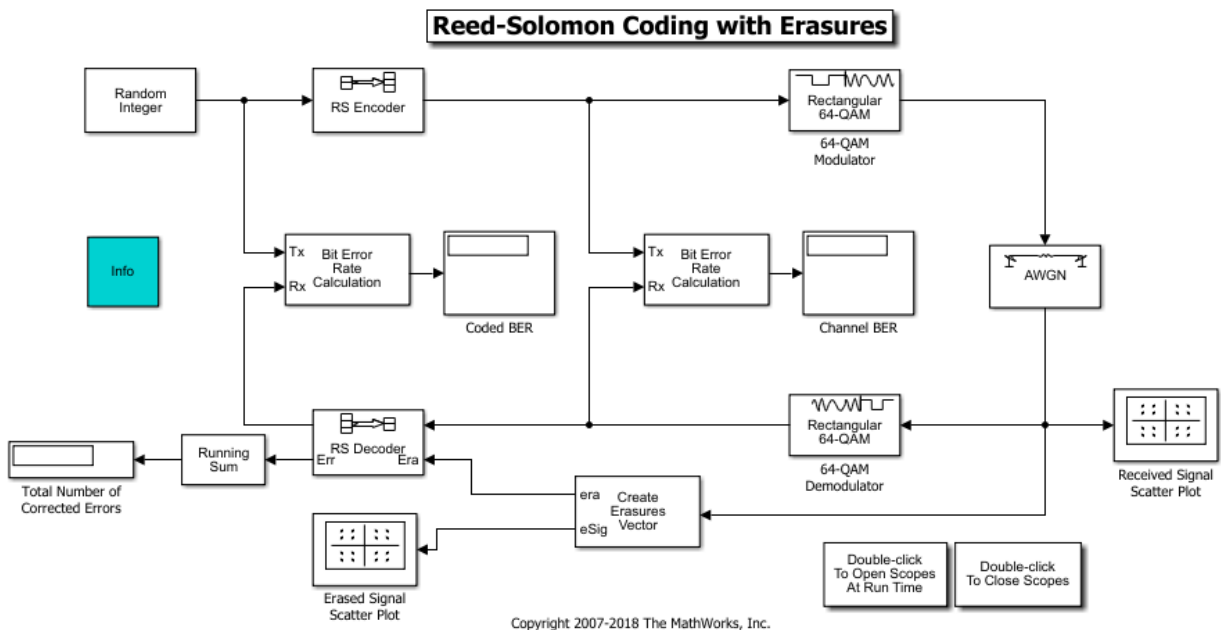
RS decoders can correct both errors and erasures. The erasures can be generated by a receiver that identifies the most unreliable symbols in a given codeword. When a receiver erases a symbol, it replaces the symbol with a zero and passes a flag to the decoder indicating that the symbol is an erasure, not a valid code symbol.

In addition, an encoder can generate punctures for which specific parity symbols are always removed from its output. The decoder, which knows the puncture pattern, inserts zeros in the puncture positions and treats those symbols as erasures. The decoder treats encoder-generated punctures and receiver-generated erasures in exactly the same way when it decodes.

Puncturing has the added benefit of making the code rate a bit more flexible, at the expense of some error correction capability. Shortened codes achieve the same code rate flexibility without degrading the error correction performance, given the same demodulator input E_b/N_0 . Note that puncturing is the removal of parity symbols from a codeword, and shortening is the removal of message symbols from a codeword.

Decoding with Receiver Generated Erasures

This example shows a (63,53) RS code operating in concert with a 64-QAM modulation scheme. Since the code can correct $(63-53)/2 = 5$ errors, it can alternatively correct $(63-53) = 10$ erasures. For each demodulated codeword, the receiver determines the six least reliable symbols by finding the symbols within a decision region that are nearest to a decision boundary. It then erases those symbols. We first open the model RSCodingErasuresExample.



Simulation and Visualization with Erasures Only

We then define system simulation parameters:

```

RS_TsUncoded = 1;           % Sample time (s)
RS_n = 63;                 % Codeword length
RS_k = 53;                 % Message length
RS_MQAM = 64;              % QAM order
RS_numBitsPerSymbol = ... % 6 bits per symbol
    log2(RS_MQAM);
RS_sigPower = 42;          % Assume points at +/-1, +/-3, +/-5, +/-7

```

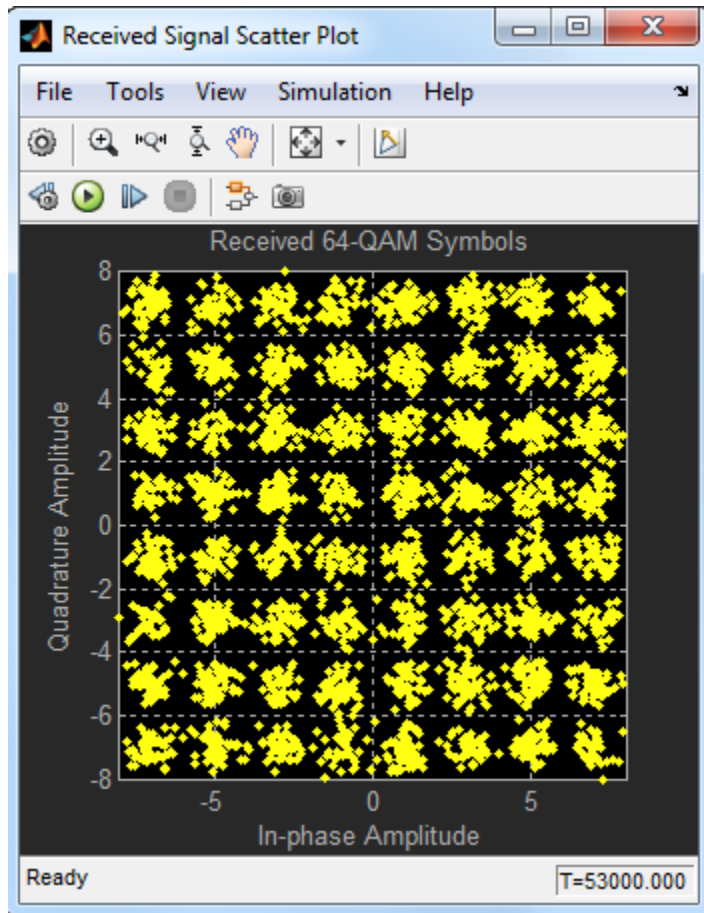
```
RS_numErasures = 6;           % Number of erasures  
RS_EbNoUncoded = 15;         % In dB
```

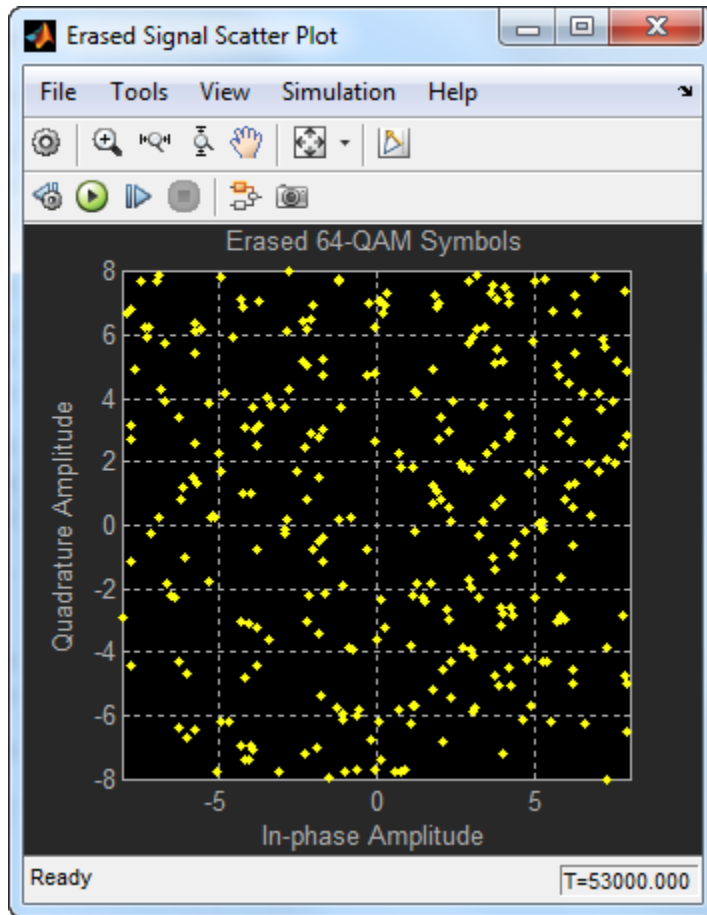
The system is simulated at an uncoded E_b/N_0 of 15 dB. However, the coded E_b/N_0 is reduced because of the redundant symbols added by the RS Encoder. Also, the period of each frame in the model remains constant at 53 seconds, corresponding to a sample time of 1 second at the output of the Random Integer Generator. Moreover, the symbol time at the output of the RS Encoder is reduced by a factor of the code rate, because 63 symbols are output over the frame time of 53 seconds. The AWGN Channel block accounts for this by using the following parameters:

```
RS_EbNoCoded = RS_EbNoUncoded + 10*log10(RS_k/RS_n);  
RS_TsymCoded = RS_TsymUncoded * (RS_k/RS_n);
```

The receiver determines which symbols to erase by finding the 64-QAM symbols, per codeword, that are closest to a decision boundary. It deletes the six least reliable code symbols, which still allows the RS Decoder to correct $(10-6)/2 = 2$ errors per codeword.

We simulate the system, showing the received symbols and those symbols that were erased:





BER Performance with Erasures Only

Now let's examine the BER performance at the output of the decoder. We set the stop time of the simulation to inf, then simulate until 100 bit errors are collected out of the RS Decoder. The 64-QAM BER is shown below, followed by the RS BER. This convention is followed for the remainder of this example.

```
BER_eras =
    1.7049e-03    2.5906e-06
```

Simulation with Erasures and Punctures

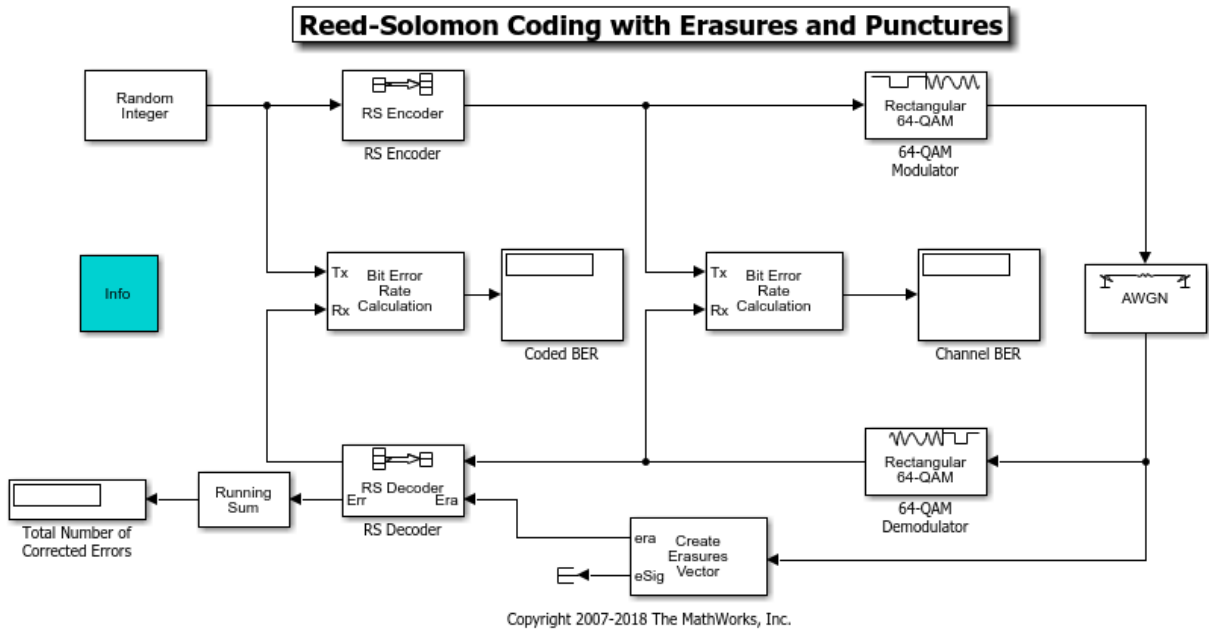
In addition to decoding receiver-generated erasures, the RS Decoder can correct encoder-generated punctures. The decoding algorithm is identical for the two cases, but the per-codeword sum of the punctures and erasures cannot exceed twice the error-correcting capability of the code. Consider the following model that performs decoding for both erasures and punctures.

The same puncture vector is specified in both the encoder and decoder blocks. This example punctures two symbols from each codeword. Vector values of "1" indicate nonpunctured symbols, while values of "0" indicate punctured symbols. In the erasures vector, however, values of "1" indicate erased symbols, while values of "0" indicate nonerased symbols.

Several of the parameters for the AWGN Channel block are now slightly different, because the length of the codeword is now different from the previous example. The block accounts for the size difference with the following code:

```
RS_EbNoCoded = RS_EbNoUncoded + 10*log10( RS_k / (RS_n - RS_numPuncs) );  
RS_TsymCoded = RS_TsUncoded * ( RS_k / (RS_n - RS_numPuncs) );
```

We simulate the model, `RSCodingErasuresPunctExample.mdl`, collecting 1000 errors out of the RS Decoder block. Due to puncturing, the signal dimensions out of the encoder are 61-by-1, rather than 63-by-1 in the model with no puncturing. The Create Erasures Vector subsystem must also account for the size differences as it creates a 61-by-1 erasures vector. Open the model `RSCodingErasuresPunctExample`.



BER Performance with Erasures and Punctures

Let's compare the BERs for erasures decoding with and without puncturing.

The BER out of the 64-QAM Demodulator is slightly better in the punctured case, because the E_b/N_0 into the demodulator is slightly higher. However, the BER out of the RS Decoder is much worse in the punctured case, because the two punctures reduce the error correcting capability of the code by one, leaving it able to correct only $(10-6-2)/2 = 1$ error per codeword.

```

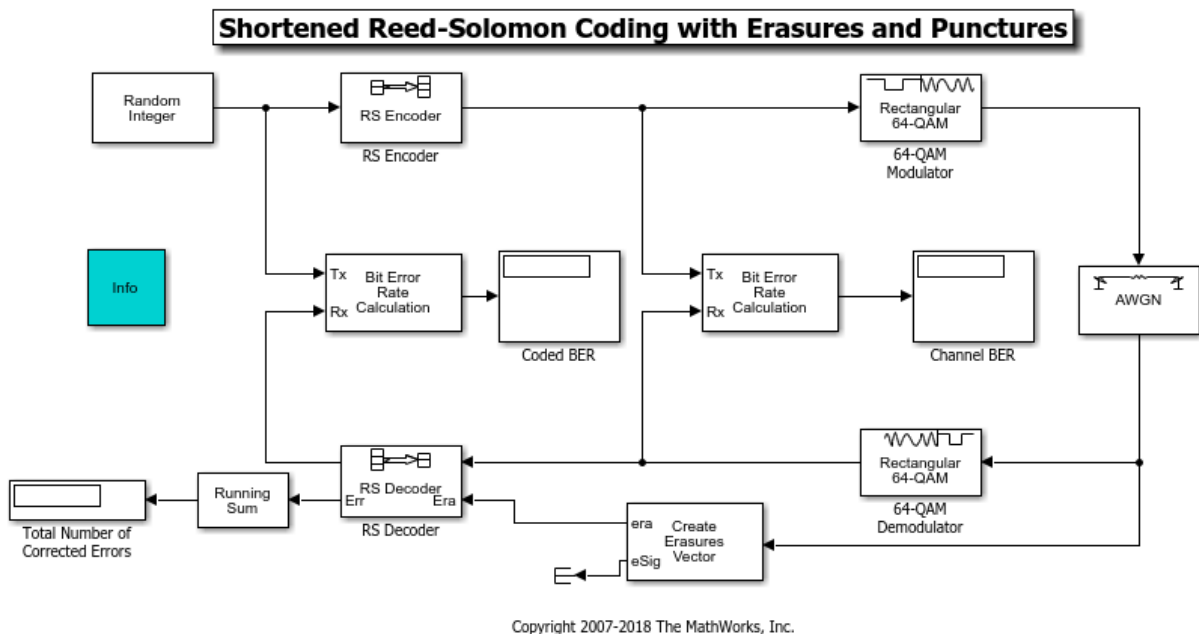
BER_eras =
    1.7049e-03    2.5906e-06
BER_eras_punc =
    1.4767e-03    6.1103e-05
    
```

Specifying a Shortened Code

Shortening a block code removes symbols from its message portion, where puncturing removes symbols from its parity portion. You can incorporate both techniques with the RS encoder and decoder blocks.

For example, to shorten a (63,53) code to a (53,43) code, you can simply enter 63, 53 and 43 for n , k , and s respectively, in the encoder and decoder block masks.

Open the model RSCodingErasuresPunctShortExample.



Simulation with Erasures, Punctures, and Shortening

Because shortening alters the code rate much like puncturing does, the AWGN parameters must be changed again. The AWGN Channel block accounts for this with the following code:

```
RS_EbNoCoded = RS_EbNoUncoded + ...
    10*log10( RS_s / (RS_n - RS_k + RS_s - RS_numPuncs) );
```

```
RS_TsymCoded = RS_TsUncoded * ...  
RS_s / (RS_n - RS_k + RS_s - RS_numPuncs);
```

We simulate the model, once again collecting 1000 errors out of the RS Decoder block. Note that the signal dimensions out of the RS Encoder are 26x1, due to 35 symbols of shortening and 2 symbols of puncturing. Once again, the Create Erasures Vector subsystem must also account for the size difference caused by the shortened code.

BER Performance with Erasures, Punctures, and Shortening

Let's compare the BER performance for decoding with erasures only, with erasures and punctures, and with erasures, punctures, and shortening.

The BER out of the 64-QAM Demodulator is worse with shortening than it is without shortening. This is because the code rate of the shortened code is much lower than the code rate of the non-shortened code and therefore the coded E_b/N_0 into the demodulator is worse with shortening. A shortened code has the same error correcting capability as non-shortened code for the same E_b/N_0 , but the reduction in E_b/N_0 manifests in the form of a higher BER out of the RS Decoder with shortening than without.

```
BER_eras =  
    1.7049e-03    2.5906e-06  
BER_eras_punc =  
    1.4767e-03    6.1103e-05  
BER_eras_punc_short =  
    3.5975e-03    9.1940e-05
```

Further Exploration

You can experiment with these systems by running them over a loop of E_b/N_0 values and generating a BER curve for them. You can then compare their performance against a theoretical 64-QAM/RS system without erasures, punctures, or shortening. Use BERTool to generate the theoretical BER curves.

Estimate LDPC Performance in AWGN

Transmit an LDPC-encoded, QPSK-modulated bit stream through an AWGN channel. Then demodulate, decode, and count errors.

Create an LDPC encoder and decoder pair.

```
ldpcEnc = comm.LDPCEncoder;
ldpcDec = comm.LDPCDecoder;
```

Create a QPSK modulator, a QPSK demodulator, and an Error Rate detector.

```
qpskMod = comm.QPSKModulator('BitInput',true);
qpskDemod = comm.QPSKDemodulator('BitOutput',true,...
    'DecisionMethod','Approximate log-likelihood ratio', ...
    'VarianceSource','Input port');
errorCnt = comm.ErrorRate;
```

Create a vector of SNR values to evaluate. Initialize the bit error rate vector.

```
snrVec = [0 0.2 0.4 0.6 0.65 0.7 0.75 0.8];
ber = zeros(length(snrVec),1);
```

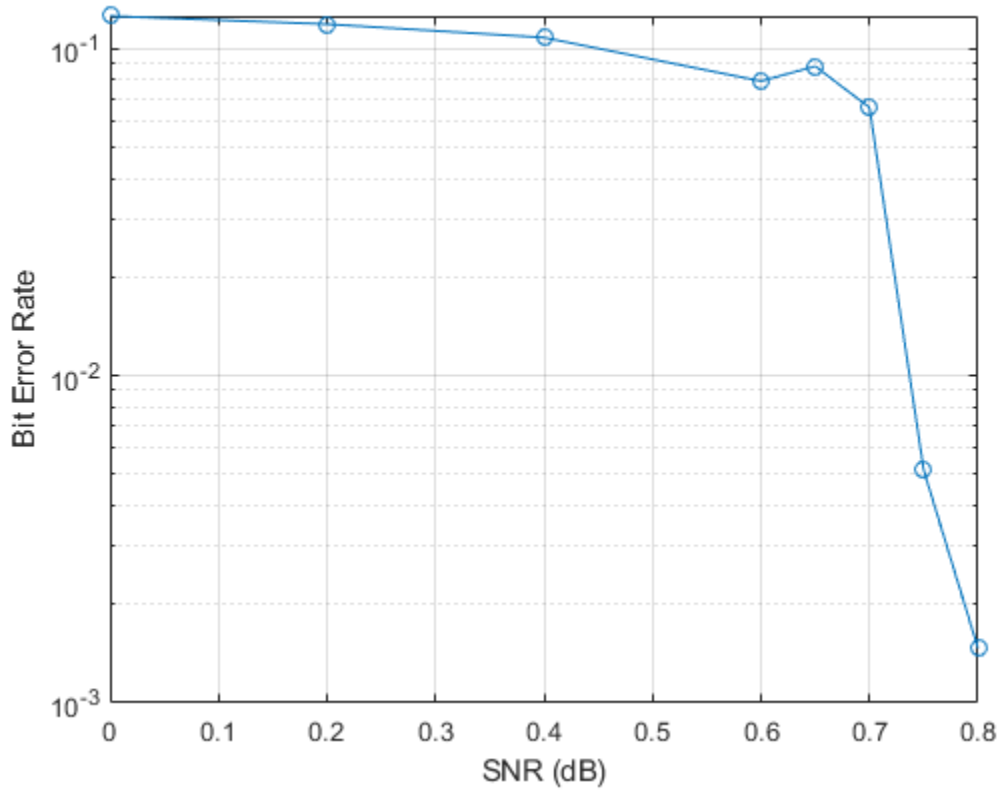
Encode, modulate, and transmit 32400-bit frames of binary data through an AWGN channel. Then, demodulate, decode, and estimate the bit error rate.

```
for k = 1:length(snrVec)
    noiseVar = 1/10^(snrVec(k)/10);
    errorStats = zeros(1,3);
    while errorStats(2) <= 200 && errorStats(3) < 5e6
        data = logical(randi([0 1],32400,1)); % Generate binary data
        encData = ldpcEnc(data); % Apply LDPC encoding
        modSig = qpskMod(encData); % Modulate
        rxSig = awgn(modSig,snrVec(k)); % Pass through AWGN channel
        demodSig = qpskDemod(rxSig,noiseVar); % Demodulate
        rxData = ldpcDec(demodSig); % Decode LDPC
        errorStats = errorCnt(data,rxData); % Compute error stats
    end

    % Save the BER for the current Eb/No and reset the error rate counter
    ber(k) = errorStats(1);
    reset(errorCnt)
end
```

Plot the bit error rate.

```
semilogy(snrVec,ber,'-o')  
grid  
xlabel('SNR (dB)')  
ylabel('Bit Error Rate')
```



Character Representation of Polynomials

You can specify polynomials as a character vector or string using a variety of syntaxes that correspond to those found in literature. The Communications Toolbox functions that support character vector and string polynomials internally convert them into the appropriate form. The form varies depending on the function. For example, the `comm.BCHEncoder` function expresses polynomials as a binary row vector whose powers are in descending order.

When specifying character vectors or strings to represent polynomials:

- The order of the polynomial, ascending or descending, does not matter.
- Spaces are ignored.
- The caret symbol, `^`, which indicates the presence of an exponent, is optional. If omitted, the function assumes that the integer following the variable name is an exponent.
- Braces, `{}`, denote an exponent. For example, you can represent x^2 as `x{2}`.
- Text appearing before the polynomial expression (with or without an equals sign) is ignored.
- Punctuation following square brackets is ignored.
- Exponents must be uniformly positive or uniformly negative. Mixed exponents are not allowed. For example, `'x^2 + x + 1'` and `'1 + z^-6 + z^-8'` are valid while `'1 + z^6 + z^-8'` is not.

To illustrate these characteristics, you can express the polynomial $1 + 2x + x^3 + 4x^5 + x^{14}$ using any of the following examples using single quotes for character vectors as shown or double quotes for strings.

- `'1+2x+x^3+4x^5+x^14'`
- `'1+2m+m3+4m5+m14'`
- `'q14 + 4q5 + q3 + 2q + 1'`
- `'g(x) = 1+2x+x3+4x5+x14'`
- `'g(z) 1+2z+z3+4z5+z14'`
- `'p(x) = x{14} + 4x{5} + x{3} + 2{x} + 1'`
- `'[D14 + 4D5 + D3 + 2D + 1]'`

Estimate BER of 8-PSK in AWGN with Reed-Solomon Coding

Transmit Reed-Solomon encoded data using 8-PSK over an AWGN channel. Demodulate and decode the received signal and collect error statistics. Plot the bit error rate estimate.

Define the example parameters.

```
M = 8;           % Modulation order
bps = log2(M);  % Bits per symbol
N = 7;         % RS codeword length
K = 5;         % RS message length
```

Create modulator, demodulator, AWGN channel, and error rate objects.

```
pskModulator = comm.PSKModulator('ModulationOrder',M,'BitInput',true);
pskDemodulator = comm.PSKDemodulator('ModulationOrder',M,'BitOutput',true);
awgnChannel = comm.AWGNChannel('BitsPerSymbol',bps);
errorRate = comm.ErrorRate;
```

Create a (7,5) Reed-Solomon encoder and decoder pair which accepts bit inputs.

```
rsEncoder = comm.RSEncoder('BitInput',true,'CodewordLength',N,'MessageLength',K);
rsDecoder = comm.RSDecoder('BitInput',true,'CodewordLength',N,'MessageLength',K);
```

Set the range of E_b/N_0 values. Initialize the error statistics matrix.

```
ebnoVec = (3:0.5:8)';
errorStats = zeros(length(ebnoVec),3);
```

Estimate the bit error rate for each E_b/N_0 value. The simulation runs until either 100 errors or 10^7 bits is encountered. The main simulation loop processing includes encoding, modulation, demodulation, and decoding.

```
for i = 1:length(ebnoVec)
    awgnChannel.EbNo = ebnoVec(i);
    reset(errorRate)
    while errorStats(i,2) < 100 && errorStats(i,3) < 1e7
        data = randi([0 1],1500,1);           % Generate binary data
        encData = rsEncoder(data);           % RS encode
        modData = pskModulator(encData);     % Modulate
        rxSig = awgnChannel(modData);       % Pass signal through AWGN
```

```

        rxData = pskDemodulator(rxSig);           % Demodulate
        decData = rsDecoder(rxData);           % RS decode
        errorStats(i,:) = errorRate(data,decData); % Collect error statistics
    end
end

```

Fit a curve to the BER data using `berfit`. Generate an estimate of 8-PSK performance without coding using the `berawgn` function.

```

berCurveFit = berfit(ebnoVec,errorStats(:,1));
berNoCoding = berawgn(ebnoVec,'psk',8,'nondiff');

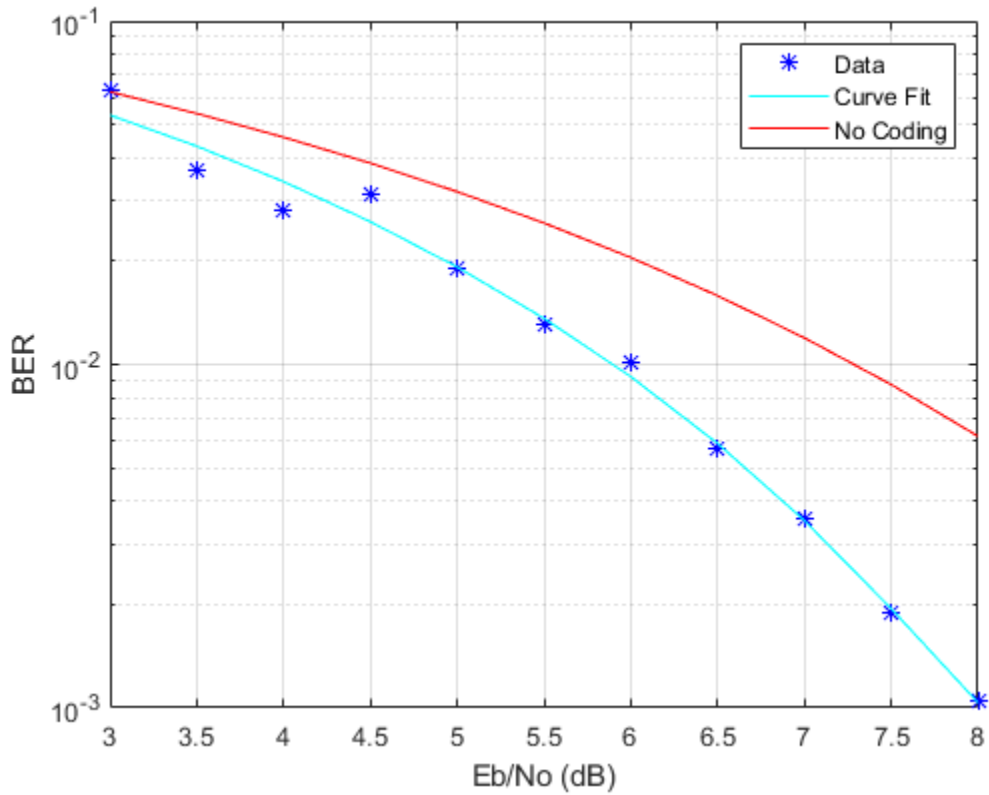
```

Plot the BER data, the BER curve fit, and the estimated performance without RS coding.

```

semilogy(ebnoVec,errorStats(:,1),'b*', ...
ebnoVec,berCurveFit,'c-',ebnoVec,berNoCoding,'r')
ylabel('BER')
xlabel('Eb/No (dB)')
legend('Data','Curve Fit','No Coding')
grid

```



The (7,5) RS code improves the E_b/N_0 required to achieve a 10^{-2} bit error rate by, approximately, 1.4 dB.

Transmit and Receive Shortened Reed-Solomon Codes

Transmit and receive standard and shortened RS-encoded, 64-QAM-modulated data through an AWGN channel. Compare the performance of the standard and shortened codes.

Set the parameters for the Reed-Solomon code, where N is the codeword length, K is the nominal message length, and S is the shortened message length. Specify the modulation order, M .

```
N = 63; % Codeword length
K = 51; % Message length
S = 39; % Shortened message length
M = 64; % Modulation order
```

Specify the simulation parameters, where `numErrors` is the number of errors per Eb/No point, and `numBits` is the maximum number of bits per Eb/No point. Specify the range of Eb/No values to be simulated. Initialize the BER arrays.

```
numErrors = 200;
numBits = 1e7;
ebnoVec = (8:13)';
[ber0,ber1] = deal(zeros(size(ebnoVec)));
```

Create an error rate object to collect error statistics.

```
errorRate = comm.ErrorRate;
```

Create a Reed-Solomon encoder and decoder pair for an RS(63,51) code. Calculate the code rate.

```
rsEncoder = comm.RSEncoder(N,K,'BitInput',true);
rsDecoder = comm.RSDecoder(N,K,'BitInput',true);
rate = K/N;
```

Execute the main processing loop.

```
for k = 1:length(ebnoVec)
    % Convert the coded Eb/No to an SNR. Initialize the error statistics
    % vector.
    snrdb = ebnoVec(k) + 10*log10(rate) + 10*log10(log2(M));
    errorStats = zeros(3,1);
```

```

while errorStats(2) < numErrors && errorStats(3) < numBits

    % Generate binary data.
    txData = randi([0 1],K*log2(M),1);

    % Encode the data.
    encData = rsEncoder(txData);

    % Apply 64-QAM modulation.
    txSig = qammod(encData,M, ...
        'UnitAveragePower',true,'InputType','bit');

    % Pass the signal through an AWGN channel.
    rxSig = awgn(txSig,snrDB);

    % Demodulated the noisy signal.
    demodSig = qamdemod(rxSig,M, ...
        'UnitAveragePower',true,'OutputType','bit');

    % Decode the data.
    rxData = rsDecoder(demodSig);

    % Compute the error statistics.
    errorStats = errorRate(txData,rxData);
end

% Save the BER data, and reset the errorRate counter.
ber0(k) = errorStats(1);
reset(errorRate)
end

```

Create a Reed-Solomon generator polynomial for an RS(63,51) code.

```
gp = rsgenpoly(N,K,[],0);
```

Create a Reed-Solomon encoder and decoder pair using shortened message length S and generator polynomial gp . Calculate the rate of the shortened code.

```
rsEncoder = comm.RSEncoder(N,K,gp,S,'BitInput',true);
rsDecoder = comm.RSDecoder(N,K,gp,S,'BitInput',true);
rate = S/(N-(K-S));
```

Execute the main processing loop using the shortened Reed-Solomon code.

```
for k = 1:length(ebnoVec)
```

```

% Convert the coded Eb/No to an SNR. Initialize the error statistics
% vector.
snrdB = ebnoVec(k) + 10*log10(rate) + 10*log10(log2(M));
errorStats = zeros(3,1);

while errorStats(2) < numErrors && errorStats(3) < numBits

    % Generate binary data.
    txData = randi([0 1],S*log2(M),1);

    % Encode the data.
    encData = rsEncoder(txData);

    % Apply 64-QAM modulation.
    txSig = qammod(encData,M, ...
        'UnitAveragePower',true,'InputType','bit');

    % Pass the signal through an AWGN channel.
    rxSig = awgn(txSig,snrdB);

    % Demodulated the noisy signal.
    demodSig = qamdemod(rxSig,M, ...
        'UnitAveragePower',true,'OutputType','bit');

    % Decode the data.
    rxData = rsDecoder(demodSig);

    % Compute the error statistics.
    errorStats = errorRate(txData,rxData);
end

% Save the BER data, and reset the errorRate counter.
ber1(k) = errorStats(1);
reset(errorRate)
end

```

Calculate the approximate BER for an RS (63,51) code.

```
berapprox = bercoding(ebnoVec, 'RS', 'hard', N, K, 'qam', 64);
```

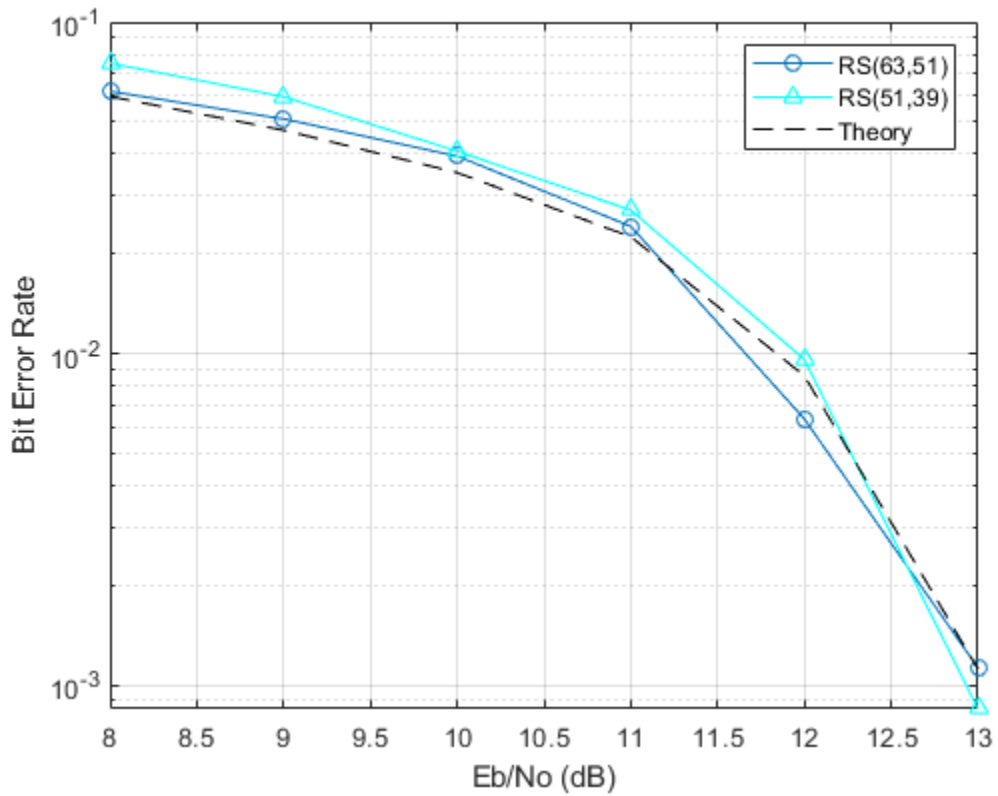
Compare the BER curves for the RS(63,51) and RS(51,39) codes. Plot the theoretically approximated BER curve. Observe that shortening the code does not affect performance.

```
semilogy(ebnoVec,ber0,'o-',ebnoVec,ber1,'c^-',ebnoVec,berapprox,'k--')
legend('RS(63,51)', 'RS(51,39)', 'Theory')
```

```

xlabel('Eb/No (dB)')
ylabel('Bit Error Rate')
grid

```



Galois Fields

- “Working with Galois Fields” on page 12-2
- “ElGamal Public Key Cryptosystem” on page 12-7

Working with Galois Fields

In this section...
“Creating Galois Field Arrays” on page 12-2
“Using Galois Field Arrays” on page 12-2
“Arithmetic in Galois Fields” on page 12-3
“Using MATLAB® Functions with Galois Arrays” on page 12-4
“Hamming Code Example” on page 12-4

This example illustrates how to work with Galois fields.

Galois fields are used in error-control coding, where a Galois field is an algebraic field with a finite number of members. A Galois field that has 2^m members is denoted by $GF(2^m)$, where m is an integer between 1 and 16 in this example.

Creating Galois Field Arrays

You create Galois field arrays using the `GF` function. To create the element 3 in the Galois field 2^2 , you can use the following command:

```
A = gf(3,2)
```

```
A = GF(2^2) array. Primitive polynomial = D^2+D+1 (7 decimal)
```

```
Array elements =
```

```
3
```

Using Galois Field Arrays

You can now use `A` as if it were a built-in MATLAB® data type. For example, here is how you can add two elements in a Galois field together:

```
A = gf(3,2);
```

```
B = gf(1,2);
```

```
C = A+B
```

```
C = GF(2^2) array. Primitive polynomial = D^2+D+1 (7 decimal)
```

Array elements =

2

Arithmetic in Galois Fields

Note that $3 + 1 = 2$ in this Galois field. The rules for arithmetic operations are different for Galois field elements compared to integers. To see some of the differences between Galois field arithmetic and integer arithmetic, first look at an addition table for integers 0 through 3:

+__0__1__2__3

0| 0 1 2 3

1| 1 2 3 4

2| 2 3 4 5

3| 3 4 5 6

You can define such a table in MATLAB with the following commands:

```
A = ones(4,1)*[0 1 2 3];
B = [0 1 2 3]'*ones(1,4);
Table = A+B
```

Table = 4×4

0	1	2	3
1	2	3	4
2	3	4	5
3	4	5	6

Similarly, create an addition table for the field $GF(2^2)$ with the following commands:

```
A = gf(ones(4,1)*[0 1 2 3],2);
B = gf([0 1 2 3]'*ones(1,4),2);
A+B
```

ans = GF(2^2) array. Primitive polynomial = D^2+D+1 (7 decimal)

```
Array elements =
```

```
 0  1  2  3
 1  0  3  2
 2  3  0  1
 3  2  1  0
```

Using MATLAB® Functions with Galois Arrays

Many other MATLAB functions will work with Galois arrays. To see this, first create a couple of arrays.

```
A = gf([1 33],8);
B = gf([1 55],8);
```

Now you can multiply two polynomials.

```
C = conv(A,B)
```

```
C = GF(2^8) array. Primitive polynomial = D^8+D^4+D^3+D^2+1 (285 decimal)
```

```
Array elements =
```

```
 1  22  153
```

You can also find roots of a polynomial. (Note that they match the original values in A and B.)

```
roots(C)
```

```
ans = GF(2^8) array. Primitive polynomial = D^8+D^4+D^3+D^2+1 (285 decimal)
```

```
Array elements =
```

```
 33
 55
```

Hamming Code Example

The most important application of Galois field theory is in error-control coding. The rest of this example uses a simple error-control code, a Hamming code. An error-control code

works by adding redundancy to information bits. For example, a (7,4) Hamming code maps 4 bits of information to 7-bit codewords. It does this by multiplying the 4-bit codeword by a 4 x 7 matrix. You can obtain this matrix with the HAMMGEN function:

```
[H,G] = hammgen(3)
```

```
H = 3x7
```

```

    1    0    0    1    0    1    1
    0    1    0    1    1    1    0
    0    0    1    0    1    1    1

```

```
G = 4x7
```

```

    1    1    0    1    0    0    0
    0    1    1    0    1    0    0
    1    1    1    0    0    1    0
    1    0    1    0    0    0    1

```

H is the parity-check matrix and G is the generator matrix. To encode the information bits [0 1 0 0], multiply the information bits [0 1 0 0] by the generator matrix G:

```
A = gf([0 1 0 0],1)
```

```
A = GF(2) array.
```

```
Array elements =
```

```

    0    1    0    0

```

```
Code = A*G
```

```
Code = GF(2) array.
```

```
Array elements =
```

```

    0    1    1    0    1    0    0

```

Suppose somewhere along transmission, an error is introduced into this codeword. (Note that a Hamming code can correct only 1 error.)

```
Code(1) = 1    % Place a 1 where there should be a 0.
```

```
Code = GF(2) array.
```

```
Array elements =
```

```
1 1 1 0 1 0 0
```

You can use the parity-check matrix H to determine where the error occurred, by multiplying the codeword by H:

```
H*Code'
```

```
ans = GF(2) array.
```

```
Array elements =
```

```
1  
0  
0
```

To find the error, look at the parity-check matrix H. The column in H that matches [1 0 0]' is the location of the error. Looking at H, you can see that the first column is [1 0 0]'. This means that the first element of the vector Code contains the error.

H

```
H = 3x7
```

```
1 0 0 1 0 1 1  
0 1 0 1 1 1 0  
0 0 1 0 1 1 1
```

ElGamal Public Key Cryptosystem

Use the Galois field array function, `gf`, to implement an ElGamal public key cryptosystem.

Key Generation

Define the polynomial degree, m .

```
m = 15;
q = 2^m;
```

Find a primitive polynomial and group generator. Set the random number generator seed to produce a repeatable result.

```
poly = primpoly(m, 'nodisplay');

primeFactors = unique(factor(2^m-1));
rng(123456);
while 1
    g = gf(randi([1,q-1]),m,poly);
    isPrimitive = true;
    for i = 1:length(primeFactors)
        if g^((q-1)/primeFactors(i)) == gf(1,m,poly)
            isPrimitive = false;
            break;
        end
    end
    if isPrimitive
        break;
    end
end
```

Construct private and public keys.

```
privateKey = 12;
publicKey = {g,g^privateKey,poly};
```

Encryption

Create and display the original message.

```
text = ['The Fourier transform decomposes a function of time (a signal)' newline ...
        'into the frequencies that make it up, in a way similar to how a' newline ...
        'musical chord can be expressed as the amplitude (or loudness) of' newline ...
```

```
    'its constituent notes.'];  
disp(text);
```

The Fourier transform decomposes a function of time (a signal) into the frequencies that make it up, in a way similar to how a musical chord can be expressed as the amplitude (or loudness) of its constituent notes.

Convert the message to binary and group them every m bits. The message uses ASCII characters. Since the ASCII table has 128 characters, seven bits per character is sufficient.

```
bitsPerChar = 7;  
binMsg = de2bi(int8(text),bitsPerChar,'left-msb');  
numPaddedBits = m - mod(numel(binMsg),m);  
if numPaddedBits == m  
    numPaddedBits = 0;  
end  
binMsg = [binMsg(:); zeros(numPaddedBits,1)];  
textToEncrypt = bi2de(reshape(binMsg,[],m),'left-msb');
```

Encrypt the original message.

```
cipherText = gf(zeros(length(textToEncrypt),2),m,poly);  
  
for i = 1:length(textToEncrypt)  
    k = randi([1 2^m-2]);  
    cipherText(i,:) = [publicKey{1}^k, ...  
        gf(textToEncrypt(i),m,poly)*publicKey{2}^k];  
end
```

Display the encrypted message.

```
tmp = cipherText.x;  
disp(de2char(tmp(:,2),bitsPerChar,m));
```

```
Y7cyg^]0T9*3a~sCoxEwSL}LWpxcWPX@/C$#Y`f&IM8i?&\ODRn/91D  
aQ @LZM5`hGpR'\N>6/cfa&nXF1:P%uwmGp6j!]%"&/LTHX$-q0>]JTMdRT`K(p\ CQ>e(HNn9358M,9\VDtMV  
9S9k[#0'Fs=:p
```

Decryption

Decrypt the encrypted original message.

```
decipherText = gf(zeros(size(cipherText,1),1),m,poly);  
for i = 1:size(cipherText,1)
```



```
    decipherText(i) = cipherText(i,2) * cipherText(i,1)^(-privateKey);
end
```

Display the decrypted message.

```
disp(de2char(decipherText.x, bitsPerChar, m));
```

The Fourier transform decomposes a function of time (a signal) into the frequencies that make it up, in a way similar to how a musical chord can be expressed as the amplitude (or loudness) of its constituent notes.

Supporting Function

de2char converts the bits to char messages.

```
function text = de2char(msg, bitsPerChar, m)
binDecipherText = de2bi(msg, m, 'left-msb');
text = char(bi2de(reshape(binDecipherText(1:end-mod(numel(binDecipherText), ...
    bitsPerChar)), bitsPerChar, []), 'left-msb'))';
end
```


Convolutional Coding

- “Punctured Convolutional Coding” on page 13-2
- “Iterative Decoding of a Serially Concatenated Convolutional Code” on page 13-8
- “Punctured Convolutional Encoding” on page 13-15
- “Rate 2/3 Convolutional Code in AWGN” on page 13-22
- “Estimate BER for Hard and Soft Decision Viterbi Decoding” on page 13-25
- “Creation, Validation, and Testing of User Defined Trellis Structure” on page 13-29

Punctured Convolutional Coding

This example shows how to use the convolutional encoder and Viterbi decoder System objects to simulate a punctured coding system. The complexity of a Viterbi decoder increases rapidly with the code rate. Puncturing is a technique that allows the encoding and decoding of higher rate codes using standard rate 1/2 encoders and decoders.

Introduction

This example showcases the simulation of a communication system consisting of a random binary source, a convolutional encoder, a BPSK modulator, an additive white Gaussian noise (AWGN) channel, and a Viterbi decoder. The example shows how to run simulations to obtain bit error rate (BER) curves and compares these curves to a theoretical bound.

Initialization

Convolutional Encoding with Puncturing

Create a rate 1/2, constraint length 7 ConvolutionalEncoder System object. This encoder takes one-bit symbols as inputs and generates 2-bit symbols as outputs. If you assume 3-bit message words as inputs, then the encoder will generate 6-bit codeword outputs.

```
convEncoder = comm.ConvolutionalEncoder(poly2trellis(7, [171 133]));
```

Specify a puncture pattern to create a rate 3/4 code from the previous rate 1/2 code using the puncture pattern vector [1;1;0;1;1;0]. The ones in the puncture pattern vector indicate that bits in positions 1, 2, 4, and 5 are transmitted, while the zeros indicate that bits in positions 3 and 6 are punctured or removed from the transmitted signal. The effect of puncturing is that now, for every 3 bits of input, the punctured code generates 4 bits of output (as opposed to the 6 bits produced before puncturing). This results in a rate 3/4 code. In the example at hand, the length of the puncture pattern vector must be an integer multiple of 6 since 3-bit inputs get converted into 6-bit outputs by the rate 1/2 convolutional encoder.

To set the desired puncture pattern in the convolutional encoder System object, hConvEnc, set the PuncturePatternSource property to Property and the PuncturePattern property to [1;1;0;1;1;0].

```
convEncoder.PuncturePatternSource = 'Property';  
convEncoder.PuncturePattern = [1;1;0;1;1;0];
```

Modulator and Channel

Create a `BPSKModulator System` object to transmit the encoded data using binary phase shift keying modulation over a channel.

```
bpskMod = comm.BPSKModulator;
```

Create an `AWGNChannel System` object. Set the `NoiseMethod` property of the channel to `Signal to noise ratio (Eb/No)` to specify the noise level using the energy per bit to noise power spectral density ratio (E_b/N_0). When running simulations, test the coding system for different values of E_b/N_0 ratio by changing the `EbNo` property of the channel object. The output of the BPSK modulator generates unit power signals; set the `SignalPower` property to 1 Watt. The system at hand is at the symbol rate; set the `SamplesPerSymbol` property to 1.

```
channel = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (Eb/No)', ...
    'SignalPower', 1, 'SamplesPerSymbol', 1);
```

Viterbi Decoding with Depuncturing

Configure a `ViterbiDecoder System` object so it decodes the punctured code specified for the convolutional encoder. This example assumes unquantized inputs to the Viterbi decoder, so set the `InputFormat` property to `Unquantized`.

```
vitDecoder = comm.ViterbiDecoder(poly2trellis(7, [171 133]), ...
    'InputFormat', 'Unquantized');
```

In general, the puncture pattern vectors you use for the convolutional encoder and Viterbi decoder must be the same. To specify the puncture pattern, set the `PuncturePatternSource` property of the Viterbi decoder System object, `hVitDec`, to `Property`. Set the `PuncturePattern` property to the same puncture pattern vector you use for the convolutional encoder.

Because the punctured bits are not transmitted, there is no information to indicate their values. As a result, the decoding process ignores them.

```
vitDecoder.PuncturePatternSource = 'Property';
vitDecoder.PuncturePattern = convEncoder.PuncturePattern;
```

For a rate $1/2$ code with no puncturing, you normally set the traceback depth of a Viterbi decoder to a value close to 40. Decoding punctured codes requires a higher value, in order to give the decoder enough data to resolve the ambiguities that the punctures introduce. This example uses a traceback depth of 96. Set this value using the `TraceBackDepth` property of the Viterbi decoder object, `hVitDec`.

```
vitDecoder.TracebackDepth = 96;
```

Calculating the Error Rate

Create an ErrorRate calculator System object to compare decoded bits to the original transmitted bits. The output of the error rate calculator object is a three-element vector containing the calculated bit error rate (BER), the number of errors observed, and the number of bits processed. The Viterbi decoder creates a delay in the output decoded bit stream equal to the traceback length. To account for this delay set the ReceiveDelay property of the error rate calculator System object to 96.

```
errorCalc = comm.ErrorRate('ReceiveDelay', vitDecoder.TracebackDepth);
```

Stream Processing Loop

Analyze the BER performance of the punctured coding system for different noise levels.

Uncoded and Coded Eb/No Ratio Values

Typically, you measure system performance according to the value of the energy per bit to noise power spectral density ratio (Eb/No) available at the input of the channel encoder. The reason for this is that this quantity is directly controlled by the systems engineer. Analyze the performance of the coding system for Eb/No values between 2 and 5 dB.

```
EbNoEncoderInput = 2:0.5:5; % in dB
```

The signal going into the AWGN channel is the encoded signal. Convert the Eb/No values so that they correspond to the energy ratio at the encoder output. If you input three bits to the encoder and obtain four bit outputs, then the energy relation is given by the 3/4 rate as follows:

```
EbNoEncoderOutput = EbNoEncoderInput + 10*log10(3/4);
```

Simulation loop

To obtain BER performance results, transmit frames of 3000 bits through the communications system. For each Eb/No value, stop simulations upon reaching a specific number of errors or transmissions. To improve the accuracy of the results, increase the target number of errors or the maximum number of transmissions.

```
frameLength = 3000;           % this value must be an integer multiple of 3  
targetErrors = 300;  
maxNumTransmissions = 5e6;
```

Loop through the encoded Eb/No values (the simulation will take a few seconds to complete).

```
BERVec = zeros(3,length(EbNoEncoderOutput)); % Allocate memory to store results
for n=1:length(EbNoEncoderOutput)
    reset(errorCalc)
    reset(convEncoder)
    reset(vitDecoder)
    channel.EbNo = EbNoEncoderOutput(n); % Set the channel EbNo value for simulation
    while (BERVec(2,n) < targetErrors) && (BERVec(3,n) < maxNumTransmissions)
        % Generate binary frames of size specified by the frameLength variable
        data = randi([0 1], frameLength, 1);
        % Convolutionally encode the data
        encData = convEncoder(data);
        % Modulate the encoded data
        modData = bpskMod(encData);
        % Pass the modulated signal through an AWGN channel
        channelOutput = channel(modData);
        % Pass the real part of the channel complex outputs as the unquantized
        % input to the Viterbi decoder.
        decData = vitDecoder(real(channelOutput));
        % Compute and accumulate errors
        BERVec(:,n) = errorCalc(data, decData);
    end
end
```

Compare Results to Theoretical Curves

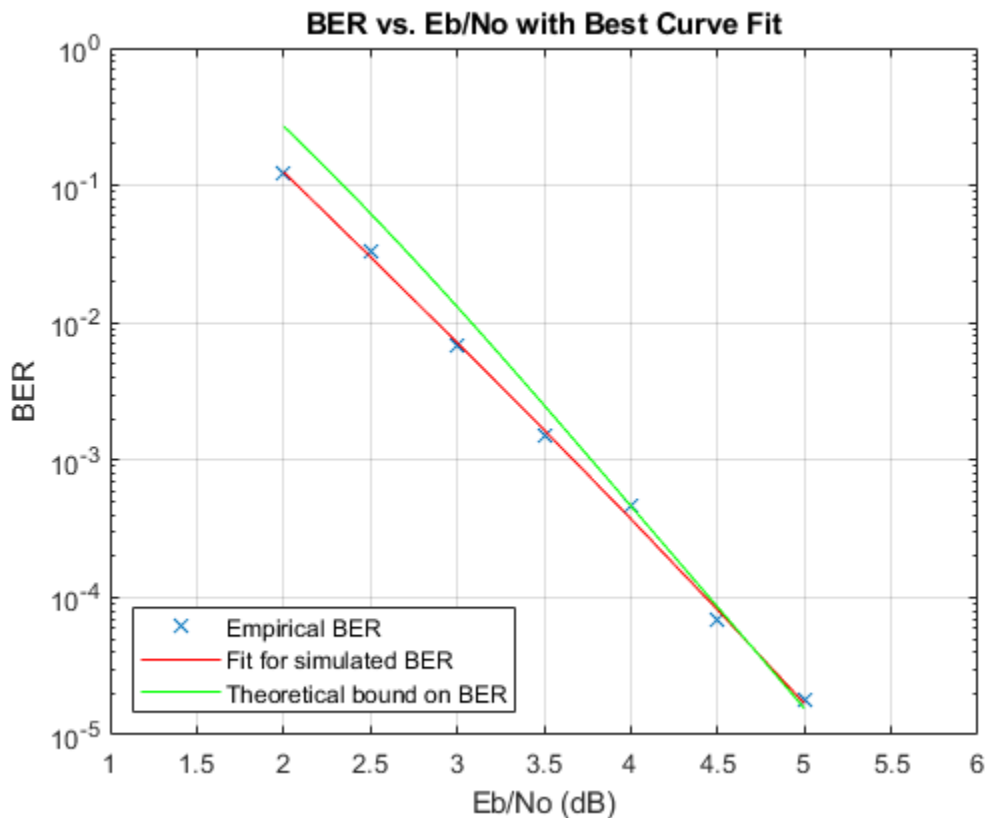
We compare the simulation results using an approximation of the bit error probability bound for a punctured code as per [1]. The following commands compute an approximation of this bound using the first seven terms of the summation for Eb/No values in 2:0.5:5. The values used for nerr are found in Table 2 of [2].

```
dist = 5:11;
nerr = [42 201 1492 10469 62935 379644 2253373];
codeRate = 3/4;
bound = nerr*(1/6)*erfc(sqrt(codeRate*(10.0.^(2:.02:5)/10))*dist)';
```

Plot results. If the target number of errors or maximum number of transmissions you specify for the simulation are too small, the curve fitting algorithm might fail.

```
berfit(EbNoEncoderInput,BERVec(1,:)); % Curve-fitted simulation results
hold on;
semilogy(2:.02:5,bound,'g'); % Theoretical results
```

```
legend('Empirical BER', 'Fit for simulated BER', 'Theoretical bound on BER')  
axis([1 6 10^-5 1])
```



In some cases, at lower bit error rates, simulation results appear to indicate error rates slightly above the bound. This results from simulation variance (if fewer than 500 bit errors are observed) or from the finite traceback depth in the decoder.

Summary

We utilized several System objects to simulate a communications system with convolutional coding and puncturing. We simulated the system to obtain BER performance versus different Eb/No ratio values. The BER results were compared to theoretical bounds.

Selected Bibliography

- 1 Yasuda, Y., K. Kashiki, and Y. Hirata, "High Rate Punctured Convolutional Codes for Soft Decision Viterbi Decoding," IEEE® Transactions on Communications, Vol. COM-32, March, 1984, pp. 315-319
- 2 Begin, G., Haccoun, D., and Paquin, C., "Further results on High-Rate Punctured Convolutional Codes for Viterbi and Sequential Decoding," IEEE Transactions on Communications, Vol. 38, No. 11, November, 1990, p. 1923

Iterative Decoding of a Serially Concatenated Convolutional Code

In this section...
"Exploring the Example" on page 13-8
"Variables in the Example" on page 13-9
"Creating a Serially Concatenated Code" on page 13-10
"Convolutional Encoding Details" on page 13-10
"Decoding Using an Iterative Process" on page 13-11
"Computations in Each Iteration" on page 13-11
"Results of the Iterative Loop" on page 13-12
"Results and Displays" on page 13-12

This model shows how to use an iterative process to decode a serially concatenated convolutional code (SCCC).

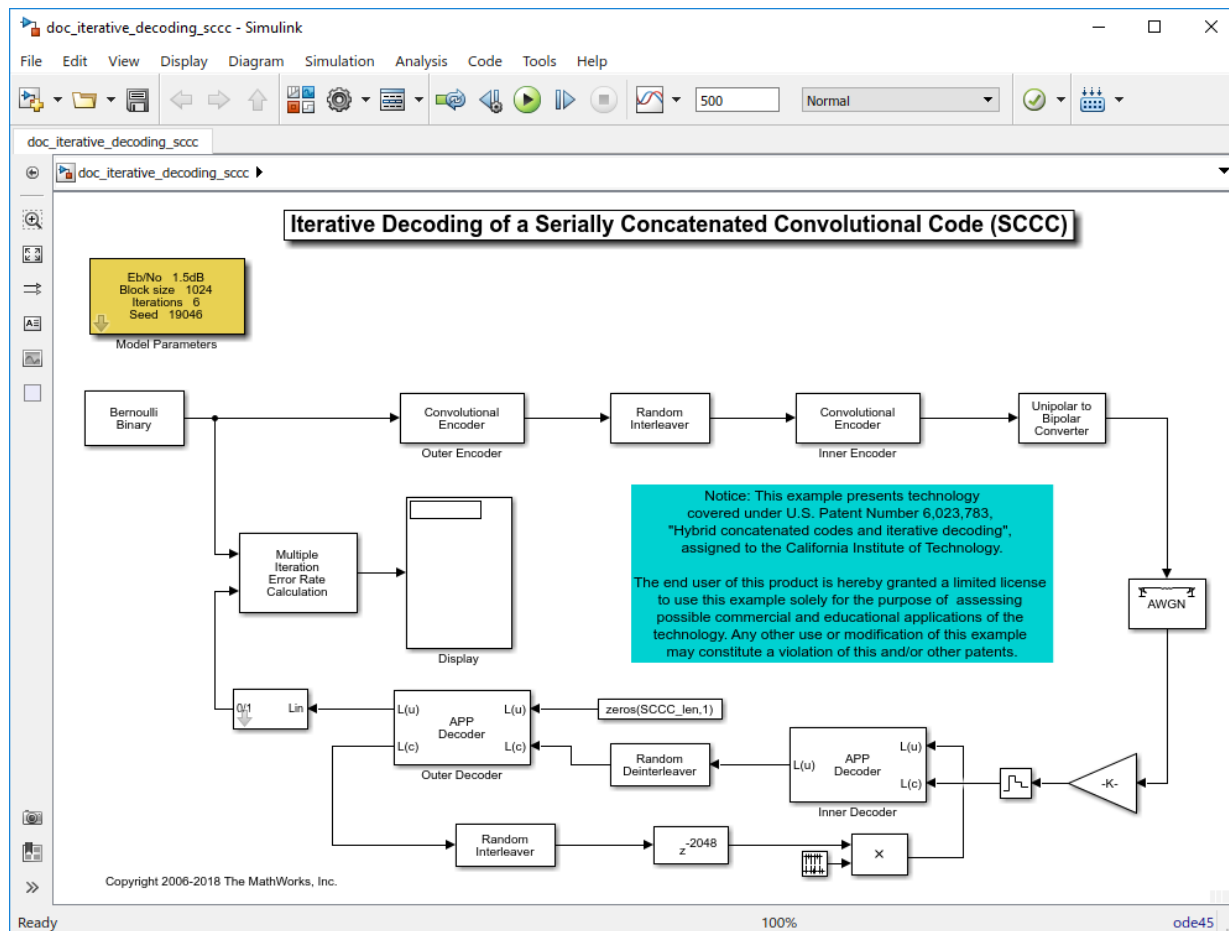
Note This example presents technology covered under U.S. Patent Number 6,023,783, "Hybrid concatenated codes and iterative decoding," assigned to the California Institute of Technology. The end user of this product is hereby granted a limited license to use this example solely for the purpose of assessing possible commercial and educational applications of the technology. Any other use or modification of this example may constitute a violation of this and/or other patents.

Exploring the Example

The simulation generates information bits, encodes them using a serially concatenated convolutional code, and transmits the coded information along a noisy channel. The simulation then decodes the received coded information, using an iterative decoding process, and computes error statistics based on different numbers of iterations. Throughout the simulation, the error rates appear in a Display block.

Open the model, `doc_iterative_decoding_sccc`, by entering the following on the MATLAB command line.

```
doc_iterative_decoding_sccc
```



Variables in the Example

The Model Parameters block lets you vary the values of some quantities that the model uses. The table below indicates their names and meanings.

Name	Meaning
E_b/N₀	E_b/N_0 in channel noise, measured in dB; used to compute the variance of the channel noise
Block size	The number of bits in each frame of uncoded data
Number of iterations	The number of iterations to use when decoding
Seed	The initial seed in the Random Interleaver and Random Deinterleaver blocks

Creating a Serially Concatenated Code

The encoding portion of the example uses a Convolutional Encoder block to encode a data frame, a Random Interleaver block to shuffle the bits in the codewords, and another Convolutional Encoder block to encode the interleaved bits. Because these blocks are connected in series with each other, the resulting code is called a serially concatenated code.

Together, these blocks encode the 1024-bit data frame into a 3072-bit frame representing a concatenated code. These sizes depend on the model's **Block size** parameter (see the Model Parameters block). The code rate of the concatenated code is 1/3.

In general, the purpose of interleaving is to protect codewords from burst errors in a noisy channel. A burst error that corrupts interleaved data actually has a small effect on each of several codewords, rather than a large effect on any one codeword. The smaller the error in an individual codeword, the greater the chance that the decoder can recover the information correctly.

Convolutional Encoding Details

The two instances of the Convolutional Encoder block use their **Trellis structure** parameters to specify the convolutional codes. The table below lists the polynomials that define each of the two convolutional codes. The second encoder has two inputs and uses two rows of memory registers.

	Outer Convolutional Code	Inner Convolutional Code
Generator Polynomials	$1+D+D^2$ and $1+D^2$	First row: $1+D+D^2$, 0, and $1+D^2$
		Second row: 0, $1+D+D^2$, and $1+D$
Feedback Polynomials	$1+D+D^2$	$1+D+D^2$ for each row
Constraint Lengths	3	3 for each row
Code rate	$1/2$	$2/3$

Decoding Using an Iterative Process

The decoding portion of this example consists of two APP Decoder blocks, a Random Deinterleaver block, and several other blocks. Together, these blocks form a loop and operate at a rate six times that of the encoding portion of the example. The loop structure and higher rate combine to make the decoding portion an iterative process. Using multiple iterations improves the decoding performance. You can control the number of iterations by setting the **Number of iterations** parameter in the model's Model Parameters block. The default number of iterations is six.

Computations in Each Iteration

In each iteration, the decoding portion of the example decodes the inner convolutional code, deinterleaves the result, and decodes the outer convolutional code. The outer decoder's $L(u)$ output signal represents the updated likelihoods of original message bits (that is, input bits to the outer encoder).

The looping strategy in this example enables the inner decoder to benefit in the next iteration from the outer decoder's work. To understand how the loop works, first recall the meanings of these signals:

- The outer decoder's $L(c)$ output signal represents the updated likelihoods of code bits from the outer encoder.
- The inner decoder's $L(u)$ input represents the likelihoods of input bits to the inner encoder.

The feedback loop recognizes that the primary distinction between these two signals is in the interleaving operation that occurs between the outer and inner encoders. Therefore,

the loop interleaves the $L(c)$ output of the outer decoder to replicate that interleaving operation, delays the interleaved data to ensure that the inner decoder's two input ports represent data from the same time steps, and resets the $L(u)$ input to the inner decoder to zero after every six iterations.

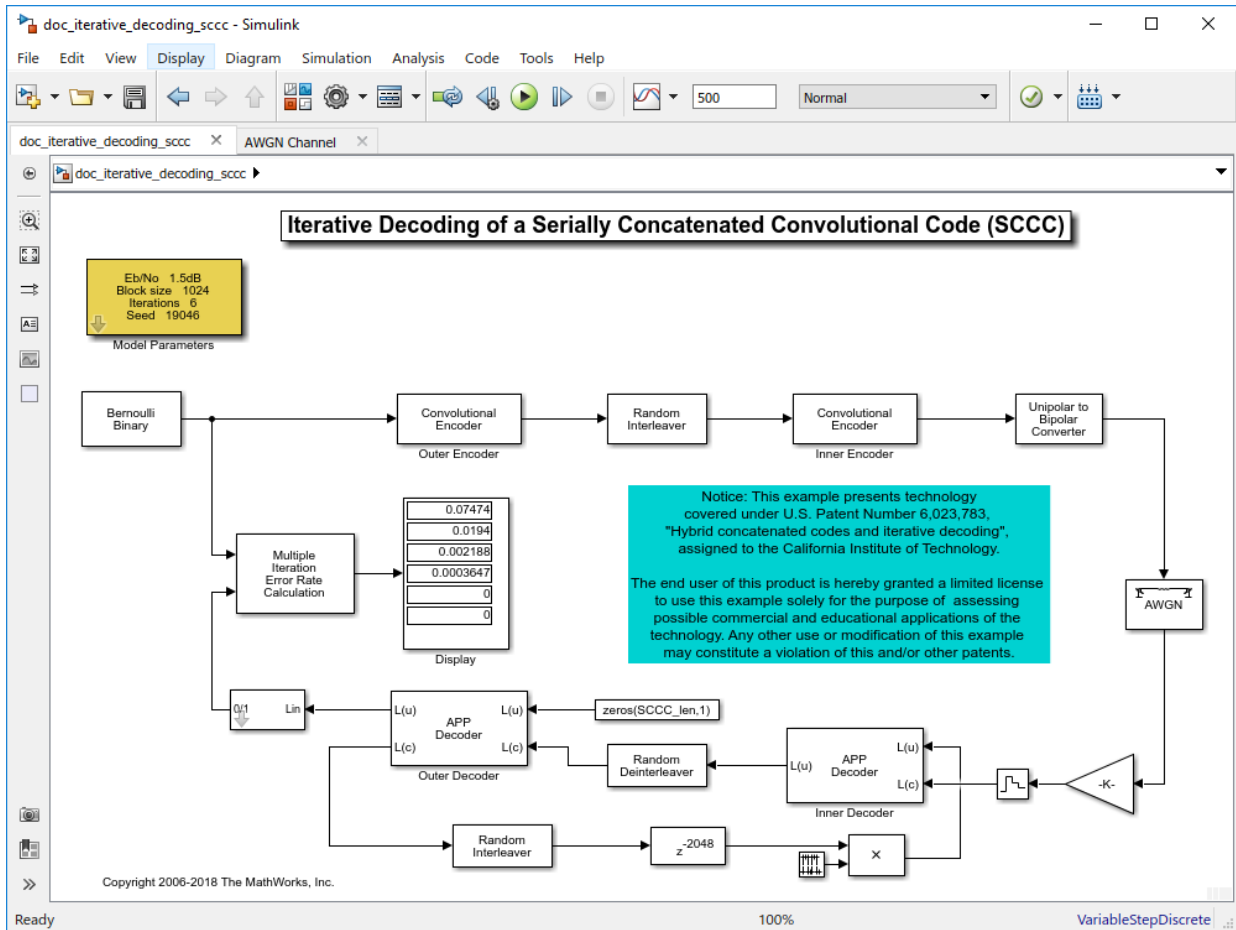
Results of the Iterative Loop

The result of decoding is a 1024-element frame whose elements indicate the likelihood that each of the 1024 message bits was a 0 or a 1. A nonnegative element indicates that the message bit was probably a 1, and a negative element indicates that the message bit was probably a 0. The Hard Decision block converts nonnegative and negative values to 1's and 0's, respectively, so that the results have the same form as the original uncoded binary data.

Results and Displays

The example includes a large Display block that shows error rates after comparing the received data with the transmitted data. The number of error rates in the display is the number of iterations in the decoding process. The first error rate reflects the performance of a decoding process that uses one iteration, the second error rate reflects the performance of a decoding process that uses two iterations, and so on. The series of error rates shows that the error rate generally decreases as the number of iterations increases.

Change the **Eb/No** to 1 dB and run the simulation. Observe that the bit error rates decrease with each iteration.



References

- [1] Benedetto, S., D. Divsalar, G. Montorsi, and F. Pollara, "Serial Concatenation of Interleaved Codes: Performance Analysis, Design, and Iterative Decoding," JPL TDA Progress Report, Vol. 42-126, August 1996.
- [2] Divsalar, Dariush, and Fabrizio Pollara, Hybrid Concatenated Codes and Iterative Decoding, U. S. Patent No. 6,023,783, Feb. 8, 2000.

[3] Heegard, Chris, and Stephen B. Wicker, Turbo Coding, Boston, Kluwer Academic Publishers, 1999.

Punctured Convolutional Encoding

In this section...

“Structure of the Example” on page 13-15
“Generating Random Data” on page 13-16
“Convolutional Encoding with Puncturing” on page 13-16
“Transmitting Data” on page 13-17
“Demodulating” on page 13-17
“Viterbi Decoding of Punctured Codes” on page 13-17
“Calculating the Error Rate” on page 13-18
“Evaluating Results” on page 13-18

This model shows how to use the Convolutional Encoder and Viterbi Decoder blocks to simulate a punctured coding system. The complexity of a Viterbi decoder increases rapidly with the code rate. Puncturing is a technique that allows the encoding and decoding of higher rate codes using standard rate 1/2 encoders and decoders.

The example is somewhat similar to the one that appears in “Soft-Decision Decoding” on page 8-62, which shows convolutional coding without puncturing.

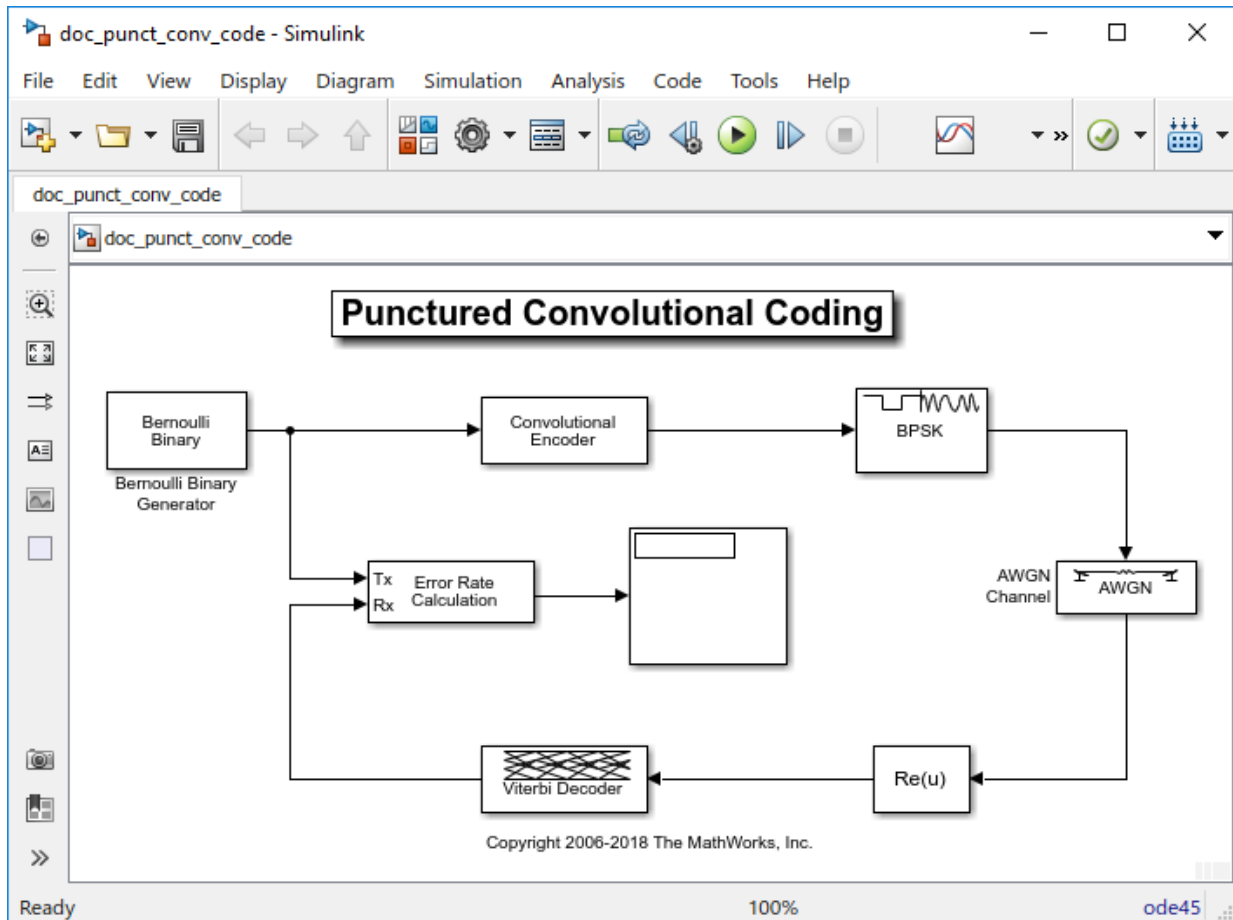
Structure of the Example

This example contains these blocks.

- Bernoulli Binary Generator: Create a sequence of random bits to use as a message.
- Convolutional Encoder: Encode the message using the convolutional encoder.
- BPSK Modulator Baseband: Modulate the encoded message.
- AWGN Channel: Pass the modulated signal through a noisy channel.
- Error Rate Calculation: Compute the number of discrepancies between the original and recovered messages.

Open the example, `doc_punct_conv_code`, by entering the following at the MATLAB command prompt.

```
doc_punct_conv_code
```



Generating Random Data

The Bernoulli Binary Generator block produces the information source for this simulation. The block generates a frame of three random bits at each sample time. The **Samples per frame** parameter determines the number of rows of the output frame.

Convolutional Encoding with Puncturing

The Convolutional Encoder block encodes the data from the Bernoulli Binary Generator. This example uses the same code as described in "Soft-Decision Decoding" on page 8-62.

The puncture pattern is specified by the **Puncture vector** parameter in the mask. The puncture vector is a binary column vector. A 1 indicates that the bit in the corresponding position of the input vector is sent to the output vector, while a 0 indicates that the bit is removed.

For example, to create a rate 3/4 code from the rate 1/2, constraint length 7 convolutional code, the optimal puncture vector is [1 1 0 1 1 0].' (where the .' after the vector indicates the transpose). Bits in positions 1, 2, 4, and 5 are transmitted, while bits in positions 3 and 6 are removed. Now, for every 3 bits of input, the punctured code generates 4 bits of output (as opposed to the 6 bits produced before puncturing). This makes the rate 3/4.

In this example, the output from the Bernoulli Binary Generator is a column vector of length 3. Because the rate 1/2 Convolutional Encoder doubles the length of each vector, the length of the puncture vector must divide 6.

Transmitting Data

The AWGN Channel block simulates transmission over a noisy channel. The parameters for the block are set in the mask as follows:

- The **Mode** parameter for this block is set to **Signal to noise ratio (Es/No)**.
- The **Es/No** parameter is set to 2 dB. This value typically is changed from one simulation run to the next.
- The preceding modulation block generates unit power signals so the **Input signal power** is set to 1 Watt.
- The **Symbol period** is set to 0.75 seconds because the code has rate 3/4.

Demodulating

In this simulation, the Viterbi Decoder block is set to accept unquantized inputs. As a result, the simulation passes the channel output through a Simulink Complex to Real-Imag block that extracts the real part of the complex samples.

Viterbi Decoding of Punctured Codes

The Viterbi Decoder block is configured to decode the same rate 1/2 code specified in the Convolutional Encoder block.

In this example, the decision type is set to **Unquantized**. For codes without puncturing, you would normally set the **Traceback depth** for this code to a value close to 40.

However, for decoding punctured codes, a higher value is required to give the decoder enough data to resolve the ambiguities introduced by the punctures.

Since the punctured bits are not transmitted, there is no information to indicate their values. As a result they are ignored in the decoding process.

The **Puncture vector** parameter indicates the locations of the punctures or the bits to ignore in the decoding process. Each 1 in the puncture vector indicates a transmitted bit while each 0 indicates a puncture or the bit to ignore in the input to the decoder.

In general, the two **Puncture vector** parameters in the Convolutional Encoder and Viterbi Decoder must be the same.

Calculating the Error Rate

The Error Rate Calculation block compares the decoded bits to the original source bits. The output of the Error Rate Calculation block is a three-element vector containing the calculated bit error rate (BER), the number of errors observed, and the number of bits processed.

In the mask for this block, the **Receive delay** parameter is set to 96, because the **Traceback depth** value of 96 in the Viterbi Decoder block creates a delay of 96. If there were other blocks in the model that created delays, the **Receive delay** would equal the sum of all the delays.

BER simulations typically run until a minimum number of errors have occurred, or until the simulation processes a maximum number of bits. The Error Rate Calculation block uses its **Stop simulation** mode to set these limits and to control the duration of the simulation.

Evaluating Results

Generating a bit error rate curve requires multiple simulations. You can perform multiple simulations using the `sim` command. Follow these steps:

- In the model window, remove the Display block and the line connected to its port.
- In the AWGN Channel block, set the **Es/No** parameter to the variable name `EsNodB`.
- In the Error Rate Calculation block, set **Output data** to `Workspace` and then set **Variable name** to `BER_Data`.

- Save the model in your working directory under a different name, such as `my_punct_conv_code.slx`.
- Execute the following code, which runs the simulation multiple times and gathers results.

```
CodeRate = 0.75;
EbNoVec = [2:.5:5];
EsNoVec = EbNoVec + 10*log10(CodeRate);
BERVec = zeros(length(EsNoVec),3);
for n=1:length(EsNoVec),
    EsNodB = EsNoVec(n);
    sim('my_commpunctcnvcod');
    BERVec(n,:) = BER_Data;
end
```

To confirm the validity of the results, compare them to an established performance bound. The bit error rate performance of a rate $r = (n-1)/n$ punctured code is bounded above by the expression:

$$P_b \leq \frac{1}{2(n-1)} \sum_{d=d_{free}}^{\infty} \omega_d \operatorname{erfc}(\sqrt{rd(E_b/N_0)})$$

In this expression, erfc denotes the complementary error function, r is the code rate, and both d_{free} and ω_d are dependent on the particular code. For the rate 3/4 code of this example, $d_{free} = 5$, $\omega_5 = 42$, $\omega_6 = 201$, $\omega_7 = 1492$, and so on. See reference [1] for more details.

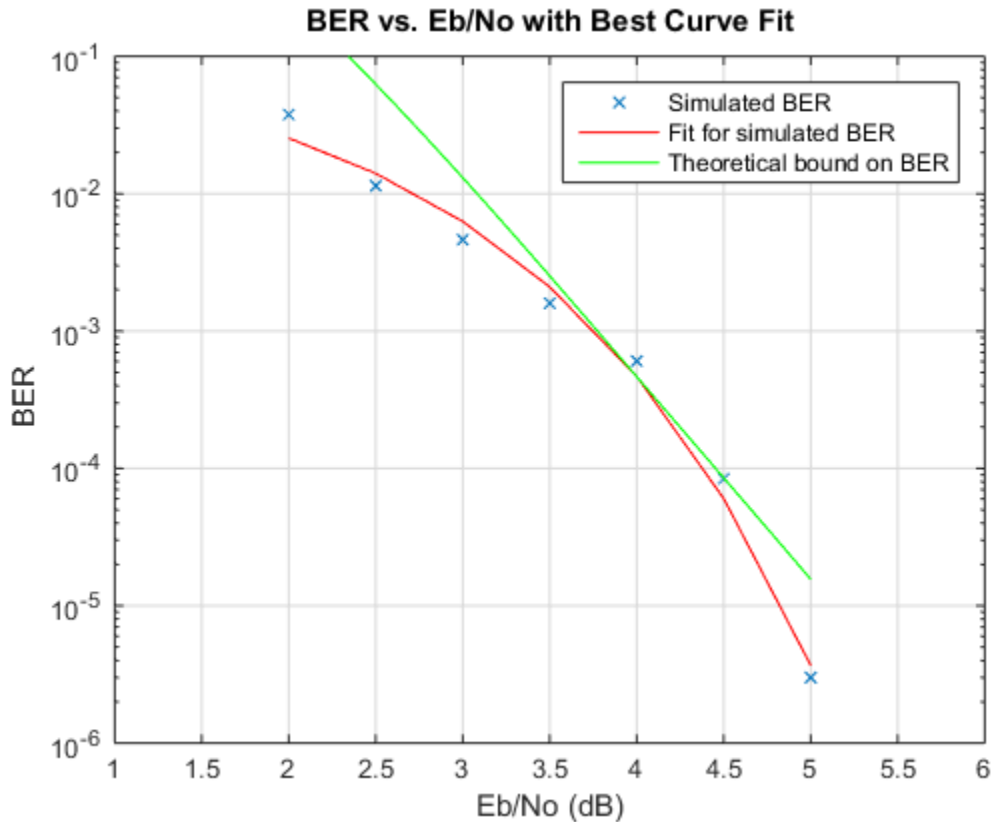
The following commands compute an approximation of this bound using the first seven terms of the summation (the values used for `nerr` are found in Table 2 of reference [2]):

```
dist = [5:11];
nerr = [42 201 1492 10469 62935 379644 2253373];
CodeRate = 3/4;
EbNo_dB = [2:.02:5];
EbNo = 10.0.^(EbNo_dB/10);
arg = sqrt(CodeRate*EbNo'*dist);
bound = nerr*(1/6)*erfc(arg)';
```

To plot the simulation and theoretical results in the same figure, use the commands below.

```
berfit(EbNoVec',BERVec(:,1)); % Curve-fitted simulation results
hold on;
```

```
semilogy(EbNo_dB,bound,'g'); % Theoretical results
legend('Simulated BER','Fit for simulated BER',...
       'Theoretical bound on BER')
```



In some cases, at the lower bit error rates, you might notice simulation results that appear to indicate error rates slightly above the bound. This can result from simulation variance (if fewer than 500 bit errors are observed) or from the finite traceback depth in the decoder.

References

- [1] Yasuda, Y., K. Kashiki, and Y. Hirata, "High Rate Punctured Convolutional Codes for Soft Decision Viterbi Decoding," IEEE Transactions on Communications, Vol. COM-32, March, 1984, pp. 315-319.
- [2] Begin, G., Haccoun, D., and Paquin, C., "Further results on High-Rate Punctured Convolutional Codes for Viterbi and Sequential Decoding," IEEE Transactions on Communications, Vol. 38, No. 11, November, 1990, p. 1923.

Rate 2/3 Convolutional Code in AWGN

This example generates a bit error rate versus E_b/N_0 curve for a link that uses 16-QAM modulation and a rate 2/3 convolutional code in AWGN.

Set the modulation order, and compute the number of bits per symbol.

```
M = 16;  
k = log2(M);
```

Create a trellis for a rate 2/3 convolutional code. Set the traceback and code rate parameters.

```
trellis = poly2trellis([5 4],[23 35 0; 0 5 13]);  
traceBack = 16;  
codeRate = 2/3;
```

Create a convolutional encoder and its equivalent Viterbi decoder.

```
convEncoder = comm.ConvolutionalEncoder('TrellisStructure',trellis);  
vitDecoder = comm.ViterbiDecoder('TrellisStructure',trellis, ...  
    'InputFormat','Hard','TracebackDepth',traceBack);
```

Create a 16-QAM modulator and demodulator pair having bit inputs and outputs. Set the constellation to normalize on average power.

```
qamModulator = comm.RectangularQAMModulator('BitInput',true,'NormalizationMethod','AveragePower');  
qamDemodulator = comm.RectangularQAMDemodulator('BitOutput',true,'NormalizationMethod','AveragePower');
```

Create an error rate object. Set the receiver delay to twice the traceback depth, which is the delay through the decoder.

```
errorRate = comm.ErrorRate('ReceiveDelay',2*traceBack);
```

Set the range of E_b/N_0 values to be simulated. Initialize the bit error rate statistics matrix.

```
ebnoVec = 0:2:10;  
errorStats = zeros(length(ebnoVec),3);
```

Simulate the link by following these steps:

- Generate binary data.

- Encode the data with a rate 2/3 convolutional code.
- Modulate the encoded data.
- Pass the signal through an AWGN channel.
- Demodulate the received signal.
- Decode the demodulated signal by using a Viterbi decoder.
- Collect the error statistics.

```

for m = 1:length(ebnoVec)
    snr = ebnoVec(m) + 10*log10(k*codeRate);

    while errorStats(m,2) <= 100 && errorStats(m,3) <= 1e7
        dataIn = randi([0 1],10000,1);
        dataEnc = convEncoder(dataIn);
        txSig = qamModulator(dataEnc);
        rxSig = awgn(txSig,snr);
        demodSig = qamDemodulator(rxSig);
        dataOut = vitDecoder(demodSig);
        errorStats(m,:) = errorRate(dataIn,dataOut);
    end
    reset(errorRate)
end

```

Compute the theoretical BER vs. Eb/No curve for the case without forward error correction coding.

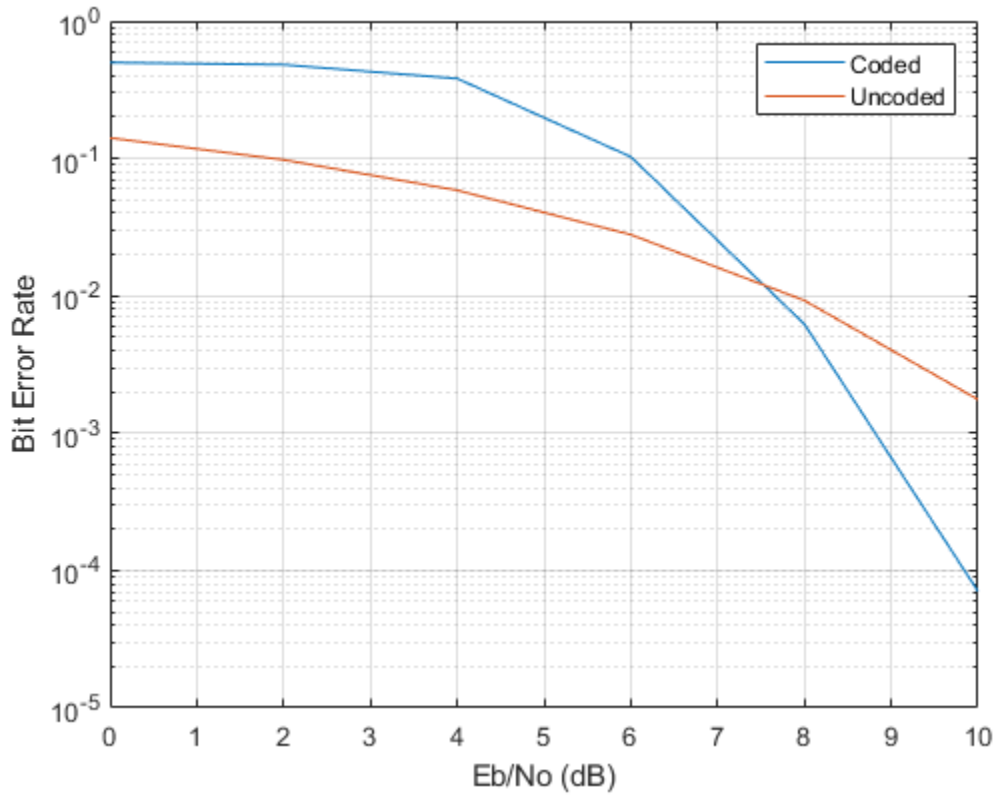
```
berUncoded = berawgn(ebnoVec, 'qam',M);
```

Plot the BER vs. Eb/No curve for the simulated coded data and the theoretical uncoded data.

```

semilogy(ebnoVec,[errorStats(:,1) berUncoded])
grid
legend('Coded','Uncoded')
xlabel('Eb/No (dB)')
ylabel('Bit Error Rate')

```



At higher Eb/No values, the error correcting code provides performance benefits.

Estimate BER for Hard and Soft Decision Viterbi Decoding

Estimate bit error rate (BER) performance for hard-decision and soft-decision Viterbi decoders in AWGN. Compare the performance to that of an uncoded 64-QAM link.

Set the simulation parameters.

```
clear; close all
rng default
M = 64;                % Modulation order
k = log2(M);          % Bits per symbol
EbNoVec = (4:10)';    % Eb/No values (dB)
numSymPerFrame = 1000; % Number of QAM symbols per frame
```

Initialize the BER results vectors.

```
berEstSoft = zeros(size(EbNoVec));
berEstHard = zeros(size(EbNoVec));
```

Set the trellis structure and traceback length for a rate 1/2, constraint length 7, convolutional code.

```
trellis = poly2trellis(7,[171 133]);
tbl = 32;
rate = 1/2;
```

The main processing loops performs these steps:

- Generate binary data.
- Convolutionally encode the data.
- Apply QAM modulation to the data symbols. Specify unit average power for the transmitted signal.
- Pass the modulated signal through an AWGN channel.
- Demodulate the received signal using hard decision and approximate LLR methods. Specify unit average power for the received signal.
- Viterbi decode the signals using hard and unquantized methods.
- Calculate the number of bit errors.

The `while` loop continues to process data until either 100 errors are encountered or $1e7$ bits are transmitted.

```
for n = 1:length(EbNoVec)
    % Convert Eb/No to SNR
    snrdB = EbNoVec(n) + 10*log10(k*rate);
    % Noise variance calculation for unity average signal power.
    noiseVar = 10.^(-snrdB/10);
    % Reset the error and bit counters
    [numErrsSoft,numErrsHard,numBits] = deal(0);

    while numErrsSoft < 100 && numBits < 1e7
        % Generate binary data and convert to symbols
        dataIn = randi([0 1],numSymPerFrame*k,1);

        % Convolutionally encode the data
        dataEnc = convenc(dataIn,trellis);

        % QAM modulate
        txSig = qammod(dataEnc,M,'InputType','bit','UnitAveragePower',true);

        % Pass through AWGN channel
        rxSig = awgn(txSig,snrdB,'measured');

        % Demodulate the noisy signal using hard decision (bit) and
        % soft decision (approximate LLR) approaches.
        rxDataHard = qamdemod(rxSig,M,'OutputType','bit','UnitAveragePower',true);
        rxDataSoft = qamdemod(rxSig,M,'OutputType','approxllr', ...
            'UnitAveragePower',true,'NoiseVariance',noiseVar);

        % Viterbi decode the demodulated data
        dataHard = vitdec(rxDataHard,trellis,tbl,'cont','hard');
        dataSoft = vitdec(rxDataSoft,trellis,tbl,'cont','unquant');

        % Calculate the number of bit errors in the frame. Adjust for the
        % decoding delay, which is equal to the traceback depth.
        numErrsInFrameHard = biterr(dataIn(1:end-tbl),dataHard(tbl+1:end));
        numErrsInFrameSoft = biterr(dataIn(1:end-tbl),dataSoft(tbl+1:end));

        % Increment the error and bit counters
        numErrsHard = numErrsHard + numErrsInFrameHard;
        numErrsSoft = numErrsSoft + numErrsInFrameSoft;
        numBits = numBits + numSymPerFrame*k;
    end

    % Estimate the BER for both methods
```

```

berEstSoft(n) = numErrsSoft/numBits;
berEstHard(n) = numErrsHard/numBits;
end

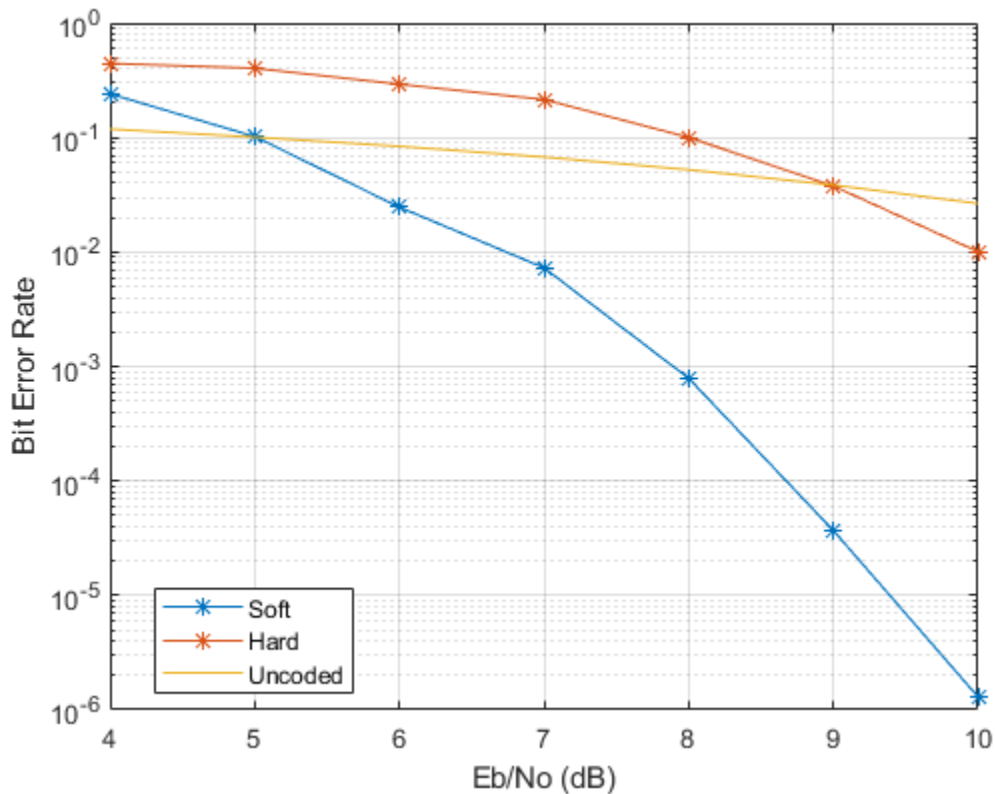
```

Plot the estimated hard and soft BER data. Plot the theoretical performance for an uncoded 64-QAM channel.

```

semilogy(EbNoVec,[berEstSoft berEstHard], '-*')
hold on
semilogy(EbNoVec,berawgn(EbNoVec,'qam',M))
legend('Soft','Hard','Uncoded','location','best')
grid
xlabel('Eb/No (dB)')
ylabel('Bit Error Rate')

```



As expected, the soft decision decoding produces the best results.

Creation, Validation, and Testing of User Defined Trellis Structure

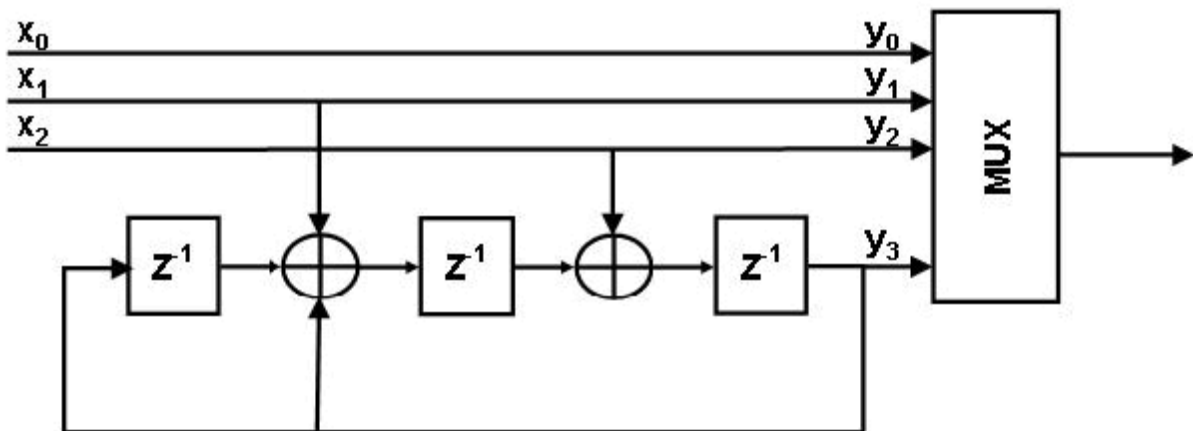
Create User Defined Trellis Structure

This example demonstrates creation of a nonstandard trellis structure for a convolutional encoder with uncoded bits and feedback. The encoder cannot be created using `poly2trellis` because the peculiar specifications for the encoder do not match the input requirements of `poly2trellis`.

Even though `poly2trellis` is not used to create the trellis structure, you can manually create the structure, then use that trellis structure as the input trellis structure for an encoder and decoder. The Convolutional Encoder and Viterbi Decoder blocks used in the “Convolutional Encoder with Uncoded Bits and Feedback” on page 13-34 model load the trellis structure created here using a `PreLoadFcn` callback.

Convolutional Encoder

Create a rate 3/4 convolutional encoder with feedback connection whose MSB bit remains uncoded.



Declare variables according to the specifications.

```
k = 3;
n = 4;
constraintLength = 4;
```

Create trellis structure

A trellis is represented by a structure with the following fields:

- `numInputSymbols` - Number of input symbols
- `numOutputSymbols` - Number of output symbols
- `numStates` - Number of states
- `nextStates` - Next state matrix
- `outputs` - Output matrix

For more information about these structure fields, see `istrellis`.

Reset any previous occurrence of `myTrellis` structure.

```
clear myTrellis;
```

Define the trellis structure fields.

```
myTrellis.numInputSymbols = 2^k;
myTrellis.numOutputSymbols = 2^n;
myTrellis.numStates = 2^(constraintLength-1);
```

Create nextStates Matrix

The `nextStates` matrix is a [`numStates` x `numInputSymbols`] matrix. The (i,j) element of the next state matrix is the resulting final state index that corresponds to a transition from the initial state i for an input equal to j .

```
myTrellis.nextStates = [0  1  2  3  0  1  2  3; ...
                        6  7  4  5  6  7  4  5; ...
                        1  0  3  2  1  0  3  2; ...
                        7  6  5  4  7  6  5  4; ...
                        2  3  0  1  2  3  0  1; ...
                        4  5  6  7  4  5  6  7; ...
                        3  2  1  0  3  2  1  0; ...
                        5  4  7  6  5  4  7  6]
```

```
myTrellis = struct with fields:
    numInputSymbols: 8
```



```

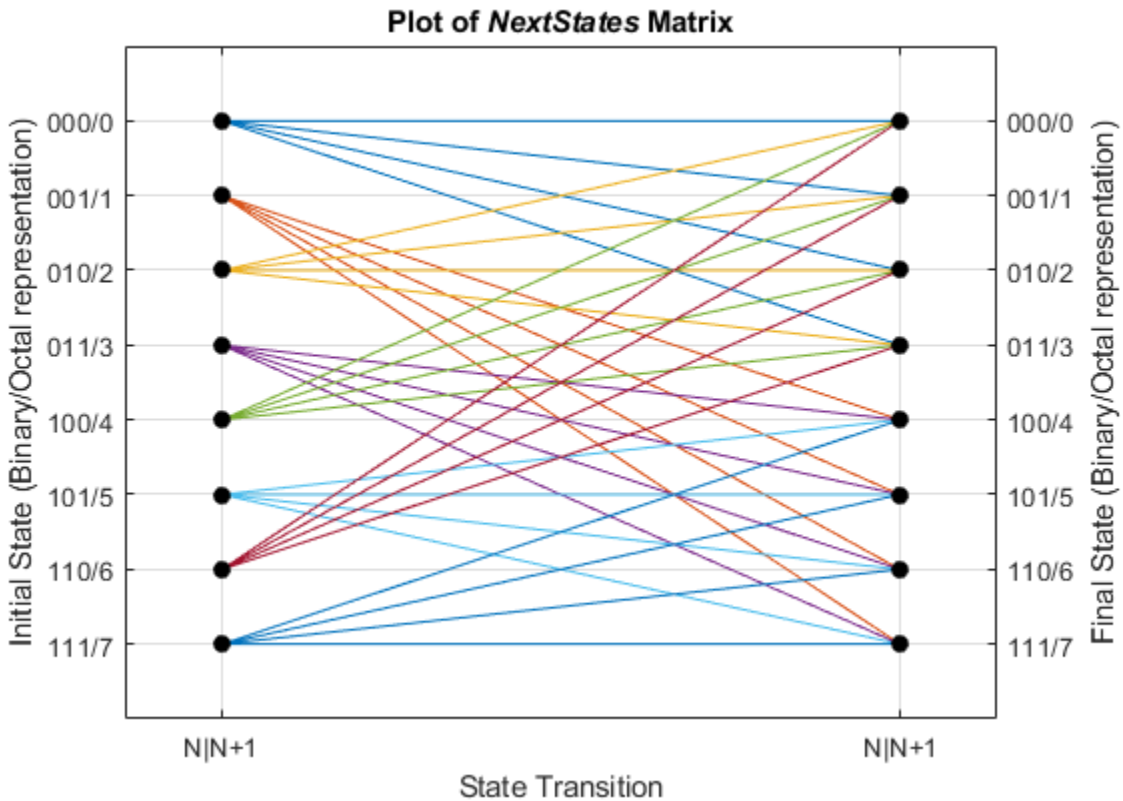
numOutputSymbols: 16
numStates: 8
nextStates: [8x8 double]

```

Plot nextStates Matrix

Use the `comcnv_plotnextstates` helper function to plot the `nextStates` matrix to illustrate the branch transitions between different states for a given input.

```
comcnv_plotnextstates(myTrellis.nextStates);
```



Create outputs Matrix

The `outputs` matrix is a `[numStates x numInputSymbols]` matrix. The (i,j) element of the output matrix is the output symbol in octal format given a current state i for an input equal to j .

```
outputs = [0  2  4  6  10 12 14 16; ...  
          1  3  5  7  11 13 15 17; ...  
          0  2  4  6  10 12 14 16; ...  
          1  3  5  7  11 13 15 17; ...  
          0  2  4  6  10 12 14 16; ...  
          1  3  5  7  11 13 15 17; ...  
          0  2  4  6  10 12 14 16; ...  
          1  3  5  7  11 13 15 17]
```

```
outputs = 8×8
```

0	2	4	6	10	12	14	16
1	3	5	7	11	13	15	17
0	2	4	6	10	12	14	16
1	3	5	7	11	13	15	17
0	2	4	6	10	12	14	16
1	3	5	7	11	13	15	17
0	2	4	6	10	12	14	16
1	3	5	7	11	13	15	17

Use `oct2dec` to display these values in decimal format.

```
outputs_dec = oct2dec(outputs)
```

```
outputs_dec = 8×8
```

0	2	4	6	8	10	12	14
1	3	5	7	9	11	13	15
0	2	4	6	8	10	12	14
1	3	5	7	9	11	13	15
0	2	4	6	8	10	12	14
1	3	5	7	9	11	13	15
0	2	4	6	8	10	12	14
1	3	5	7	9	11	13	15

Copy `outputs` matrix into the `myTrellis` structure.

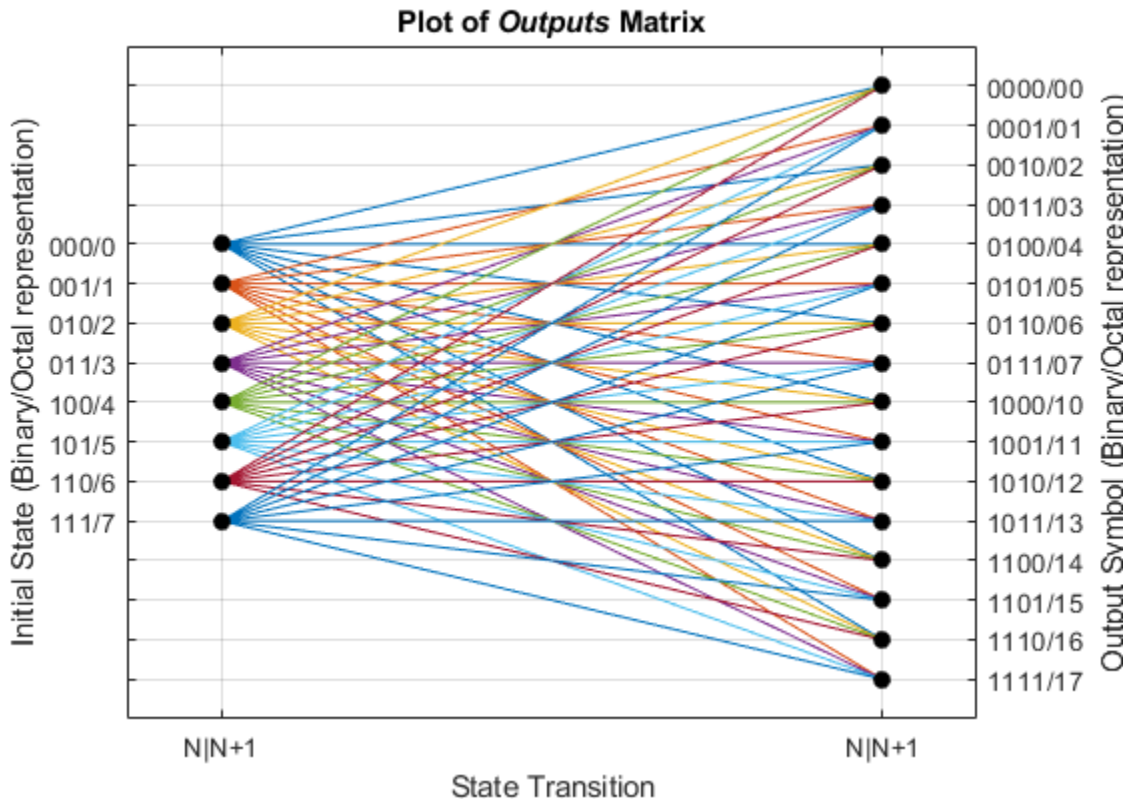
```
myTrellis.outputs = outputs
```

```
myTrellis = struct with fields:
    numInputSymbols: 8
    numOutputSymbols: 16
    numStates: 8
    nextStates: [8x8 double]
    outputs: [8x8 double]
```

Plot outputs Matrix

Use the `comcnv_plotoutputs` helper function to plot the outputs matrix to illustrate the possible output symbols for a given state depending on the input symbol.

```
comcnv_plotoutputs(myTrellis.outputs, myTrellis.numOutputSymbols);
```



Check Resulting Trellis Structure

```
istrellis(myTrellis)
```

```
ans = logical
     1
```

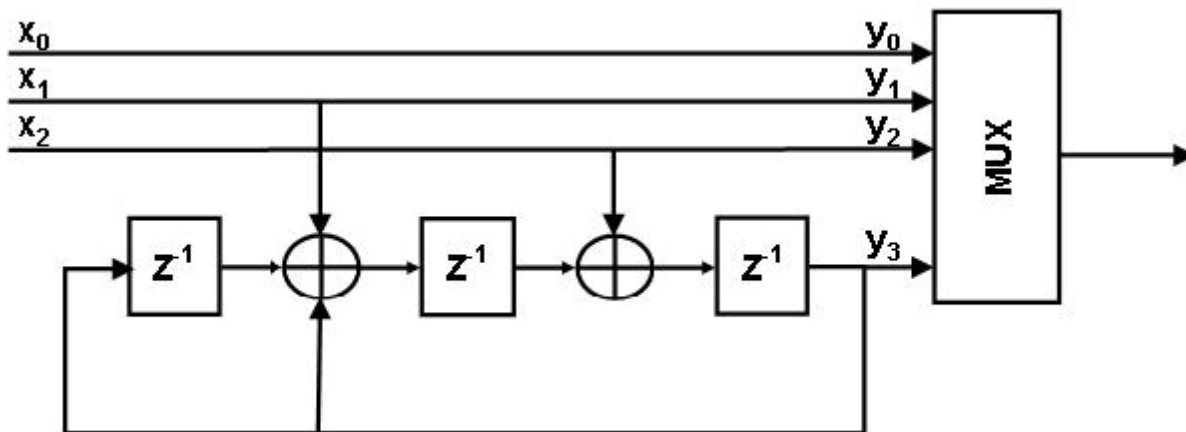
A return value of 1 confirms the trellis structure is valid.

Convolutional Encoder with Uncoded Bits and Feedback

The model serves as a unit test bench for the convolutional code implemented. The model shows how to define and use a trellis that describes a convolutional code. The particular code in this example cannot be described by a set of generator and feedback connection polynomials. The code's trellis cannot be created by the `poly2trellis` because that function expects generator and feedback connection polynomials as input arguments.

Structure of the Convolutional Code

This figure shows the convolutional code.

**Structure of the Example**

The major components in this example include:

- A transmit path that builds a representation of the convolutional encoder using discrete low-level delay and sum (XOR) blocks. This representation looks very similar to the figure showing the structure of the convolutional code.
- A transmit-receive path that builds a representation of the same convolutional encoder using the Convolutional Encoder block. In this case, the description of the encoder is within the block's Trellis structure parameter. This portion of the model also includes the Viterbi Decoder block, which decodes the convolutional code.
- Both paths compute the number of bit errors.

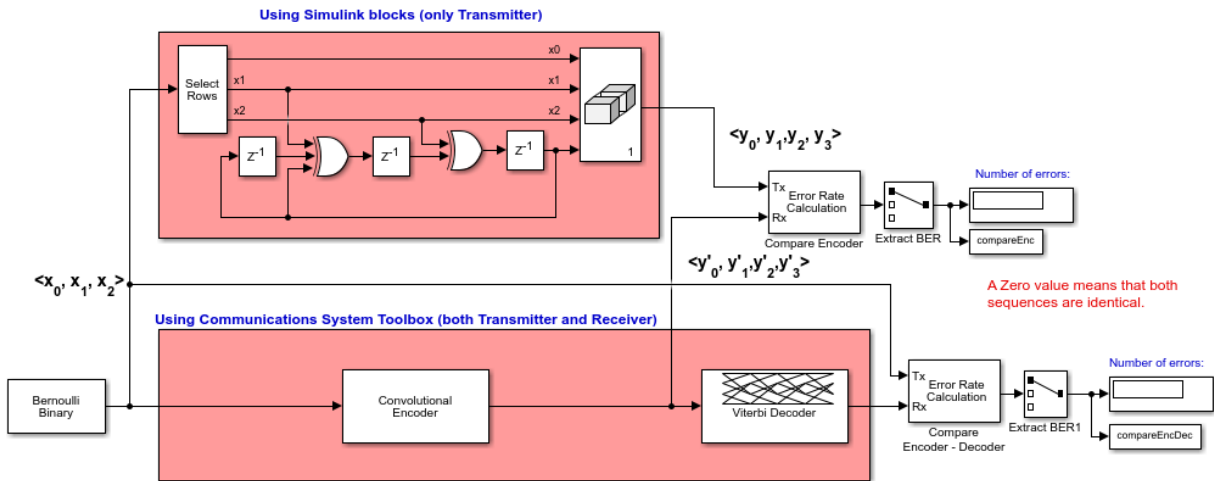
Open Example Model and Explore Its Contents

Open the example model `sllex_commcnencoder`.

**Convolutional Encoder with Uncoded Bits and Feedback:
(using User-defined Trellis structure)**

Double-click on the block to view the block diagram of the encoder

Display Block Diagram



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Results and Displays

When you run the simulation, the block labeled Compare Encoder checks that the two representations of the encoder yield the same result. The block labeled Compare Encoder - Decoder checks that the encoder and decoder work properly as a pair. Each Display block in the model shows an error rate of zero, as expected.

Error rate for Compare Encoder signal: 0.000
Error rate for Compare Encoder-Decoder signal: 0.000

See Also

Channel Modeling and RF Impairments

- “AWGN Channel” on page 14-2
- “Configure Eb/No for AWGN Channels with Coding” on page 14-6
- “Using AWGN Channel Block for Coded Signals” on page 14-8
- “Fading Channels” on page 14-10
- “Using Channel Visualization” on page 14-46
- “MIMO Channel” on page 14-47
- “WINNER II Channel” on page 14-48
- “Mapping of WINNER II Open Source Download to WINNER II Channel Model for Communications Toolbox” on page 14-49
- “RF Impairments” on page 14-53

AWGN Channel

In this section...
“Section Overview” on page 14-2
“AWGN Channel Noise Level” on page 14-2

Section Overview

An AWGN channel adds white Gaussian noise to the signal that passes through it. You can create an AWGN channel in a model using the `comm.AWGNChannel` System object, the AWGN Channel block, or the `awgn` function.

The following examples use an AWGN Channel: “QPSK Transmitter and Receiver” and “General QAM Modulation in AWGN Channel” on page 9-60.

AWGN Channel Noise Level

Typical quantities used to describe the relative power of noise in an AWGN channel include

- Signal-to-noise ratio (SNR) per sample. SNR is the actual input parameter to the `awgn` function.
- Ratio of bit energy to noise power spectral density (E_b/N_0). This quantity is used by BER Analyzer Tool and performance evaluation functions in this toolbox.
- Ratio of symbol energy to noise power spectral density (E_s/N_0)

Relationship Between E_s/N_0 and E_b/N_0

The relationship between E_s/N_0 and E_b/N_0 , both expressed in dB, is as follows:

$$E_s/N_0 \text{ (dB)} = E_b/N_0 \text{ (dB)} + 10\log_{10}(k)$$

where k is the number of information bits per symbol.

In a communications system, k might be influenced by the size of the modulation alphabet or the code rate of an error-control code. For example, in a system using a rate-1/2 code and 8-PSK modulation, the number of information bits per symbol (k) is the product of the code rate and the number of coded bits per modulated symbol. Specifically, $(1/2) \log_2(8) =$

3/2. In such a system, three information bits correspond to six coded bits, which in turn correspond to two 8-PSK symbols.

Relationship Between EsN0 and SNR

The relationship between EsN0 and SNR, both expressed in dB, is as follows:

$$E_s/N_0 \text{ (dB)} = 10\log_{10}(T_{sym}/T_{samp}) + SNR \text{ (dB)} \quad \text{for complex input signals}$$

$$E_s/N_0 \text{ (dB)} = 10\log_{10}(0.5T_{sym}/T_{samp}) + SNR \text{ (dB)} \quad \text{for real input signals}$$

where T_{sym} is the symbol period of the signal and T_{samp} is the sampling period of the signal.

For a complex baseband signal oversampled by a factor of 4, the EsN0 exceeds the corresponding SNR by $10 \log_{10}(4)$.

Derivation for Complex Input Signals

You can derive the relationship between EsN0 and SNR for complex input signals as follows:

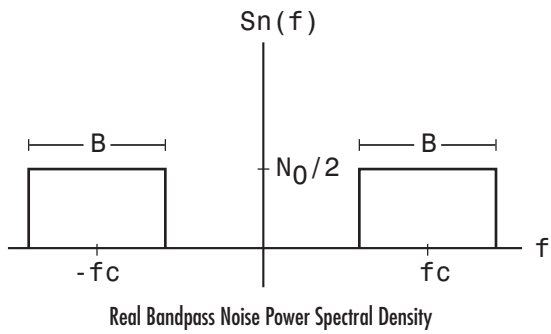
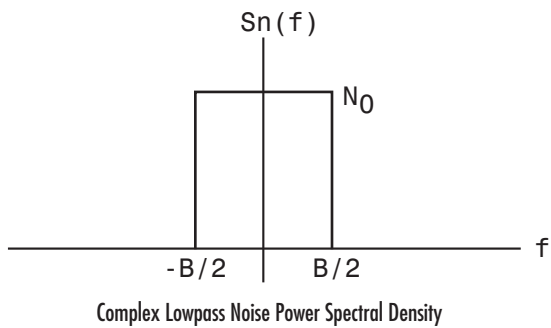
$$\begin{aligned} E_s/N_0 \text{ (dB)} &= 10\log_{10}((S \cdot T_{sym})/(N/B_n)) \\ &= 10\log_{10}((T_{sym}F_s) \cdot (S/N)) \\ &= 10\log_{10}(T_{sym}/T_{samp}) + SNR \text{ (dB)} \end{aligned}$$

where

- S = Input signal power, in watts
- N = Noise power, in watts
- B_n = Noise bandwidth, in Hertz = $F_s = 1/T_{samp}$.
- F_s = Sampling frequency, in Hertz

Behavior for Real and Complex Input Signals

These figures illustrate the difference between the real and complex cases by showing the noise power spectral densities of a real bandpass white noise process and its complex lowpass equivalent.



See Also

Objects

`comm.AWGNChannel`

Blocks

AWGN Channel

Functions

`awgn`

More About

- “QPSK Transmitter and Receiver”

- “General QAM Modulation in AWGN Channel” on page 9-60

Configure Eb/No for AWGN Channels with Coding

This example shows how to set the bit energy to noise density ratio (Eb/No) for communication links employing channel coding.

Specify the codeword and message length for a Reed-Solomon code. Specify the modulation order.

```
N = 15;      % R-S codeword length in symbols
K = 9;      % R-S message length in symbols
M = 16;     % Modulation order
```

Construct a (15,9) Reed-Solomon encoder and a 16-PSK modulator. Specify the objects so that they accept bit inputs.

```
rsEncoder = comm.RSEncoder('CodewordLength',N,'MessageLength',K, ...
    'BitInput',true);
pskModulator = comm.PSKModulator('ModulationOrder',M,'BitInput',true);
```

Create the corresponding Reed-Solomon decoder and 16-PSK demodulator objects.

```
rsDecoder = comm.RSDecoder('CodewordLength',N,'MessageLength',K, ...
    'BitInput',true);
pskDemodulator = comm.PSKDemodulator('ModulationOrder',M,'BitOutput',true);
```

Calculate the Reed-Solomon code rate based on the ratio of message symbols to the codeword length. Determine the bits per symbol for the PSK modulator.

```
codeRate = K/N;
bitsPerSymbol = log2(M);
```

Specify the uncoded Eb/No in dB. Convert the uncoded Eb/No to the corresponding coded Eb/No using the code rate.

```
UncodedEbNo = 6;
CodedEbNo = UncodedEbNo + 10*log10(codeRate);
```

Construct an AWGN channel taking into account the number of bits per symbol. Set the EbNo property of channel to the coded Eb/No.

```
channel = comm.AWGNChannel('BitsPerSymbol',bitsPerSymbol);
channel.EbNo = CodedEbNo;
```

Set the total number of errors and bits for the simulation. For accuracy, the simulation should run until a sufficient number of bit errors are encountered. The number of total bits is used to ensure that the simulation does not run too long.

```
totalErrors = 100;
totalBits = 1e6;
```

Construct an error rate calculator System object™ and initialize the error rate vector.

```
errorRate = comm.ErrorRate;
errorVec = zeros(3,1);
```

Run the simulation to determine the BER.

```
while errorVec(2) < totalErrors && errorVec(3) < totalBits
    % Generate random bits
    dataIn = randi([0,1],360,1);
    % Use the RS (15,9) encoder to add error correction capability
    dataEnc = rsEncoder(dataIn);
    % Apply 16-PSK modulation
    txSig = pskModulator(dataIn);
    % Pass the modulated data through the AWGN channel
    rxSig = channel(txSig);
    % Demodulate the received signal
    demodData = pskDemodulator(rxSig);
    % Decode the demodulated data with the RS (15,9) decoder
    dataOut = rsDecoder(demodData);
    % Collect error statistics
    errorVec = errorRate(dataIn,demodData);
end
```

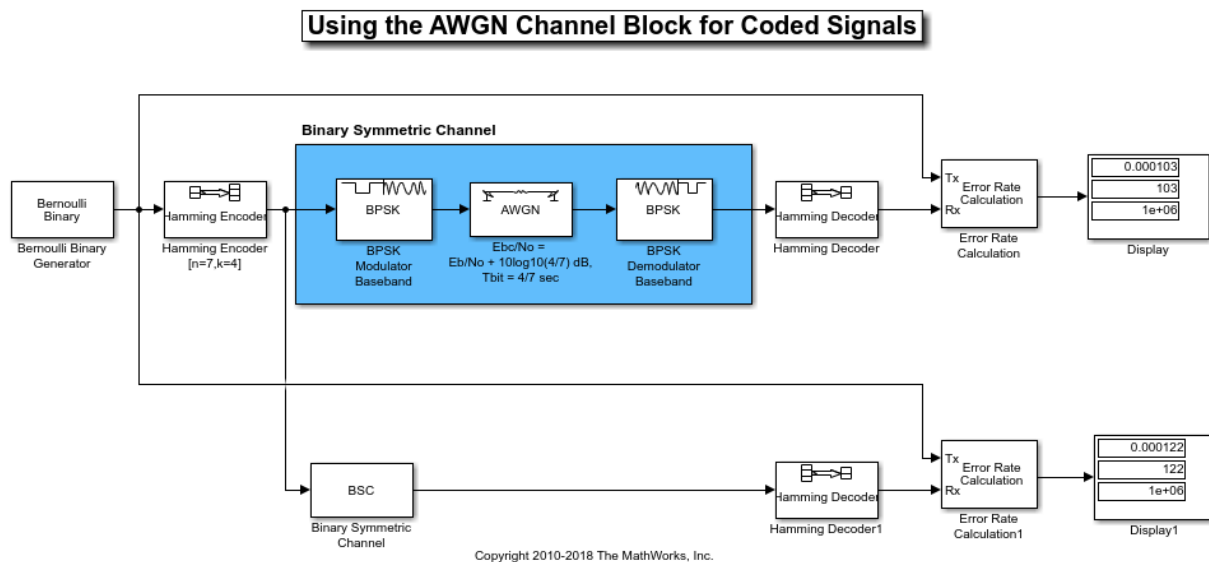
Display the resultant bit error rate.

```
ber = errorVec(1)
ber = 0.0935
```

Using AWGN Channel Block for Coded Signals

Two links perform error control coding on a signal that has passed through an impairment channel. Both links are set for an uncoded E_b/N_0 of 8 dB.

In the top link, the AWGN channel block is set to provide a coded E_b/N_0 of $8 + 10\log_{10}(4/7)$ dB, where $4/7$ is the code rate. This accounts for the fact that the coded E_b/N_0 is always lower (by a factor of the code rate) than the uncoded E_b/N_0 . The blue shaded portion of the top link is simply a binary symmetric channel, which is modeled more compactly in the bottom link. The channel error probability of the top link is $Q(\sqrt{2 \cdot E_{bc}/N_0})$, where $Q()$ is the standard Q function and E_{bc}/N_0 is the coded E_b/N_0 (in absolute terms, not in dB).



For this example, it is important to note that the bit period at the input of the AWGN Channel block is $4/7$ sec. It is 1 sec at the input of the Hamming Encoder block, but that block decreases the bit time by a factor of the code rate.

If you allow the model to run for $1e6$ bits, you'll note that the BERs are virtually identical. The difference lies in the stochastic nature of the two random number generators.

You can also check these BER results against expected analytical results by typing this command at the MATLAB® command prompt.

$\text{BER} = \text{bercoding}(8, 'block', 'hard', 7, 4, 3)$

This expression finds the upper bound of the BER of a linear, rate 4/7 block code with a minimum distance of 3, and hard decision decoding.

Fading Channels

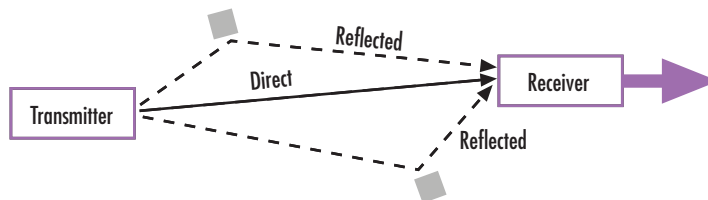
In this section...

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Overview of Fading Channels

Using Communications Toolbox you can implement fading channels using objects or blocks. Rayleigh and Rician fading channels are useful models of real-world phenomena in wireless communications. These phenomena include multipath scattering effects, time dispersion, and Doppler shifts that arise from relative motion between the transmitter and receiver. This section gives a brief overview of fading channels and describes how to implement them using the toolbox.

This figure depicts direct and major reflected paths between a stationary radio transmitter and a moving receiver. The shaded shapes represent reflectors such as buildings.

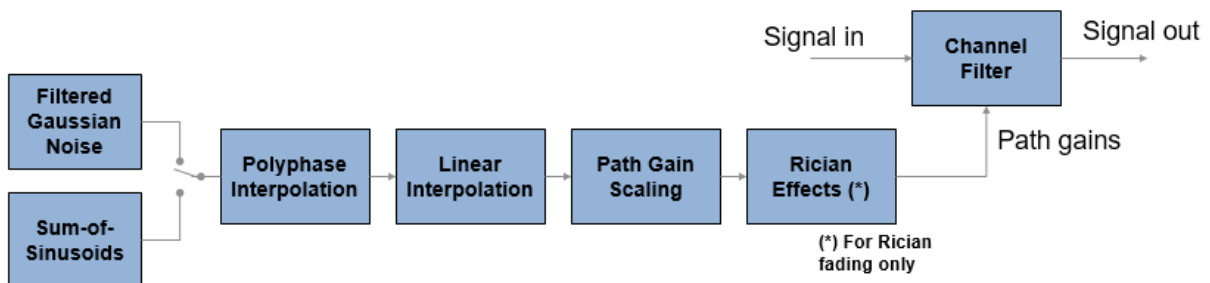


The major paths result in the arrival of delayed versions of the signal at the receiver. In addition, the radio signal undergoes scattering on a *local* scale for each major path. Such local scattering typically results from reflections off objects near the mobile. These irresolvable components combine at the receiver and cause a phenomenon known as

multipath fading. Due to this phenomenon, each major path behaves as a discrete fading path. Typically, the fading process is characterized by a Rayleigh distribution for a non line-of-sight path and a Rician distribution for a line-of-sight path.

The relative motion between the transmitter and receiver causes Doppler shifts. Local scattering typically comes from many angles around the mobile. This scenario causes a range of Doppler shifts, known as the *Doppler spectrum*. The *maximum* Doppler shift corresponds to the local scattering components whose direction exactly opposes the trajectory of the mobile.

The channel filter applies path gains to the input signal, `Signal in`. The path gains are configured based on settings chosen in the fading channel object or block.



These blocks and objects enable you to model SISO or MIMO fading channels.

Tool	SISO	MIMO
MATLAB	<code>comm.RayleighChannel</code> <code>comm.RicianChannel</code>	<code>comm.MIMOChannel</code>
Simulink	SISO Fading Channel	MIMO Fading Channel

Implement Fading Channel Using an Object

A baseband channel model for multipath propagation scenarios that you implement using objects includes:

- N discrete fading paths. Each path has its own delay and average power gain. A channel for which $N = 1$ is called a *frequency-flat fading channel*. A channel for which

$N > 1$ is experienced as a *frequency-selective fading channel* by a signal of sufficiently wide bandwidth.

- A Rayleigh or Rician model for each path.
- Default channel path modeling using a Jakes Doppler spectrum, with a maximum Doppler shift that can be specified. Other types of Doppler spectra allowed (identical or different for all paths) include: flat, restricted Jakes, asymmetrical Jakes, Gaussian, bi-Gaussian, rounded, and bell.

If the maximum Doppler shift is set to 0 or omitted during the construction of a channel object, then the object models the channel as static. For this configuration, the fading does not evolve with time and the Doppler spectrum specified has no effect on the fading process.

Some additional information about typical values for delays and gains is in Choosing Realistic Channel Property Values on page 14-29.

Implement Fading Channel Using a Block

The Channels block library includes MIMO and SISO fading blocks that can simulate real-world phenomena in mobile communications. These phenomena include multipath scattering effects, in addition to Doppler shifts that arise from relative motion between the transmitter and receiver.

Tip To model a channel that involves both fading and additive white Gaussian noise, use a fading channel block followed by an AWGN Channel block.

The MIMO Fading Channel and SISO Fading Channel blocks can be set to model Rayleigh or Rician fading distributions of the channel. Based on the type of signal path, choose the fading distribution to use.

Signal Path	Fading Distribution
Direct line-of-sight path from transmitter to receiver	Rician
One or more major reflected paths from transmitter to receiver	Rayleigh

You can use a single instance of a fading channel block configured for Rayleigh fading distribution to model multiple major reflected paths simultaneously.

Choosing appropriate block parameters for your situation is important. For more information, see Choosing Realistic Channel Property Values on page 14-29, and the MIMO Fading Channel and SISO Fading Channel block reference pages.

Visualize a Fading Channel

You can view the characteristics of a fading channel using channel visualization tools. For more information, see Channel Visualization on page 17-29.

Compensate for Fading Response

A communications system involving a fading channel usually requires components that compensate for the fading response. Typical approaches to compensate for fading include:

- Differential modulation or a one-tap equalizer helps compensate for a frequency-flat fading channel. For information about implementing differential modulation, see the M-DPSK Modulator Baseband block reference page.
- An equalizer with multiple taps helps compensate for a frequency-selective fading channel. See Equalization on page 6-2 for more information.

The Communications Link with Adaptive Equalization example illustrates why compensating for a fading channel is necessary.

Methodology for Simulating Multipath Fading Channels

The Rayleigh and Rician multipath fading channel simulators in Communications Toolbox use the band-limited discrete multipath channel model of section 9.1.3.5.2 in [1]. This implementation assumes that the delay power profile and the Doppler spectrum of the channel are separable [1]. The multipath fading channel is therefore modeled as a linear finite impulse-response (FIR) filter. Let $\{s_i\}$ denote the set of samples at the input to the channel. Then the samples $\{y_i\}$ at the output of the channel are related to $\{s_i\}$ through:

$$y_i = \sum_{n=-N_1}^{N_2} s_{i-n} g_n$$

where $\{g_n\}$ is the set of tap weights given by:

$$g_n = \sum_{k=1}^K a_k \text{sinc}\left[\frac{\tau_k}{T_s} - n\right], \quad -N_1 \leq n \leq N_2$$

In the equations:

- T_s is the input sample period to the channel.
- $\{\tau_k\}$, where $1 \leq k \leq K$, is the set of path delays. K is the total number of paths in the multipath fading channel.
- $\{a_k\}$, where $1 \leq k \leq K$, is the set of complex path gains of the multipath fading channel. These path gains are uncorrelated with each other.
- N_1 and N_2 are chosen so that $|g_n|$ is small when n is less than $-N_1$ or greater than N_2 .

Two techniques, filtered Gaussian noise and sum-of-sinusoids, are used to generate the set of complex path gains, a_k .

Each path gain process a_k is generated by the following steps:

Filtered Gaussian Noise Technique

- 1** A complex uncorrelated (white) Gaussian process with zero mean and unit variance is generated in discrete time.
- 2** The complex Gaussian process is filtered by a Doppler filter with frequency response $H(f) = \sqrt{S(f)}$, where $S(f)$ denotes the desired Doppler power spectrum.
- 3** The filtered complex Gaussian process is interpolated so that its sample period is consistent with the sample period of the input signal. A combination of linear and polyphase interpolation is used.

Sum-of-sinusoids Technique

- 1** Mutually uncorrelated Rayleigh fading waveforms are generated using the method described in [2], where $i = 1$ corresponds to the in-phase component and $i = 2$ corresponds to the quadrature component.

$$z_k(t) = \mu_k^{(1)}(t) + j\mu_k^{(2)}(t), \quad k = 1, 2, \dots, K$$

$$\mu_k^{(i)}(t) = \sqrt{\frac{2}{N_k}} \sum_{n=1}^{N_k} \cos(2\pi f_{k,n}^{(i)} t + \theta_{k,n}^{(i)}), \quad i = 1, 2$$

Where

- N_k specifies the number of sinusoids used to model a single path.

- $f_{k,n}^{(i)}$ is the discrete Doppler frequency and is calculated for each sinusoid component within a single path.
- $\theta_{k,n}^{(i)}$ is the phase of the n th component of $\mu_k^{(i)}$ and is an i.i.d. random variable having a uniform distribution over the interval $(0, 2\pi]$.
- t is the fading process time.

When modeling a Jakes Doppler spectrum, the discrete Doppler frequencies, $f_{k,n}^{(i)}$, with maximum shift f_{\max} are given by

$$\begin{aligned} f_{k,n}^{(i)} &= f_{\max} \cos(\alpha_{k,n}^{(i)}) \\ &= f_{\max} \cos\left[\frac{\pi}{2N_k} \left(n - \frac{1}{2}\right) + \alpha_{k,0}^{(i)}\right] \end{aligned}$$

where

$$\alpha_{k,0}^{(i)} \triangleq (-1)^{i-1} \frac{\pi}{4N_k} \cdot \frac{k}{K+2}, \quad i = 1, 2 \text{ and } k = 1, 2, \dots, K$$

- 2 To advance the fading process in time, an initial time parameter, t_{init} , is introduced. The fading waveforms become

$$\mu_k^{(i)}(t) = \sqrt{\frac{2}{N_k}} \sum_{n=1}^{N_k} \cos\left(2\pi f_{k,n}^{(i)}(t + t_{\text{init}}) + \theta_{k,n}^{(i)}\right), \quad i = 1, 2$$

When $t_{\text{init}} = 0$, the fading process starts at time zero. A positive value of t_{init} advances the fading process relative to time zero while maintaining its continuity.

- 3 Channel fading samples are generated using the GMEDS₁ [2] algorithm.

Calculate Complex Coefficients

The complex process resulting from either technique, z_k , is scaled to obtain the correct average path gain. In the case of a Rayleigh channel, the fading process is obtained as:

$$a_k = \sqrt{\Omega_k} z_k$$

where

$$\Omega_k = E[|a_k|^2]$$

In the case of a Rician channel, the fading process is obtained as:

$$a_k = \sqrt{\Omega_k} \left[\frac{z_k}{\sqrt{K_{r,k} + 1}} + \sqrt{\frac{K_{r,k}}{K_{r,k} + 1}} e^{j(2\pi f_{d,LOS,k} t + \theta_{LOS,k})} \right]$$

where $K_{r,k}$ is the Rician K-factor of the k th path, $f_{d,LOS,k}$ is the Doppler shift of the line-of-sight component of the k th path (in Hz), and $\theta_{LOS,k}$ is the initial phase of the line-of-sight component of the k th path (in rad).

At the input to the band-limited multipath channel model, the transmitted symbols must be oversampled by a factor at least equal to the bandwidth expansion factor introduced by pulse shaping. For example, if sinc pulse shaping is used, for which the bandwidth of the pulse-shaped signal is equal to the symbol rate, then the bandwidth expansion factor is 1, and at least one sample per symbol is required at the input to the channel. If a raised cosine (RC) filter with a factor more than 1 is used, for which the bandwidth of the pulse-shaped signal is equal to twice the symbol rate, then the bandwidth expansion factor is 2, and at least two samples per symbol are required at the input to the channel.

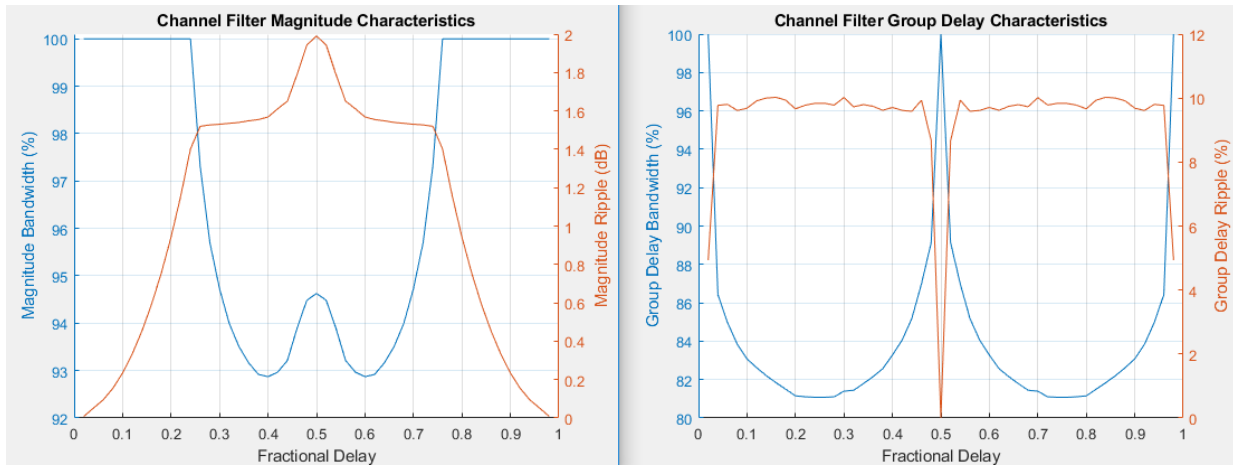
Channel Filter Model Characteristics

The channel model implements a fractional delay (FD) finite impulse response (FIR) bandpass filter with a length of 16 coefficients and with a delay error limited to 1%. For each discrete path, the FIR design is selected by choosing the optimal set of coefficients from among 50 fractional delay (0.02, 0.04, ..., 0.98) candidate sets of coefficients.

Each discrete path is rounded to its selected fractional delay. To achieve a group delay bandwidth exceeding 80% and a magnitude bandwidth exceeding 90%, the algorithm selects the FIR coefficient values for each fractional delay, while satisfying the following criteria:

- Group delay ripple $\leq 10\%$
- Magnitude ripple ≤ 2 dB
- Magnitude bandedge attenuation = 3 dB

The plots show bandwidths that satisfy the design criteria for group delay ripple, magnitude ripple, and magnitude bandedge attenuation.



For additional information, see the article *A Matlab-based Object-Oriented Approach to Multipath Fading Channel Simulation* at MATLAB Central.

References

- [1] Jeruchim, M. C., Balaban, P., and Shanmugan, K. S., *Simulation of Communication Systems*, Second Edition, New York, Kluwer Academic/Plenum, 2000.
- [2] Pätzold, Matthias, Cheng-Xiang Wang, and Bjorn Olav Hogstand. "Two New Sum-of-Sinusoids-Based Methods for the Efficient Generation of Multiple Uncorrelated Rayleigh Fading Waveforms." *IEEE Transactions on Wireless Communications*. Vol. 8, Number 6, 2009, pp. 3122–3131.

Specify Fading Channels

Communications Toolbox models a fading channel as a linear FIR filter. Filtering a signal using a fading channel involves these steps:

- 1 Create a channel object that describes the channel that you want to use. A channel object is a type of MATLAB variable that contains information about the channel, such as the maximum Doppler shift.
- 2 Adjust properties of the channel object, if necessary, to tailor it to your needs. For example, you can change the path delays or average path gains.

3 Apply the channel object to your signal using calling the object.

This section describes how to define, inspect, and manipulate channel objects. The topics are:

- “Creating Channel Objects” on page 14-18
- “Duplicate and Copy Objects” on page 14-18
- “Displaying and Changing Object Properties” on page 14-19
- “Relationships Among Channel Object Properties” on page 14-21

Creating Channel Objects

To create a fading channel object suitable for your modeling situation, select one of these System objects.

Function	Object	Situation Modeled
<code>comm.RayleighChannel</code>	Rayleigh fading channel object	One or more major reflected paths
<code>comm.RicianChannel</code>	Rician fading channel object	One direct line-of-sight path, possibly combined with one or more major reflected paths

For example, this command creates a channel object representing a Rayleigh fading channel that acts on a signal sampled at 100,000 Hz. The maximum Doppler shift of the channel is 130 Hz.

```
rayChan1 = comm.RayleighChannel('SampleRate',1e5,...
    'MaximumDopplerShift',130); % Rayleigh fading channel object
```

To learn how to call the `rayChan1` fading channel object to filter the transmitted signal through the channel, see [Using Fading Channels](#) on page 14-31.

Duplicate and Copy Objects

You can also create another object by duplicating an existing object and then adjust the properties of the new object, if necessary. To duplicate an object always use the `clone` function such as:

```
rayChan2 = clone(rayChan1); % Copy rayChan1 to create an independent rayChan2.
```

instead of `rayChan2 = rayChan1`. The `clone` command creates a copy of `rayChan1` that is independent of `rayChan1`. By contrast, the command `rayChan2 = rayChan1`

creates `rayChan2` as merely a reference to `rayChan1`, so that `rayChan1` and `rayChan2` always have indistinguishable content.

Displaying and Changing Object Properties

A channel object has numerous properties that record information about the channel model, about the state of a channel that has already filtered a signal, and about the channel operation on a future signal.

You can view the properties in these ways:

- To view all properties of a channel object, enter the object name in the Command Window.
- You can view a property of a channel object or assign the value to a variable by entering the object name followed by a dot (period), followed by the name of the property.

You can change the writable properties of a channel object in these ways:

- To change the default value of a channel object property, enter the desired value in the object creation syntax.
- To change the value of a writeable property of a channel object, issue an assignment statement that uses dot notation on the channel object. More specifically, dot notation means an expression that consists of the object name, followed by a dot, followed by the name of the property.

Display Rayleigh Channel Object Properties

Create a Rayleigh channel object. Display the object to show the properties initialized by default and the ones specified when creating the object. Entering `rayChan` displays all properties of the channel object. Some of the properties values were assigned when the object was created, while other properties have default values. For more information about specific channel properties, see the reference page for the `comm.RayleighChannel` object.

```
rayChan = comm.RayleighChannel('SampleRate',1e5,'MaximumDopplerShift',130);
rayChan % View all the properties
```

```
rayChan =
  comm.RayleighChannel with properties:
```

```
    SampleRate: 100000
```

```
        PathDelays: 0
    AveragePathGains: 0
    NormalizePathGains: true
    MaximumDopplerShift: 130
        DopplerSpectrum: [1x1 struct]
```

Show all properties

```
g = rayChan.AveragePathGains % Retrieve the AveragePathGains property of rayChan
g = 0
```

Adjust Rician Channel Object Properties

A Rician fading channel object has an additional property that does not appear for a Rayleigh fading channel object, namely, a scalar `KFactor` property. For more information about Rician channel properties, see the reference page for the `comm.RicianChannel` object.

Change Rician Channel Object Properties

Create a Rician channel object. The default setting for the `Visualization` property is 'Off'. Changing the default setting to 'Impulse response' generates an impulse response plot of the output signal when the object is called. The output displays a subset of all the properties of the channel object. Select all properties to see the complete set of properties for `ricChan`.

```
ricChan= comm.RicianChannel; % Create object
ricChan.Visualization = 'Impulse response' % Enables the impulse response channel visu

ricChan =
    comm.RicianChannel with properties:

        SampleRate: 1
        PathDelays: 0
        AveragePathGains: 0
        NormalizePathGains: true
        KFactor: 3
    DirectPathDopplerShift: 0
    DirectPathInitialPhase: 0
    MaximumDopplerShift: 1.0000e-03
        DopplerSpectrum: [1x1 struct]
```

Show all properties

Relationships Among Channel Object Properties

Some properties of a channel object are related to each other such that when one property's value changes, another property's value must change in some corresponding way to keep the channel object consistent. For example, if you change the vector length of `PathDelays`, then the value of `AveragePathGains` must change so that its vector length equals that of the new value of `PathDelays`. This is because the length of each of the two vectors equals the number of discrete paths of the channel. For details about linked properties and an example, see `comm.RayleighChannel` or `comm.RicianChannel`.

Specify Doppler Spectrum of Fading Channel

The Doppler spectrum of a channel object is specified through its `DopplerSpectrum` property. The value of this property must be either:

- A Doppler spectrum structure. In this case, the same Doppler spectrum applies to each path of the channel object.
- A cell array of Doppler spectrum structures of the same length as the `PathDelays` vector property. In this case, the Doppler spectrum of each path is given by the corresponding Doppler spectrum structure in the vector.
 - When the vector length of the `PathDelays` property is increased, the length of `DopplerSpectrum` is automatically increased to match the length of `PathDelays`, by appending Jakes Doppler spectrum structures.
 - If the length of the `PathDelays` vector property is decreased, the length of `DopplerSpectrum` is automatically decreased to match the length of `PathDelays`, by removing the last Doppler spectrum structures.

A Doppler spectrum structure contains the properties used to characterize the Doppler spectrum, but the maximum Doppler shift is a property of the channel object. This section describes how to create and manipulate Doppler spectrum structures, and how to assign them to the `DopplerSpectrum` property of channel objects.

Create a Doppler Spectrum Structure

To create Doppler spectrum structures, use the `doppler` function. The sole purpose of the `doppler` function is to create Doppler spectrum structures used to specify the value of the `DopplerSpectrum` property of channel objects. Select from the following:

- `doppler('Jakes')`
- `doppler('Flat')`
- `doppler('Rounded', ...)`
- `doppler('Bell', ...)`
- `doppler('Asymmetric Jakes', ...)`
- `doppler('Restricted Jakes', ...)`
- `doppler('Gaussian', ...)`
- `doppler('BiGaussian', ...)`

For example, a Gaussian spectrum with a normalized (by the maximum Doppler shift of the channel) standard deviation of 0.1, can be created as:

```
doppl = doppler('Gaussian',0.1);
```

Note When creating a Doppler spectrum structure, consider the following dependencies:

- If you assign a single Doppler spectrum structure to `DopplerSpectrum`, all paths have the same specified Doppler spectrum.
 - If the `FadingTechnique` property is 'Sum of sinusoids', `DopplerSpectrum` must be `doppler('Jakes')`;
 - If you assign a row cell array of different Doppler spectrum structures to `DopplerSpectrum`, each path has the Doppler spectrum specified by the corresponding structure in the cell array. In this case, the length of `DopplerSpectrum` must be equal to the length of `PathDelays`.
 - To generate C code, specify `DopplerSpectrum` to a single Doppler spectrum structure.
-

View and Change Doppler Spectrum Structure Properties

Create a Doppler spectrum structure by specifying the type of Doppler spectrum as 'Rounded', then modify settings of the polynomial type

A rounded Doppler spectrum structure with default properties is created and displayed, and the third element of the Polynomial field is modified.

```
doppRound = doppler('Rounded')
doppRound = struct with fields:
    SpectrumType: 'Rounded'
    Polynomial: [1 -1.7200 0.7850]
```

Adjust the third coefficient of the polynomial.

```
doppRound.Polynomial(3) = 0.825
doppRound = struct with fields:
    SpectrumType: 'Rounded'
    Polynomial: [1 -1.7200 0.8250]
```

Be aware that it is possible to modify a Doppler spectrum structure to an invalid configuration. Validation of the Doppler spectrum structure settings is performed when the structure is used by a fading channel object. The `doppRound` spectrum structure defined above is valid.

```
ricianCh = comm.RicianChannel('DopplerSpectrum',doppRound)
ricianCh =
    comm.RicianChannel with properties:
        SampleRate: 1
        PathDelays: 0
        AveragePathGains: 0
        NormalizePathGains: true
        KFactor: 3
        DirectPathDopplerShift: 0
        DirectPathInitialPhase: 0
        MaximumDopplerShift: 1.0000e-03
        DopplerSpectrum: [1x1 struct]
```

Show all properties

Use Doppler Spectrum Structures Within Channel Objects

The `DopplerSpectrum` property of a channel object can be changed by assigning to it a Doppler spectrum structure or a vector of Doppler spectrum structures.

Create Rayleigh Channel with Flat Doppler Spectrum

This example illustrates how to change the default Jakes Doppler spectrum of a constructed Rayleigh channel object to a flat Doppler spectrum.

Create a Rayleigh channel object

Set the sample rate to 9600 Hz and the maximum Doppler shift to 100 Hz.

```
rayChan = comm.RayleighChannel('SampleRate',9600, 'MaximumDopplerShift',100)
```

```
rayChan =  
    comm.RayleighChannel with properties:
```

```
        SampleRate: 9600  
        PathDelays: 0  
    AveragePathGains: 0  
    NormalizePathGains: true  
MaximumDopplerShift: 100  
    DopplerSpectrum: [1x1 struct]
```

```
Show all properties
```

```
rayChan.DopplerSpectrum
```

```
ans = struct with fields:  
    SpectrumType: 'Jakes'
```

Modify the Doppler spectrum

Create a flat Doppler spectrum structure and assign it in the `rayChan` object.

```
doppFlat = doppler('Flat')
```

```
doppFlat = struct with fields:  
    SpectrumType: 'Flat'
```

```
rayChan.DopplerSpectrum = doppFlat
```

```
rayChan =
  comm.RayleighChannel with properties:
```

```
    SampleRate: 9600
    PathDelays: 0
  AveragePathGains: 0
  NormalizePathGains: true
  MaximumDopplerShift: 100
  DopplerSpectrum: [1x1 struct]
```

```
Show all properties
```

```
rayChan.DopplerSpectrum
```

```
ans = struct with fields:
  SpectrumType: 'Flat'
```

Create Rician Channel with Gaussian Doppler Spectrum

This example shows how to change the default Jakes Doppler spectrum of a constructed Rician channel object to a Gaussian Doppler spectrum with normalized standard deviation of 0.3, and subsequently display the `DopplerSpectrum` property, and change the value of the normalized standard deviation to 1.1.

Create a Rician channel object

Set the sample rate to 9600 Hz, the maximum Doppler shift to 100 Hz, and K factor to 2.

```
ricChan = comm.RicianChannel('SampleRate',9600, 'MaximumDopplerShift',100, 'KFactor',2)
```

```
ricChan =
  comm.RicianChannel with properties:
```

```
    SampleRate: 9600
    PathDelays: 0
  AveragePathGains: 0
  NormalizePathGains: true
    KFactor: 2
  DirectPathDopplerShift: 0
  DirectPathInitialPhase: 0
  MaximumDopplerShift: 100
  DopplerSpectrum: [1x1 struct]
```

Show all properties

```
ricChan.DopplerSpectrum
```

```
ans = struct with fields:  
    SpectrumType: 'Jakes'
```

Modify the Doppler spectrum

Create a Gaussian Doppler spectrum structure with normalized standard deviation of 0.3 and assign it in the `ricChan` object.

```
doppGaus = doppler('Gaussian',0.3)
```

```
doppGaus = struct with fields:  
    SpectrumType: 'Gaussian'  
    NormalizedStandardDeviation: 0.3000
```

```
ricChan.DopplerSpectrum = doppGaus
```

```
ricChan =  
    comm.RicianChannel with properties:  
  
        SampleRate: 9600  
        PathDelays: 0  
        AveragePathGains: 0  
        NormalizePathGains: true  
        KFactor: 2  
        DirectPathDopplerShift: 0  
        DirectPathInitialPhase: 0  
        MaximumDopplerShift: 100  
        DopplerSpectrum: [1x1 struct]
```

Show all properties

```
ricChan.DopplerSpectrum
```

```
ans = struct with fields:  
    SpectrumType: 'Gaussian'  
    NormalizedStandardDeviation: 0.3000
```



```
ricChan.DopplerSpectrum.NormalizedStandardDeviation = 1.1;
ricChan.DopplerSpectrum

ans = struct with fields:
    SpectrumType: 'Gaussian'
    NormalizedStandardDeviation: 1.1000
```

Create Rayleigh Channel Using Independent Doppler Spectrum

This example illustrates how to change the default Jakes Doppler spectrum of a constructed three-path Rayleigh channel object to a cell array of different Doppler spectra, and then change the properties of the Doppler spectrum of the third path.

Create a Rayleigh channel object

Set the sample rate to 9600 Hz, the maximum Doppler shift to 100 Hz, and specify path delays of 0, 1e-4, and 2.1e-4 seconds.

```
rayChan = comm.RayleighChannel('SampleRate',9600, 'MaximumDopplerShift',100, 'PathDelays',[0 1e-4 2.1e-4])

rayChan =
    comm.RayleighChannel with properties:
        SampleRate: 9600
        PathDelays: [0 1.0000e-04 2.1000e-04]
        AveragePathGains: 0
        NormalizePathGains: true
        MaximumDopplerShift: 100
        DopplerSpectrum: [1x1 struct]

Show all properties
```

```
rayChan.DopplerSpectrum

ans = struct with fields:
    SpectrumType: 'Jakes'
```

Modify the Doppler spectrum

Specify the `DopplerSpectrum` property as a cell array with an independent Doppler spectrum for each path.

```
rayChan.DopplerSpectrum = {doppler('Flat') doppler('Flat') doppler('Rounded')}
```

```
rayChan =  
  comm.RayleighChannel with properties:  
  
      SampleRate: 9600  
      PathDelays: [0 1.0000e-04 2.1000e-04]  
      AveragePathGains: 0  
      NormalizePathGains: true  
      MaximumDopplerShift: 100  
      DopplerSpectrum: {[1x1 struct] [1x1 struct] [1x1 struct]}
```

Show all properties

```
rayChan.DopplerSpectrum{:}
```

```
ans = struct with fields:  
  SpectrumType: 'Flat'
```

```
ans = struct with fields:  
  SpectrumType: 'Flat'
```

```
ans = struct with fields:  
  SpectrumType: 'Rounded'  
  Polynomial: [1 -1.7200 0.7850]
```

Change the **Polynomial** property for the third path.

```
rayChan.DopplerSpectrum{3}.Polynomial = [1 -1.21 0.7]
```

```
rayChan =  
  comm.RayleighChannel with properties:  
  
      SampleRate: 9600  
      PathDelays: [0 1.0000e-04 2.1000e-04]  
      AveragePathGains: 0  
      NormalizePathGains: true  
      MaximumDopplerShift: 100  
      DopplerSpectrum: {[1x1 struct] [1x1 struct] [1x1 struct]}
```

Show all properties

```
rayChan.DopplerSpectrum{:}
```

```
ans = struct with fields:
  SpectrumType: 'Flat'

ans = struct with fields:
  SpectrumType: 'Flat'

ans = struct with fields:
  SpectrumType: 'Rounded'
  Polynomial: [1 -1.2100 0.7000]
```

Configure Channel Objects

Before you filter a signal using a channel object, make sure that the properties of the channel have suitable values for the situation you want to model. This section offers some guidelines to help you choose realistic values that are appropriate for your modeling needs. The topics are

- “Choose Realistic Channel Property Values” on page 14-29
- “Configure Channel Objects Based on Simulation Needs” on page 14-31

The syntaxes for viewing and changing values of properties of channel objects are described in Specifying a Fading Channel on page 14-17.

Choose Realistic Channel Property Values

Here are some tips for choosing property values that describe realistic channels:

Path Delays

- By convention, the first delay is typically set to zero. The first delay corresponds to the first arriving path.
- For indoor environments, path delays after the first are typically between 1e-9 seconds and 1e-7 seconds.
- For outdoor environments, path delays after the first are typically between 1e-7 seconds and 1e-5 seconds. Large delays in this range might correspond, for example, to an area surrounded by mountains.
- The ability of a signal to resolve discrete paths is related to its bandwidth. If the difference between the largest and smallest path delays is less than about 1% of the

symbol period, then the signal experiences the channel as if it had only one discrete path.

Average Path Gains

- The average path gains in the channel object indicate the average power gain of each fading path. In practice, an average path gain value is a large negative dB value. However, computer models typically use average path gains in the range of [-20, 0] dB.
- The dB values in a vector of average path gains often decay roughly linearly as a function of delay, but the specific delay profile depends on the propagation environment.
- To ensure that the expected total power of the combined path gains is 1, you can normalize path gains via the `NormalizePathGains` property of the channel object.

Maximum Doppler Shifts

- Some wireless applications, such as standard GSM (Global System for Mobile Communication) systems, prefer to specify Doppler shifts in terms of the speed of the mobile. If the mobile moves at speed v (m/s), then the maximum Doppler shift is calculated as follows, where f is the transmission carrier frequency in Hertz and c is the speed of light (3e8 m/s).

$$f_d = \frac{vf}{c}$$

- Applying this formula at a transmission carrier frequency of 900 MHz. A signal from a moving car on a freeway might experience a maximum Doppler shift of about 80 Hz. A signal from a moving pedestrian might experience a maximum Doppler shift of about 4 Hz.
- A maximum Doppler shift of 0 corresponds to a static channel that comes from a Rayleigh or Rician distribution.

K-Factor for Rician Fading Channels

- The Rician K-factor specifies the ratio of specular-to-diffuse power for a direct line-of-sight path. The ratio is expressed linearly, not in dB.
- For Rician fading, the K-factor is typically in the range [1, 10].
- A K-factor of 0 corresponds to Rayleigh fading.

Doppler Spectrum Parameters

- See the `doppler` reference page for the respective Doppler spectrum structures for descriptions of the parameters and their significance.

Configure Channel Objects Based on Simulation Needs

Tips for configuring a channel object to customize the filtering process:

- If your data is partitioned into a series of vectors (that you process within a loop, for example), you can call the channel object multiple times (in each iteration within a loop). The state information of the channel is updated and saved after each invocation. The channel output is irrelevant to how the data is partitioned (vector length).
- If you want the channel output to be repeatable, choose the `seed` option for the `RandomStream` property of the channel object. To repeat the output, call the `reset` object function to reset both the internal filters and the internal random number generator.
- If you want to model discontinuously transmitted data, set the `FadingTechnique` property to `'Sum of sinusoids'` and the `InitialTimeSource` property to `'Input port'` for the channel object. When calling the object, specify the start time of each data vector/frame to be processed by the channel via an input.
- If you want to normalize the fading process so that the expected value of the path gains' total power is 1 (the channel does not contribute additional power gain or loss), set the `NormalizePathGains` property of the channel object to `true`.

Use Fading Channels

After you have created a channel object as described in [Specifying a Fading Channel](#) on page 14-17, you can call the object to pass a signal through the channel. Provide the signal as an input argument to the channel object. At the end of the filtering operation, the channel object retains its state so that you can find out the final path gains or the total number of samples that the channel has processed by calling the `info` object function with the object as the input argument.

For an example that illustrates the basic syntax and state retention, see [Power of a Faded Signal](#) on page 14-34.

To visualize the characteristics of the channel, set the `Visualization` property to `'Impulse response'`, `'Frequency response'`, or `'Doppler spectrum'`. For more information, see [Channel Visualization](#) on page 17-29.

Visualize Three-Path Rayleigh Channel

This example shows you how to visualize impulse response of a channel.

Create a channel object

While creating the channel object, use name-value pairs to set the `Visualization` property

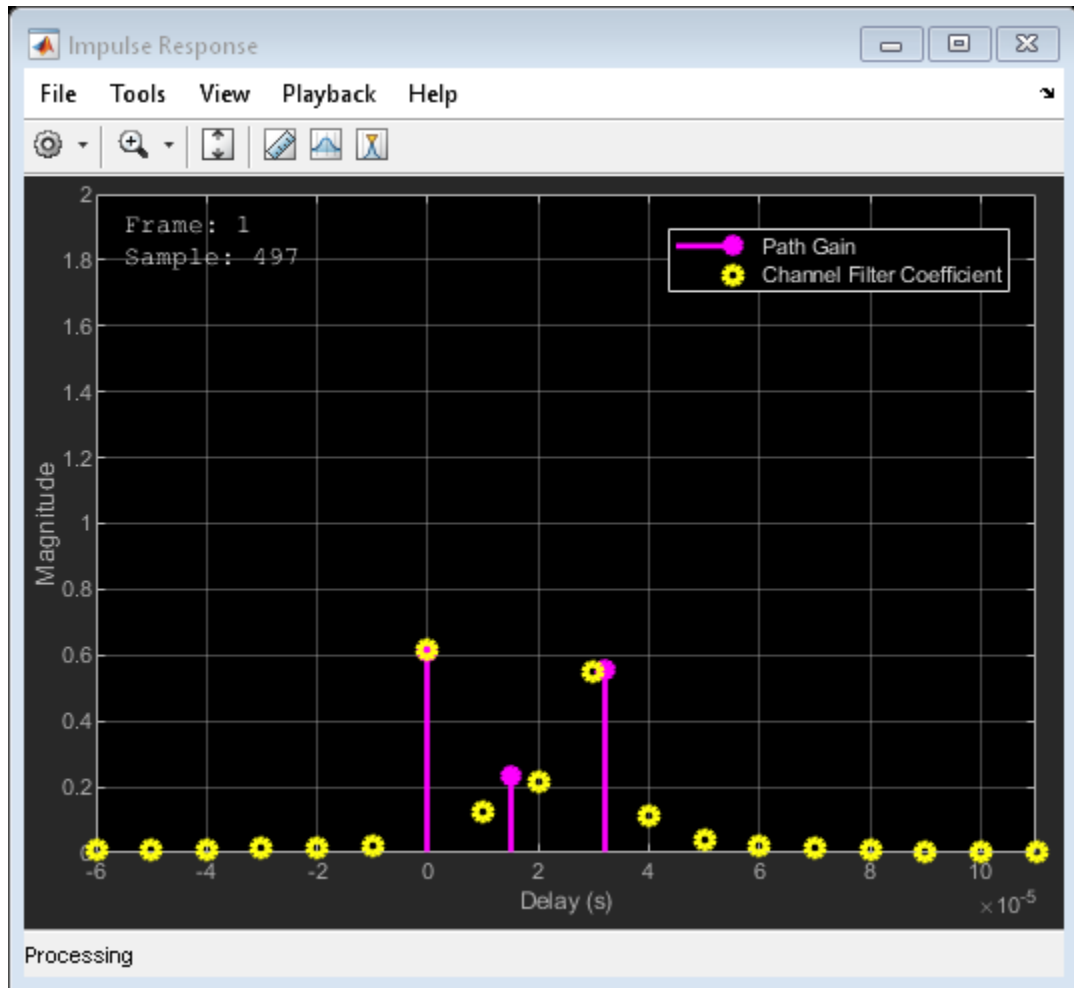
to 'Impulse response'.

```
rayChan = comm.RayleighChannel('SampleRate',100000,'MaximumDopplerShift',130,...  
    'PathDelays',[0 1.5e-5 3.2e-5],'AveragePathGains',[0, -3, -3],...  
    'Visualization','Impulse response');
```

Generate a bit stream and create a modulator object. Modulate the bit stream and pass the

modulated DBPSK signal through the channel by calling the channel object.

```
tx = randi([0 1],500,1);  
dbpskMod = comm.DBPSKModulator;  
dpskSig = dbpskMod(tx);  
y = rayChan(dpskSig);
```



The impulse response is plotted when the object is called.

Rayleigh Fading Channel

These examples use fading channels:

- "Power of Faded Signal" on page 14-34

- “DBPSK Empirical Versus Theoretical Performance in Fading Conditions” on page 14-35
- “Work with Channel Filter Delays” on page 14-37
- “Channel Filtering Using a For Loop” on page 14-38

Power of Faded Signal

This example plots the power of a faded signal versus sample number. It illustrates the syntax of creating and calling a `comm.RayleighChannel` fading channel object and the state retention of the channel object.

```
rayChan = comm.RayleighChannel('SampleRate',10000,'MaximumDopplerShift',100);  
sig = j*ones(2000,1); % Signal  
out = rayChan(sig); % Pass signal through channel.  
rayChan % Display all properties of the channel object.
```

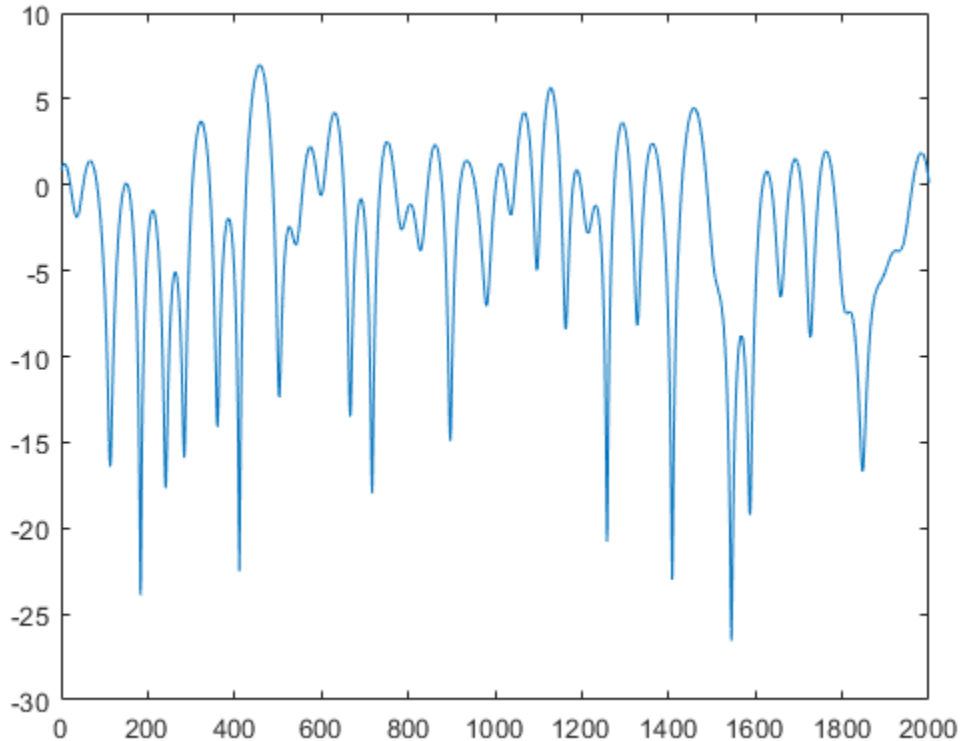
```
rayChan =  
comm.RayleighChannel with properties:
```

```
    SampleRate: 10000  
    PathDelays: 0  
    AveragePathGains: 0  
    NormalizePathGains: true  
    MaximumDopplerShift: 100  
    DopplerSpectrum: [1x1 struct]
```

```
Show all properties
```

Plot power of faded signal, versus sample number.

```
plot(20*log10(abs(out)))
```

DBPSK Empirical Versus Theoretical Performance in Fading Conditions

This example creates a frequency-flat Rayleigh fading channel object and calls it to process a DBPSK signal consisting of a single vector. Bit error rate is computed for different values of the signal-to-noise ratio. The fading channel filter is applied before AWGN. This is the recommended sequence to use when you combine fading with AWGN.

Create Rayleigh fading channel, modulator, and demodulator objects

```
chan = comm.RayleighChannel('SampleRate',1e4,'MaximumDopplerShift',100);
```

Create DBPSK modulator and demodulator objects with the modulation order set to 2. Generate DBPSK modulated data and pass it through the channel.

```
M = 2; % DBPSK modulation order
tx = randi([0 M-1],50000,1); % Generate a random bit stream

mod = comm.DBPSKModulator;
demod = comm.DBPSKDemodulator;
```

```
dpskSig = mod(tx);
fadedSig = chan(dpskSig); % Apply the channel effects
```

Create an AWGN channel object and an error rate calculator object.

```
awgnChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)');
errorCalc = comm.ErrorRate;
```

Compute error rate for different values of SNR.

```
SNR = 0:2:20; % Range of SNR values, in dB.
numSNR = length(SNR);
berVec = zeros(3, numSNR); % Preallocate a vector for BER results
```

```
for n = 1:numSNR
    awgnChan.SNR = SNR(n);
    rxSig = awgnChan(fadedSig); % Add Gaussian noise
    rx = demod(rxSig); % Demodulate
    reset(errorCalc)

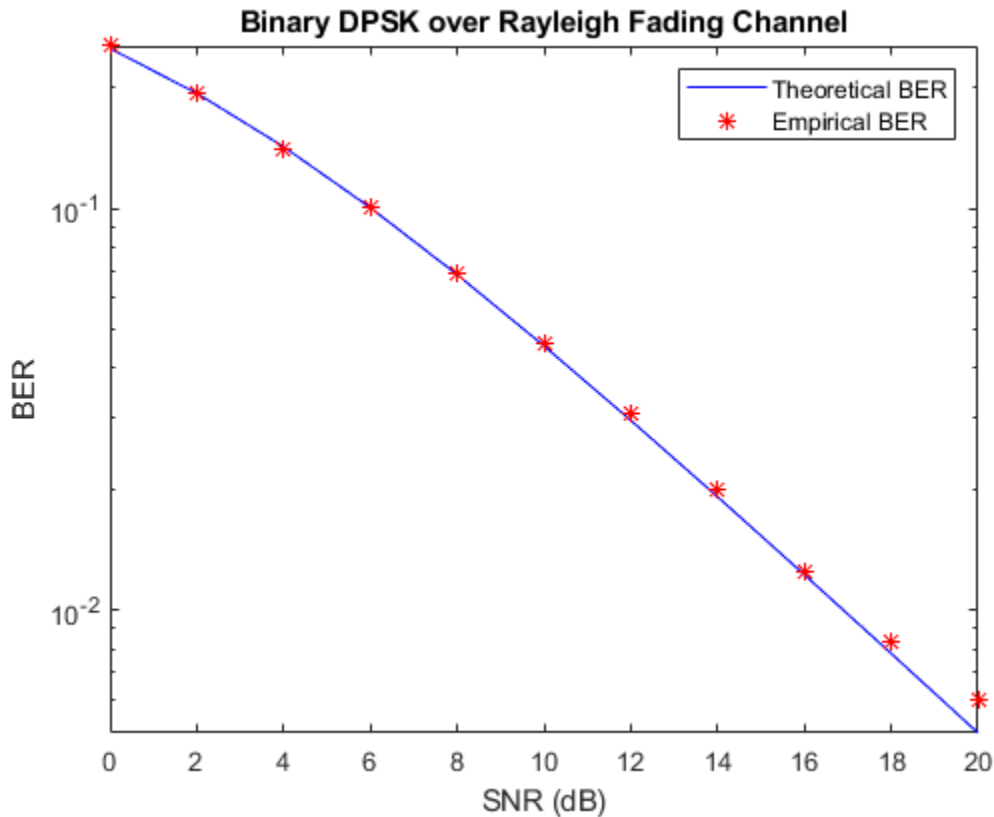
    berVec(:,n) = errorCalc(tx,rx); % Compute error rate.
end
BER = berVec(1,:);
```

Compute theoretical performance results, for comparison.

```
BERtheory = berfading(SNR, 'dpsk', M, 1);
```

Plot BER results.

```
semilogy(SNR, BERtheory, 'b-', SNR, BER, 'r*');
legend('Theoretical BER', 'Empirical BER');
xlabel('SNR (dB)'); ylabel('BER');
title('Binary DPSK over Rayleigh Fading Channel');
```



Work with Channel Filter Delays

The value of a channel object's `ChannelFilterDelay` property is the number of samples by which the output of the channel lags the input. If you compare the input and output data sets directly, you must take the delay into account by using appropriate truncating or padding operations.

The example illustrates a way to account for the delay before computing a bit error rate.

Create DBPSK modulator and demodulator objects with the modulation order set to 2. Generate DBPSK modulated data and pass it through the channel.

```
bitRate = 50000;
M = 2; % DQPSK modulation order
```

```
mod = comm.DBPSKModulator;  
demod = comm.DBPSKDemodulator;
```

Create Rayleigh fading channel object.

```
rayChan = comm.RayleighChannel('SampleRate',bitRate,'MaximumDopplerShift',4,...  
    'PathDelays',[0 0.5/bitRate],'AveragePathGains',[0 -10]);  
chInfo = info(rayChan);  
delay = chInfo.ChannelFilterDelay;
```

Generate random bit stream data. Modulate the data, pass it through the fading channel, and demodulate it.

```
tx = randi([0 M-1],50000,1);  
  
dpskSig = mod(tx);  
fadedSig = rayChan(dpskSig);  
rx = demod(fadedSig);
```

Compute bit error rate, taking delay into account.

```
errorCalc = comm.ErrorRate('ReceiveDelay', delay);  
berVec = step(errorCalc,tx,rx);  
ber = berVec(1)  
  
ber = 0.0147  
  
num = berVec(2)  
  
num = 737
```

Channel Filtering Using a For Loop

This example filters input data through a Rayleigh fading channel within a for loop. It uses the small data sets from successive iterations to create an animated effect. The Rayleigh fading channel has two discrete major paths. For information on filtering data through a channel multiple times while maintaining continuity from one invocation to the next, see “Configure Channel Objects Based on Simulation Needs” on page 14-31.

Set up parameters. Specify a bit rate of 50e3 Hz, and a loop iteration count of 125. Create a QPSK modulator and Rayleigh fading channel objects.

```
bitRate = 50000;    % Data rate is 50 kb/s  
numTrials = 125;   % Number of iterations of loop
```

```
M = 4; % QPSK modulation order
qpskMod = comm.QPSKModulator;

rayChan = comm.RayleighChannel('SampleRate',bitRate,'MaximumDopplerShift',4,'PathDelay');

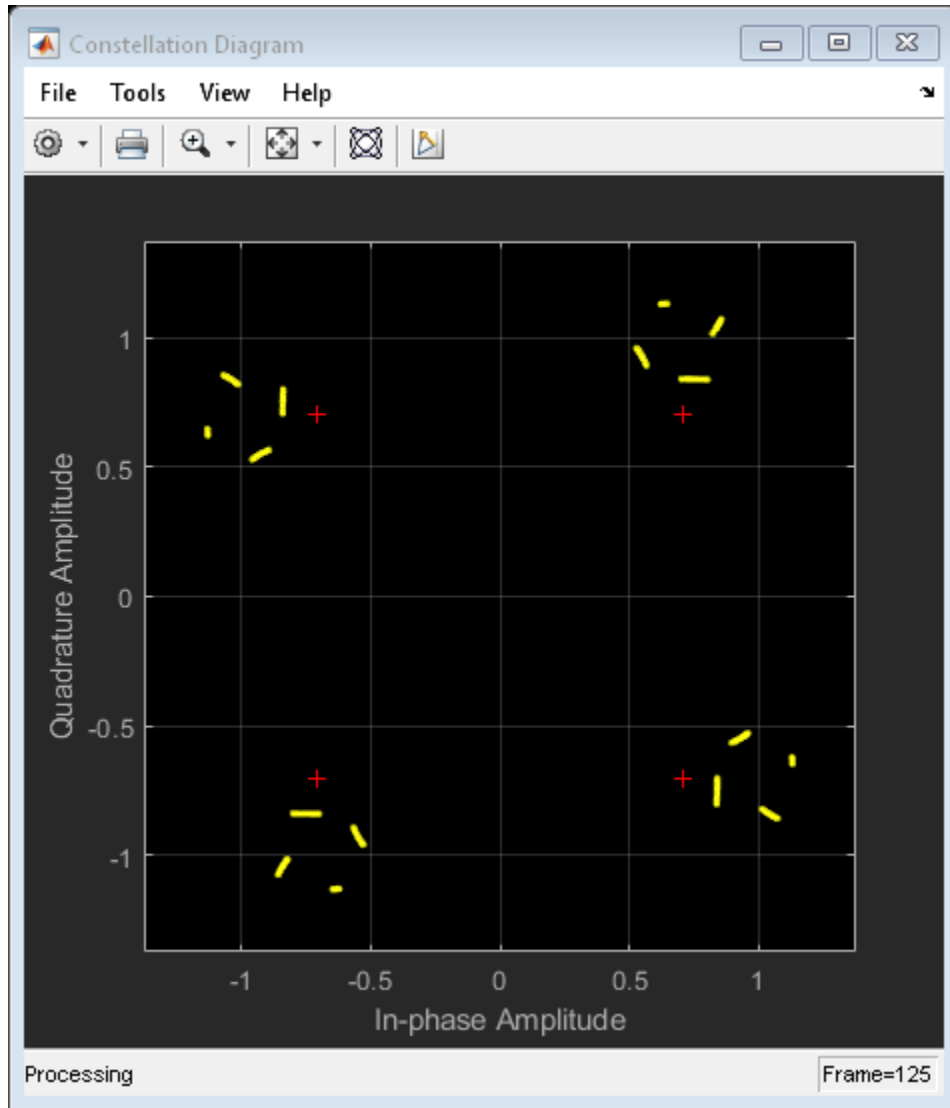
Initialize a scatter plot.

scatterPlot = comm.ConstellationDiagram;

Apply channel in a loop, maintaining continuity. Plot only the current data in each
iteration.

for n = 1:numTrials
    tx = randi([0 M-1],500,1); % Generate random bit stream
    pskSig = qpskMod(tx); % PSK modulate signal
    fadedSig = rayChan(pskSig); % Apply channel effects

    % Plot the new data from this iteration.
    update(scatterPlot,fadedSig);
end
```



Rician Fading Channel

Quasi-Static Channel Modeling

Typically, a path gain in a fading channel changes insignificantly over a period of $1/$

($100fd$) seconds, where fd is the maximum Doppler shift. Because this period corresponds to a very large number of bits in many modern wireless data applications, assessing performance over a statistically significant range of channel fading requires simulating a prohibitively large amount of data. This example illustrates quasi-static channel modeling approach to gathering a statistically significant number of errors. Quasi-static channel modeling provides a more tractable approach, which you can implement using these steps:

- 1 Generate a random channel realization using a maximum Doppler shift of 0.
- 2 Process some large number of bits.
- 3 Compute error statistics.
- 4 Repeat these steps many times to produce a distribution of the performance metric.

Create modulator and demodulator system objects with a modulation order of 4.

```
M = 4;
dpskMod = comm.DPSKModulator('ModulationOrder', M);
dpskDemod = comm.DPSKDemodulator('ModulationOrder', M);
numBits = 10000; % Each trial uses 10000 bits.
numTrials = 20; % Number of BER computations
```

Note: In reality, `numTrials` would be a large number to get an accurate estimate of outage probabilities or packet error rate. Use 20 here just to make the example run more quickly.

Create a Rician channel object and set the maximum Doppler shift to zero.

```
ricianChan = comm.RicianChannel('KFactor',3,'MaximumDopplerShift',0);
```

Within a `for` loop, generate a random bit stream, DPSK modulate the signal, filter the modulated signal through a Rician fading and AWGN channels, and demodulate the faded signal.

```
nErrors = zeros(1,numTrials);
for n = 1:numTrials
    tx = randi([0 M-1],numBits,1); % Generate random bit stream
    dpskSig = dpskMod(tx); % DPSK modulate signal
    fadedSig = ricianChan(dpskSig); % Apply channel effects
    rxSig = awgn(fadedSig,15,'measured'); % Add Gaussian noise.
    rx = dpskDemod(rxSig); % Demodulate.
```

Compute number of symbol errors. Ignore first sample because of DPSK initial condition.

```
        nErrors(n) = symerr(tx(2:end),rx(2:end))
    end
nErrors = 1x20
    0    0    0    0    0    0    0    0    0    0    0    0    0    0
nErrors = 1x20
    0    0    0    0    0    0    0    0    0    0    0    0    0    0
nErrors = 1x20
    0    0    1    0    0    0    0    0    0    0    0    0    0    0
nErrors = 1x20
    0    0    1    0    0    0    0    0    0    0    0    0    0    0
nErrors = 1x20
    0    0    1    0    0    0    0    0    0    0    0    0    0    0
nErrors = 1x20
    0    0    1    0    0    0    0    0    0    0    0    0    0    0
nErrors = 1x20
    0    0    1    0    0    0    0    0    0    0    0    0    0    0
nErrors = 1x20
    0    0    1    0    0    0    0    0    0    0    0    0    0    0
```


nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

0 0 1 0 0 0 0 0 0 0 0 0 0 0

nErrors = 1x20

```

0 0 1 0 0 0 0 0 0 0 0 0 0 0
nErrors = 1x20
0 0 1 0 0 0 0 0 0 0 0 0 0 0

```

While the example runs, the Command Window displays the growing list of symbol error counts in the vector `nErrors`. It also displays the packet error rate at the end. The sample output below shows a final value of `nErrors` and omits intermediate values. Your results might vary because of randomness in the example.

```

per = mean(nErrors > 0) % Proportion of packets that had errors
per = 0.0500

```

More About the Quasi-Static Technique

As an example to show how the quasi-static channel modeling approach can save computation, consider a wireless local area network (LAN) in which the carrier frequency is 2.4 GHz, mobile speed is 1 m/s, and bit rate is 10 Mb/s. The following expression shows that the channel changes insignificantly over 12,500 bits:

$$\begin{aligned}
 \left(\frac{1}{100f_d} \text{ s}\right)(10 \text{ Mb/s}) &= \left(\frac{c}{100vf} \text{ s}\right)(10 \text{ Mb/s}) \\
 &= \frac{3 \times 10^8 \text{ m/s}}{100(1 \text{ m/s})(2.4 \text{ GHz})}(10 \text{ Mb/s}) \\
 &= 12,500 \text{ b}
 \end{aligned}$$

A traditional Monte Carlo approach for computing the error rate of this system would entail simulating thousands of times, totalling tens of millions of bits. By contrast, a quasi-static channel modeling approach would simulate a few packets at each of about 100 locations to arrive at a spatial distribution of error rates. From this distribution one could determine, for example, how reliable the communication link is for a random location within the indoor space. If each simulation contains 5,000 bits, 100 simulations would process half a million bits in total. This is substantially fewer bits compared to the traditional Monte Carlo approach.

Additional Examples Using Fading Channels

These models include the use of fading channels:

- Rayleigh Fading Channel, which illustrates the channel's effect on a QPSK modulated signal
- IEEE 802.11a WLAN Physical Layer
- Defense Communications: US MIL-STD-188-110B
- WCDMA End-to-End Physical Layer

Using Channel Visualization

Communications Toolbox channel objects include a `Visualization` property that enables you to visualize the characteristics of a fading channel when calling the channel object. For more information, see [Channel Visualization](#) on page 17-29.

MIMO Channel

The Communications Toolbox software provides a Multiple Input Multiple Output (MIMO) Multipath Fading Channel System object and block. Multipath MIMO fading channels allow for design of communications systems with multiple antenna elements at the transmitter and receiver.

For more information, see the `comm.MIMOChannel` reference page and MIMO Fading Channel.

The following demos illustrate MIMO fading channel techniques using MATLAB System objects:

- Introduction to MIMO Systems
- IEEE 802.16 Channel Models

WINNER II Channel

The WINNER II Channel Model for Communications Toolbox builds on the WINNER II open source download, WINNER II Channel Models on page 14-48, by generating channel coefficients, which enables channel filtering. The WINNER II Channel Model also provides all the capability currently available in the open source download, including:

- Dipole antenna pattern generation
- Antenna array response calculation
- Antenna array configuration
- Layout configuration
- Spatial system configuration
- Channel path gain generation

To model a WINNER II Channel, first download the WINNER II Channel Model for Communications Toolbox from the Add-On Explorer. For more information on downloading add-ons, see [Get Add-Ons \(MATLAB\)](#).

References

- [1] Kyosti, Pekka, Juha Meinila, et al. *WINNER II Channel Models*. D1.1.2 V1.2. IST-4-027756 WINNER II, September 2007.

See Also

More About

- [Mapping WINNER II Public Download to WINNER2Channel on page 14-49](#)
- [Simultaneous Simulation of Multiple Fading Channels with WINNER II Channel Model](#)
- [802.11ac Multi-User MIMO Precoding with WINNER II Channel Model](#)

Mapping of WINNER II Open Source Download to WINNER II Channel Model for Communications Toolbox

The WINNER II Channel Model for Communications Toolbox is composed of the `comm.WINNER2Channel` System object and functions. The functions in the WINNER II Channel Model for Communications Toolbox map to functions in the WINNER II open source download.

Function in WINNER II Open Source Download	Corresponding Function in WINNER II Channel Model for Communications Toolbox
AntennaArray	<code>winner2.AntennaArray</code>
AntennaResponse	<code>winner2.internal.calcAntennaResponse</code>
arrayparset	<code>winner2.arrayparset</code>
dipole	<code>winner2.dipole</code>
layoutparset	<code>winner2.layoutparset</code>
layout2link	<code>winner2.internal.layout2Link</code>
wim	<code>winner2.wim</code>
wimparset	<code>winner2.wimparset</code>

The following table shows the behavioral changes between the WINNER II open source download and the WINNER II Channel Model for Communications Toolbox.

Behavioral Condition	WINNER II Public Download Behavior	WINNER II Channel Model for Communications Toolbox Behavior
Default value of the <code>SampleDensity</code> field in the structure returned by the <code>winner2.wimparset</code> function	2	2e6

Behavioral Condition	WINNER II Public Download Behavior	WINNER II Channel Model for Communications Toolbox Behavior
Default velocity returned for each MS by <code>winner2.layoutparset</code> .	10 m/s	1.43 m/s Corresponding to approximate typical walking speed, V_{MS} . $V_{MS} = C / F_{center} \times 25 = (2.9979 \times 10^8 / 5.25 \times 10^9) \times 25 \sim 1.43 \text{ m/s}$
The length of the third dimension of the channel coefficients output of the <code>winner2.wim</code> function	Equals the maximum number of paths or maximum number of paths plus four for all links with zero padding, NaN padding, or a combination of zero and NaN padding	Equals the number of paths for the specific link
The number of paths shown in the channel coefficients and path delay outputs of the <code>winner2.wim</code> function	Mismatched for many cases when there is more than one link	Matched for each link
Strongest cluster segregation when the <code>IntraClusterDsUsed</code> field is set to 'yes'	The two strongest clusters are divided into three subclusters only for the links that have the maximum number of paths.	For each link, the two strongest clusters are divided into three subclusters.
	If the second and third strongest paths have the same power, only the single strongest cluster is divided into three subclusters.	
Updating of the <code>Phi_LOS</code> field in the third structure output of the <code>winner2.wim</code> function	Updated when the <code>IntraClusterDsUsed</code> field is set to 'no'	Updated regardless of the setting of the <code>IntraClusterDsUsed</code> field

Behavioral Condition	WINNER II Public Download Behavior	WINNER II Channel Model for Communications Toolbox Behavior
Padding of the path delay output of the <code>winner2.wim</code> function, when the rows (number of links) have fewer than the maximum number of paths	Zero padded when the <code>IntraClusterDsUsed</code> field is set to 'yes'	NaN padded
The channel coefficients calculation specified by WINNER II Channel Models on page 14-52, Equation 4.14 and Table 4-2, when the <code>IntraClusterDsUsed</code> and <code>PolarisedArrays</code> fields are set to 'yes'	Incorrect	Correct
Path loss calculation for A1 NLOS links	Incorrect when the <code>PathLossOption</code> field is set to 'CR_heavy' or 'CR_light'	Correct
The <code>Phi_LOS</code> field per step 10 on page 40 of WINNER II Channel Models on page 14-52 for the input initial condition and output final condition of <code>winner2.wim</code> , should be of size N_L -by-2 to log the phases for both VV and HH polarization each link. N_L is the number of links.	Incorrect, <code>Phi_LOS</code> is of size N_L -by-1 for VV polarization only. The phase for HH polarization is not included. This causes issues for a link with LOS path.	Correct, <code>Phi_LOS</code> is of size N_L -by-2 to log the phases for both VV and HH polarization each link.

To use `comm.WINNER2Channel`, first download the WINNER II Channel Model for Communications Toolbox from the Add-On Explorer. For more information on downloading add-ons, see Get Add-Ons (MATLAB).

References

- [1] Kyosti, Pekka, Juha Meinila, et al. *WINNER II Channel Models*. D1.1.2 V1.2. IST-4-027756 WINNER II, September 2007.

See Also

More About

- WINNER II Channel on page 14-48

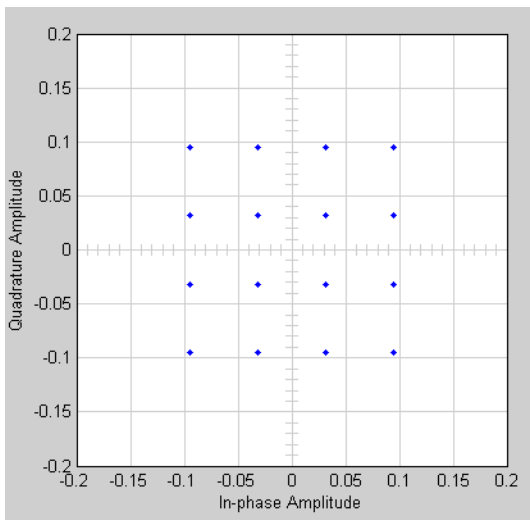
RF Impairments

In this section...

- “Illustrate RF Impairments That Distort a Signal” on page 14-53
- “Phase/Frequency Offsets and Phase Noise” on page 14-57
- “Receiver Thermal Noise and Free Space Path Loss” on page 14-57
- “Nonlinearity and I/Q Imbalance” on page 14-58
- “Apply Nonlinear Distortion to Input Signal” on page 14-58
- “View Phase Noise Effects on Signal Spectrum” on page 14-59
- “Add RF Impairments to DQPSK Signal” on page 14-63
- “Selected Bibliography for Channel Modeling” on page 14-69

Illustrate RF Impairments That Distort a Signal

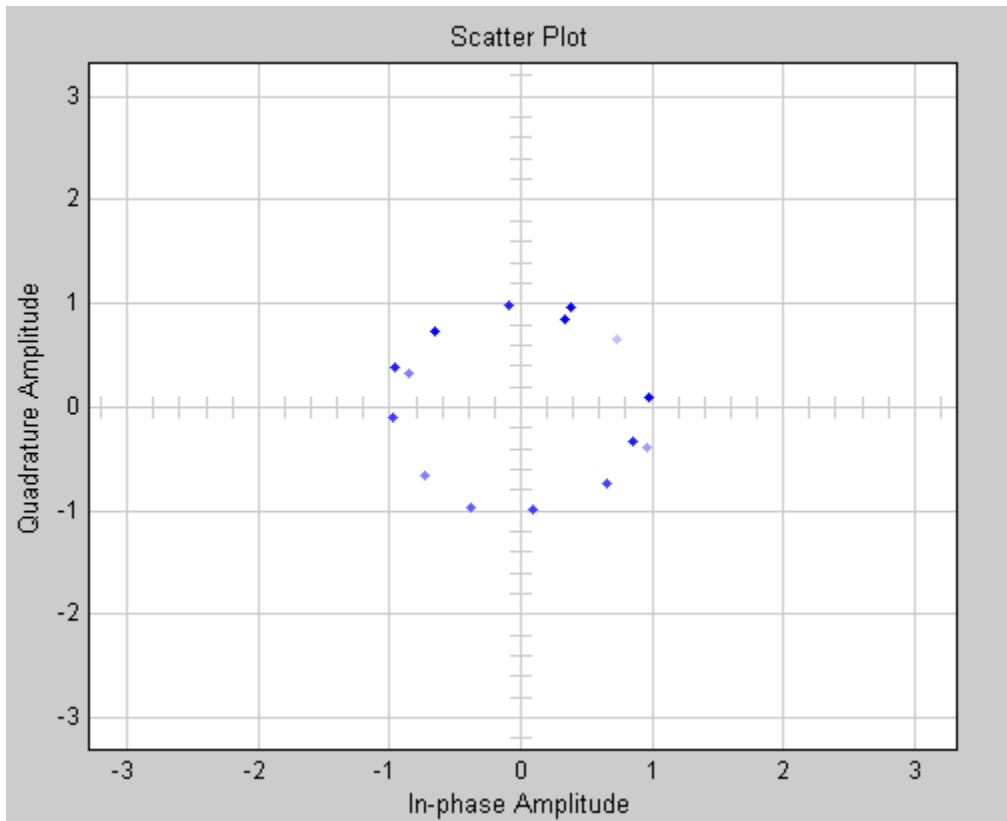
This section presents scatter plots that illustrate how blocks in the RF Impairments library distort a signal modulated by 16-ary quadrature amplitude modulation (QAM). The usual 16-ary QAM constellation without distortion is shown in the following figure.



As the scatter plots show, the first two blocks distort both the magnitude and angle of points in the constellation, while the last two alter just the angle.

You can create these scatter plots with models similar to the following, which produces the scatter plot for the Memoryless Nonlinearity block:

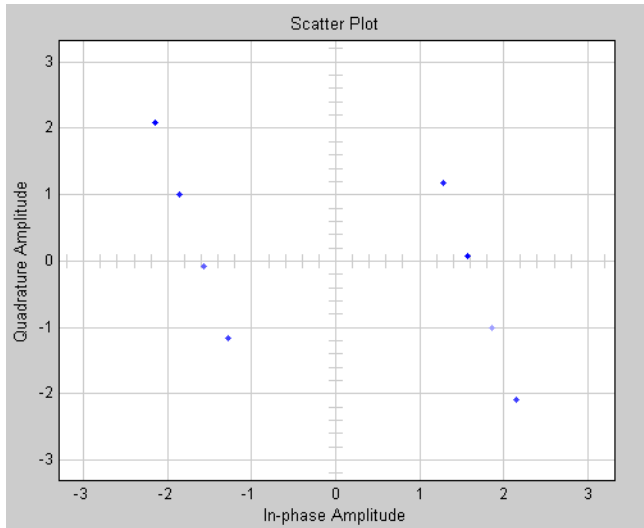
16-ary QAM Model



The model uses the Rectangular QAM Modulator Baseband block, from AM in the Digital Baseband Modulation sublibrary of the Modulation library. You control the power of the block's output signal with the **Normalization method** parameter. To open this model, enter `doc_16qam_plot` at the MATLAB command line.

I/Q Imbalance Block

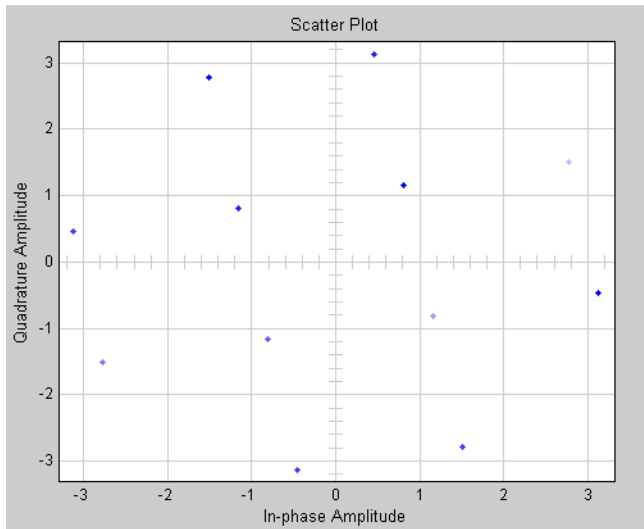
You can generate the next scatter plot by replacing the Memoryless Nonlinearity block in the 16-ary QAM Model with the I/Q Imbalance block. Set the block's **I/Q amplitude imbalance (dB)** parameter to 10 and the **I/Q phase imbalance (deg)** parameter to 30.



For more examples of scatter plots produced using this block, see the I/Q Imbalance block reference page.

Phase/Frequency Offset Block

You can generate the next scatter plot by replacing the Memoryless Nonlinearity block in the 16-ary QAM Model with the Phase/Frequency Offset block. Set the block's **Frequency offset (Hz)** parameter to 0 and the **Phase offset (deg)** parameter to 70.

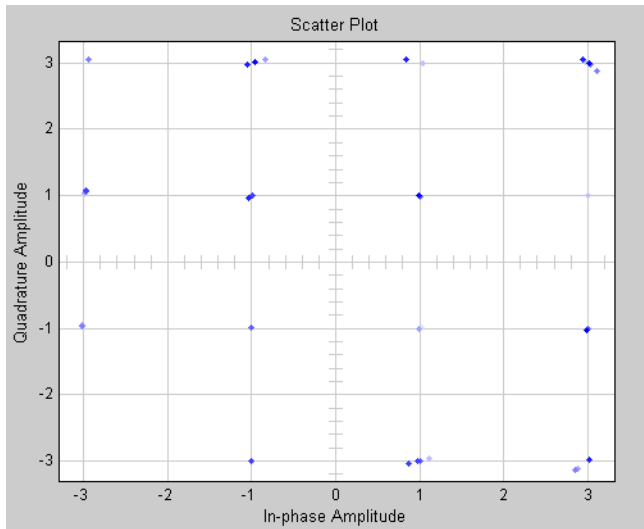


The **Frequency offset (Hz)** parameter adds a constant to the phase of the signal. The scatter plot corresponds to the standard constellation rotated by a fixed angle of 70 degrees.

The **Frequency offset (Hz)** parameter determines the rate of change of the signal's phase. In this example, **Frequency offset (Hz)** is set to 0, so the scatter plot always falls on the grid shown in the preceding figure. If you set **Frequency offset (Hz)** to a positive number, the points on the scatter plot fall on a rotating grid, corresponding to the standard constellation, which revolves at a constant rate in the counterclockwise direction. For an example, see the Phase/Frequency Offset block reference page.

Phase Noise Block

You can generate the next scatter plot by replacing the Memoryless Nonlinearity block in the 16-ary QAM Model with the Phase Noise block. Set the **Phase noise level (dBc/Hz)** parameter to -60 and the **Frequency offset (Hz)** parameter to 100.



The phase noise adds a random error to the signal's phase, so that the points in the scatter plot are spread in a radial pattern around the constellation points.

Phase/Frequency Offsets and Phase Noise

The RF Impairments library contains two blocks that simulate phase/frequency offsets and phase noise:

- The Phase/Frequency Offset block applies phase and frequency offsets to a signal.
- The Phase Noise block applies phase noise to a signal.

The Phase/Frequency Offset block and the Phase Noise block alter only the phase and frequency of the signal.

Receiver Thermal Noise and Free Space Path Loss

The RF Impairments Library contains two blocks that simulate signal impairments due to thermal noise and signal attenuation due to the distance from the transmitter to the receiver:

- The Receiver Thermal Noise block simulates the effects of thermal noise on a complex baseband signal.

- The Free Space Path Loss block simulates the loss of signal power due to the distance from the transmitter and signal frequency.

Nonlinearity and I/Q Imbalance

The following two blocks model signal impairments due to nonlinear devices or imbalances between the in-phase and quadrature components of a modulated signal:

- The Memoryless Nonlinearity block models the AM-to-AM and AM-to-PM distortion in nonlinear amplifiers.
- The I/Q Imbalance block models imbalances between the in-phase and quadrature components of a signal caused by differences in the physical channels carrying the separate components.

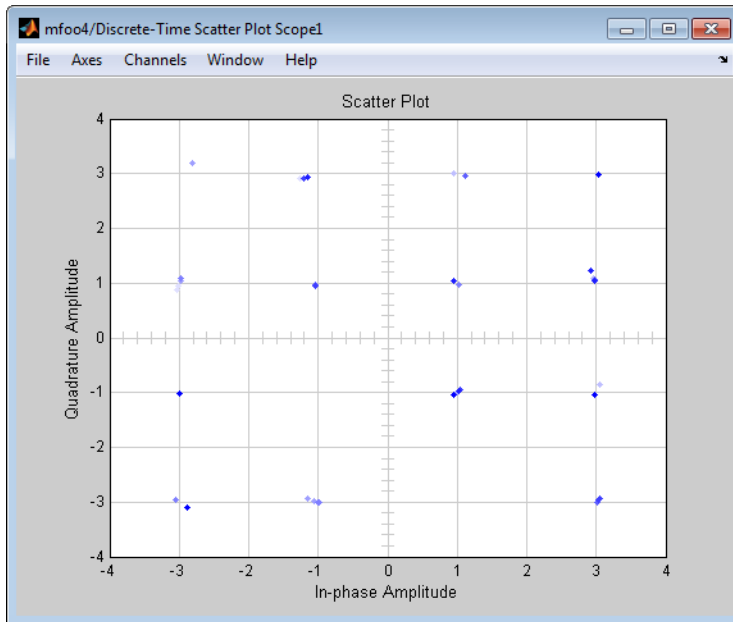
These blocks distort both the phase and amplitude of the signal.

Apply Nonlinear Distortion to Input Signal

The Memoryless Nonlinearity block applies a nonlinear distortion to the input signal. This distortion models the AM-to-AM and AM-to-PM conversions in nonlinear amplifiers. The block provides several methods, which you specify by the **Method** parameter, for modeling the nonlinear characteristics of amplifiers:

- Cubic polynomial
- Hyperbolic tangent
- Saleh model
- Ghorbani model
- Rapp model

In the model shown in the preceding figure, the **Method** parameter is set to Ghorbani model. The following figure shows the scatter plot the model generates.



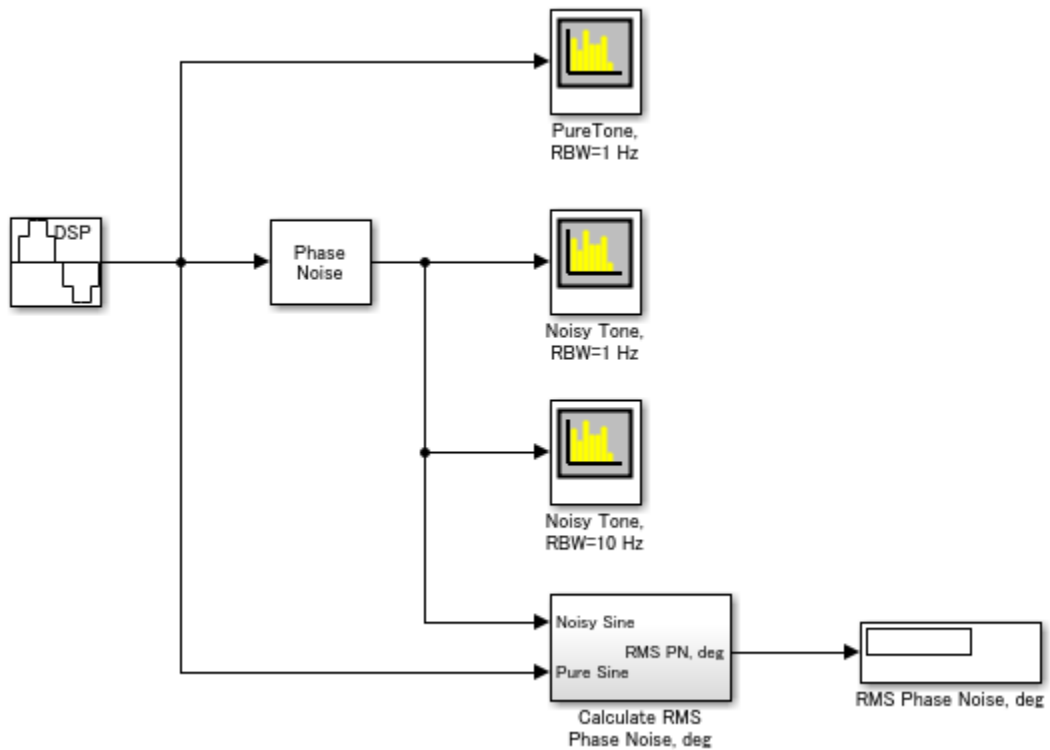
For another example of a scatter plot produced using this block, see the Memoryless Nonlinearity block reference page.

View Phase Noise Effects on Signal Spectrum

This example shows the effects that spectral and phase noise have on a 100 kHz sine wave.

Open Example Model and Explore Its Contents

Open the example model `slex_phasenoise`.



A Sine Wave block generates a 100 kHz tone. A Phase Noise block adds phase noise of:

- -85 dBc/Hz at a frequency offset of 1e3 Hz
- -118 dBc/Hz at a frequency offset of 9.5e3 Hz
- -125 dBc/Hz at a frequency offset of 19.5e3 Hz
- -145 dBc/Hz at a frequency offset of 195e3 Hz

To analyze the spectrum and phase noise, the model includes three Spectrum Analyzer blocks. The Spectrum Analyzer blocks use the default Hann windowing setting, the units are set to dBW/Hz, and the number of spectral averages is set to 10.

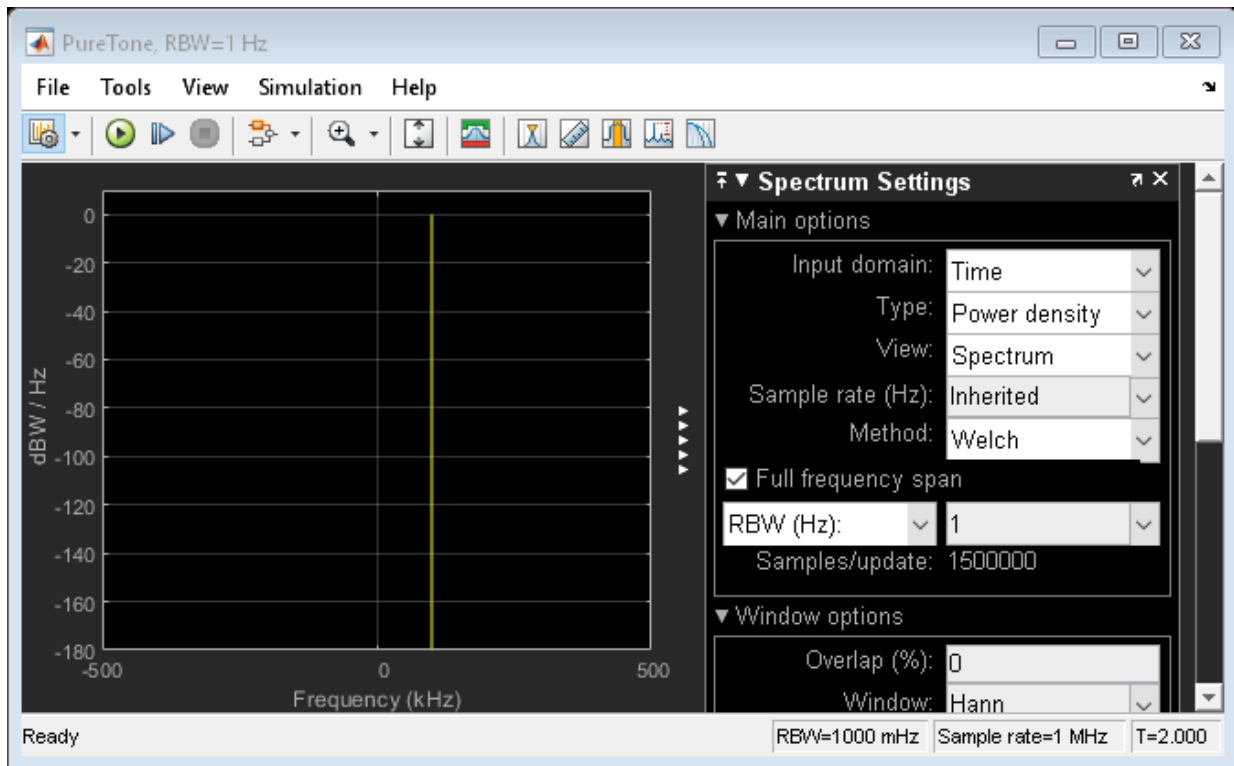
Additionally, the model includes blocks that calculate and display the RMS phase noise. The subsystem that calculates the RMS phase noise finds the phase error between the pure and noisy sine waves, then calculates the RMS phase noise in degrees. In general, to

accurately determine the phase error, the pure signal must be time aligned with the noisy signal. However, the periodicity of the sine wave in this model makes this step unnecessary.

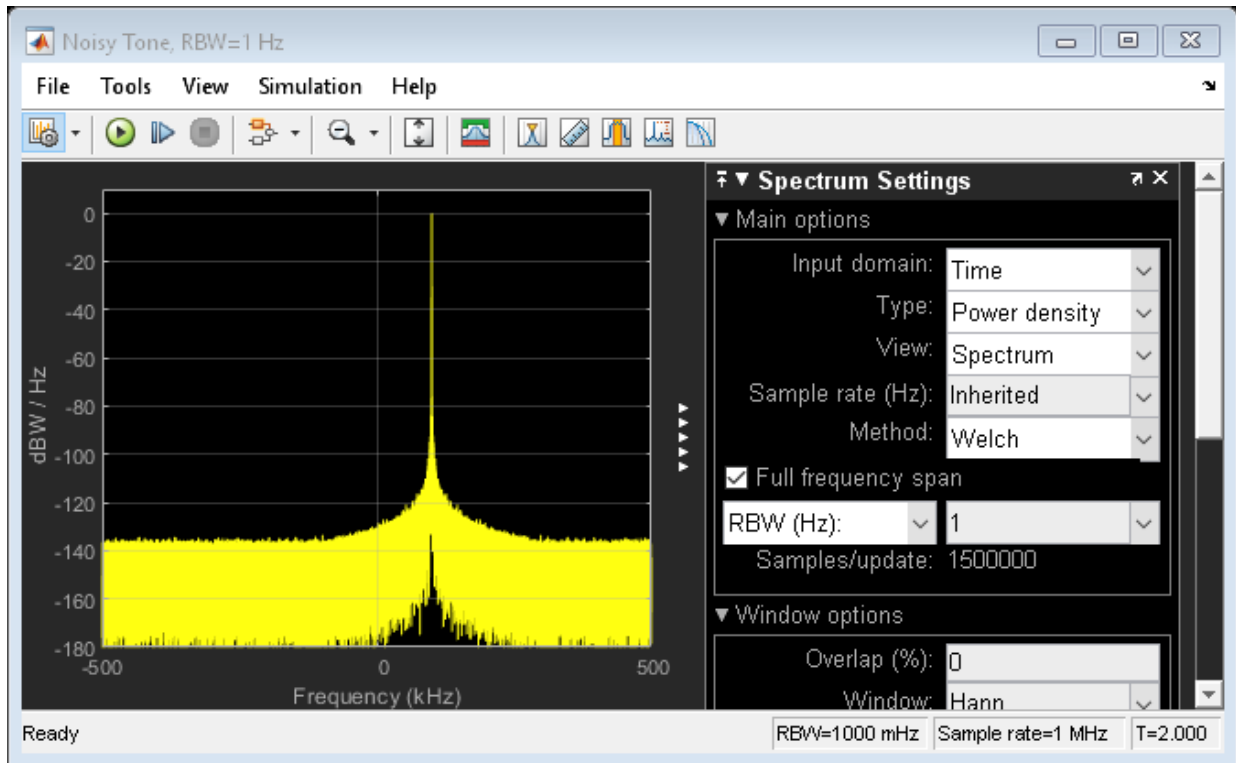
Run the Model to Generate Results

In the Simulink Editor, click **Run** to simulate the model.

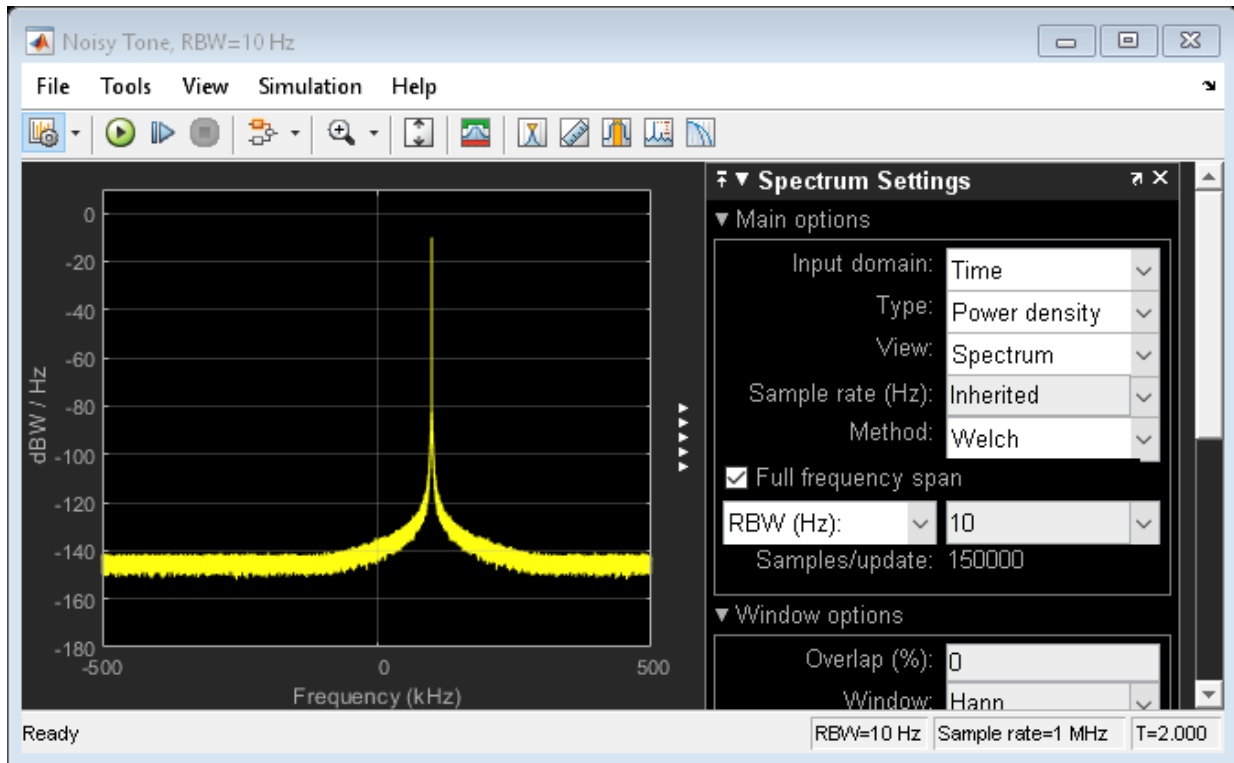
When the resolution bandwidth is 1 Hz, the dBW/Hz view for the spectrum analyzer shows the tone at 0 dBW/Hz. The Spectrum Analyzer block corrects for the power spreading effect of the Hann windowing.



The visual average of the phase noise achieves the spectrum defined by the Phase Noise block.



When the resolution bandwidth is 10 Hz, the dBW/Hz view for the spectrum analyzer shows the tone at -10 dBW/Hz. That same tone energy is now spread across 10 Hz instead of 1 Hz, so the sine wave PSD level reduces by 10 dB. With the resolution bandwidth at 10 Hz, the visual average of the phase noise still achieves the phase noise defined by the Phase Noise block.



The Spectrum Analyzer block still corrects for the power spreading effect of the Hann window, and it achieves better spectral averaging with the wider resolution bandwidth. For more information, see “Why Use Windows?” (Signal Processing Toolbox).

Further Exploration

In the Phase Noise block, change the **Phase noise level (dBc/Hz)** parameter, rerun the model, and notice how the spectrum shape changes. With more noise, the side lobes increase in amplitude. As more phase noise is added, the 100 Hz signal becomes less distinct and the measured RMS phase noise increases.

Add RF Impairments to DQPSK Signal

This model applies RF impairments to a signal modulated by differential quadrature phase shift keying (DQPSK). To demonstrate and visualize the RF impairments, levels

applied in this model are exaggerated and not representative of typical levels for modern radios.

A random signal is DQPSK modulated and various RF impairments are applied to the signal. The model uses impairment blocks from the RF Impairments library. After the impairment blocks, the signal forks into two paths. One path applies DC blocking, automatic gain control (AGC), and I/Q imbalance compensation to the signal before demodulation. Since the signal is DQPSK modulated, no carrier synchronization is required. The second path goes directly to demodulation. After demodulation, an error rate calculation is performed on both signals. To analyze the constellation, the model includes Constellation Diagram blocks after modulation, before correction, and after correction.

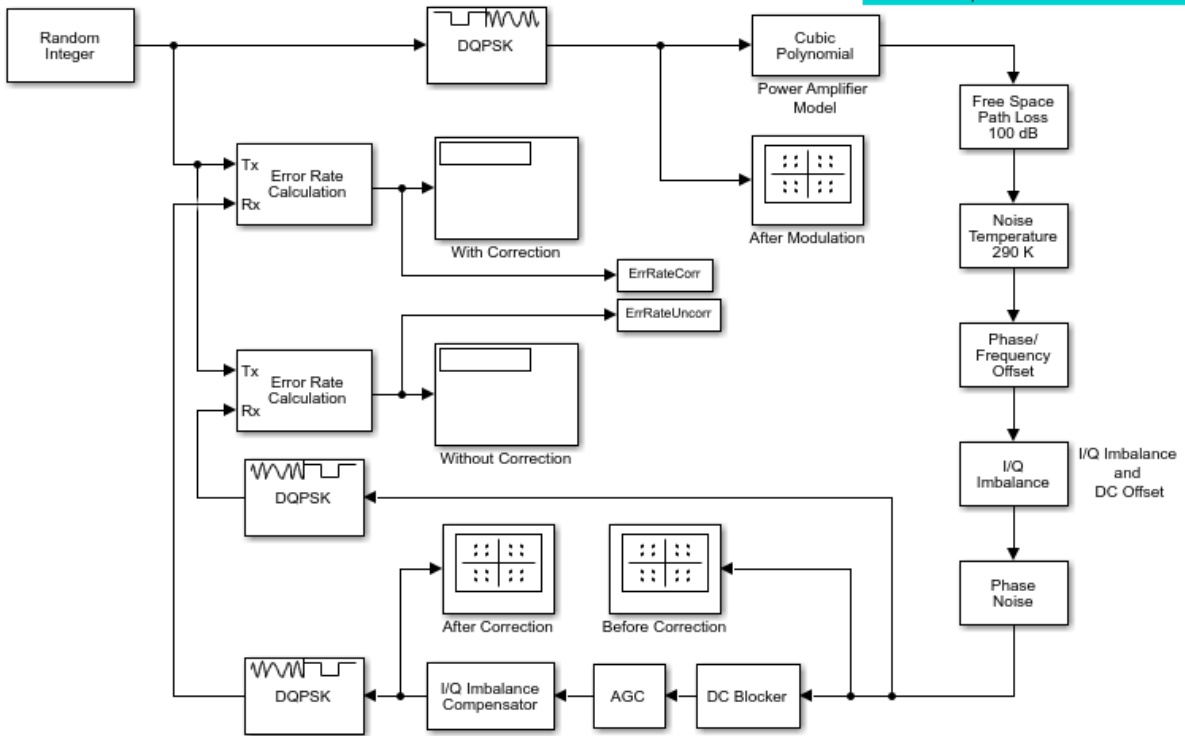
The error rate for the demodulated signal without AGC is primarily caused by free space path loss and I/Q imbalance. The QPSK modulation minimizes the effects of the other impairments.

Open Example Model and Explore Its Contents

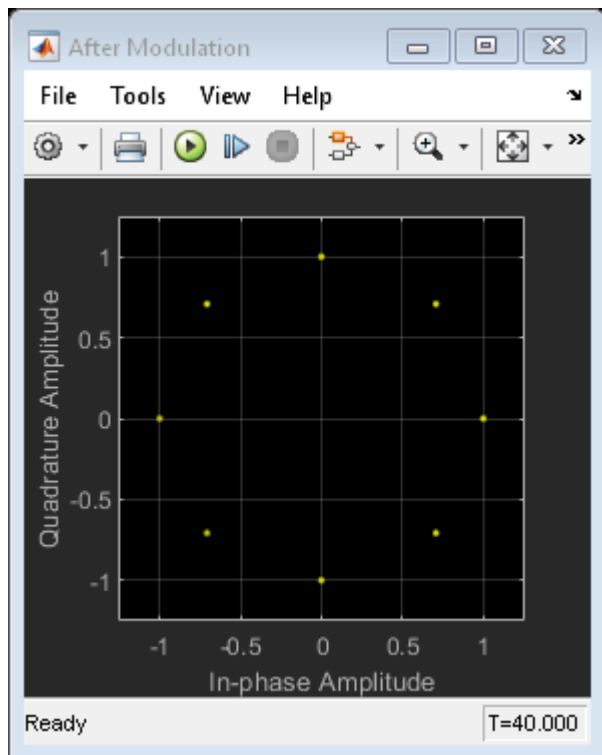
Open the example model `slex_rcvrimpairments_dqpsk`.

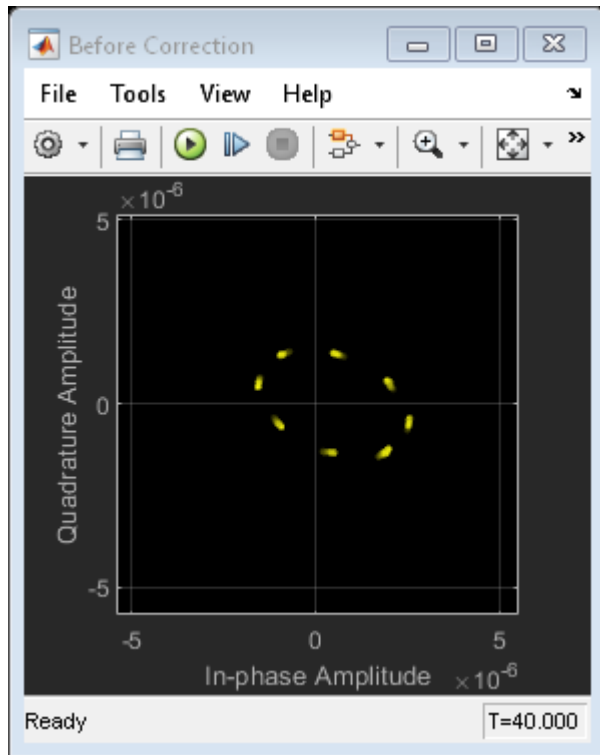
Using RF Impairments

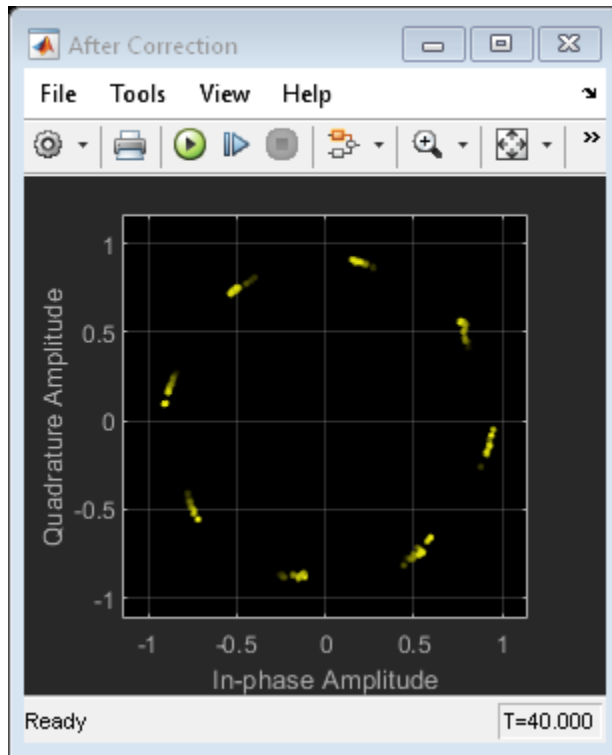
The variable to initialize free space loss is set in: **File>Model Properties>Callbacks>PreLoadFcn**



The After Modulation diagram, shows the clean reference DQPSK modulated constellation.







The transmitted signal is distorted by various RF impairments. The Before Correction diagram shows the attenuated and distorted constellation.

The signal on the correction path is adjusted by the DC Blocker, AGC Block, and I/Q Imbalance Compensator blocks. The After Correction diagram shows the constellation has been amplified and improved after the correction blocks.

Display the BER for the signal with and without correction.

```
Error rate for corrected signal: 0.000
Error rate for uncorrected signal: 0.042
```

Further Exploration

To run the model yourself, open the example using the button provided or by entering `open slx_rcvrmpairments_dqpsk` at the MATLAB® command prompt. Adjust the RF impairments, rerun the model, and notice the changes to the constellation diagrams

and error rates. Consider modifying the model to add an equalizer stage before the demodulation. Equalization has inherent ability to reduce some of the distortion caused by impairments. For more information, see “Equalization” on page 6-2.

Selected Bibliography for Channel Modeling

- [1] Simon, M. K., and Alouini, M. S., *Digital Communication over Fading Channels - A Unified Approach to Performance Analysis*, 1st Ed., Wiley, 2000.
- [2] 3rd Generation Partnership Project, Technical Specification Group Radio Access Network, Evolved Universal Terrestrial Radio Access (E-UTRA), Base Station (BS) radio transmission and reception, Release 10, 3GPP TS 36.104, v10.0.0, 2010-09.
- [3] 3rd Generation Partnership Project, Technical Specification Group Radio Access Network, Evolved Universal Terrestrial Radio Access (E-UTRA), User Equipment (UE) radio transmission and reception, Release 10, 3GPP TS 36.101, v10.0.0, 2010-10.

Measurements

- “Bit Error Rate (BER)” on page 15-2
- “Error Vector Magnitude (EVM)” on page 15-85
- “Modulation Error Ratio (MER)” on page 15-90
- “Adjacent Channel Power Ratio (ACPR)” on page 15-91
- “Complementary Cumulative Distribution Function CCDF” on page 15-99
- “Selected Bibliography for Measurements” on page 15-100

Bit Error Rate (BER)

In this section...
“Theoretical Results” on page 15-2
“Performance Results via Simulation” on page 15-21
“Performance Results via the Semianalytic Technique” on page 15-24
“Theoretical Performance Results” on page 15-27
“Error Rate Plots” on page 15-33
“BERTool” on page 15-37

Theoretical Results

Common Notation

The following notation is used throughout this Appendix:

Quantity or Operation	Notation
Size of modulation constellation	M
Number of bits per symbol	$k = \log_2 M$
Energy per bit-to-noise power-spectral-density ratio	$\frac{E_b}{N_0}$
Energy per symbol-to-noise power-spectral-density ratio	$\frac{E_s}{N_0} = k \frac{E_b}{N_0}$
Bit error rate (BER)	P_b
Symbol error rate (SER)	P_s
Real part	$\text{Re}[\cdot]$
Largest integer smaller than	$\lfloor \cdot \rfloor$

The following mathematical functions are used:

Function	Mathematical Expression
Q function	$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^{\infty} \exp(-t^2/2) dt$
Marcum Q function	$Q(a, b) = \int_b^{\infty} t \exp\left(-\frac{t^2 + a^2}{2}\right) I_0(at) dt$
Modified Bessel function of the first kind of order ν	$I_{\nu}(z) = \sum_{k=0}^{\infty} \frac{(z/2)^{\nu+2k}}{k! \Gamma(\nu+k+1)}$ <p>where</p> $\Gamma(x) = \int_0^{\infty} e^{-t} t^{x-1} dt$ <p>is the gamma function.</p>
Confluent hypergeometric function	${}_1F_1(a, c; x) = \sum_{k=0}^{\infty} \frac{(a)_k x^k}{(c)_k k!}$ <p>where the Pochhammer symbol, $(\lambda)_k$, is defined as $(\lambda)_0 = 1$, $(\lambda)_k = \lambda(\lambda+1)(\lambda+2)\dots(\lambda+k-1)$.</p>

The following acronyms are used:

Acronym	Definition
M-PSK	M -ary phase-shift keying
DE-M-PSK	Differentially encoded M -ary phase-shift keying
BPSK	Binary phase-shift keying
DE-BPSK	Differentially encoded binary phase-shift keying
QPSK	Quaternary phase-shift keying

Acronym	Definition
DE-QPSK	Differentially encoded quadrature phase-shift keying
OQPSK	Offset quadrature phase-shift keying
DE-OQPSK	Differentially encoded offset quadrature phase-shift keying
M-DPSK	<i>M</i> -ary differential phase-shift keying
M-PAM	<i>M</i> -ary pulse amplitude modulation
M-QAM	<i>M</i> -ary quadrature amplitude modulation
M-FSK	<i>M</i> -ary frequency-shift keying
MSK	Minimum shift keying
M-CPFSK	<i>M</i> -ary continuous-phase frequency-shift keying

Analytical Expressions Used in berawgn

- “M-PSK” on page 15-5
- “DE-M-PSK” on page 15-5
- “OQPSK” on page 15-6
- “DE-OQPSK” on page 15-6
- “M-DPSK” on page 15-6
- “M-PAM” on page 15-7
- “M-QAM” on page 15-7
- “Orthogonal M-FSK with Coherent Detection” on page 15-8
- “Nonorthogonal 2-FSK with Coherent Detection” on page 15-8
- “Orthogonal M-FSK with Noncoherent Detection” on page 15-9
- “Nonorthogonal 2-FSK with Noncoherent Detection” on page 15-10
- “Precoded MSK with Coherent Detection” on page 15-10
- “Differentially Encoded MSK with Coherent Detection” on page 15-10
- “MSK with Noncoherent Detection (Optimum Block-by-Block)” on page 15-10
- “CPFSK Coherent Detection (Optimum Block-by-Block)” on page 15-10

M-PSK

From equation 8.22 in [2]

$$P_s = \frac{1}{\pi} \int_0^{(M-1)\pi/M} \exp\left(-\frac{kE_b \sin^2[\pi/M]}{N_0 \sin^2\theta}\right) d\theta$$

The following expression is very close, but not strictly equal, to the exact BER (from [4] and equation 8.29 from [2]):

$$P_b = \frac{1}{k} \left(\sum_{i=1}^{M/2} (w_i) P_i \right)$$

where $w'_i = w_i + w_{M-i}$, $w'_{M/2} = w_{M/2}$, w_i is the Hamming weight of bits assigned to symbol i , and

$$P_i = \frac{1}{2\pi} \int_0^{\pi(1-(2i-1)/M)} \exp\left(-\frac{kE_b \sin^2[(2i-1)\pi/M]}{N_0 \sin^2\theta}\right) d\theta$$

$$- \frac{1}{2\pi} \int_0^{\pi(1-(2i+1)/M)} \exp\left(-\frac{kE_b \sin^2[(2i+1)\pi/M]}{N_0 \sin^2\theta}\right) d\theta$$

Special case of $M = 2$, e.g., BPSK (equation 5.2-57 from [1]):

$$P_s = P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right)$$

Special case of $M = 4$, e.g., QPSK (equations 5.2-59 and 5.2-62 from [1]):

$$P_s = 2Q\left(\sqrt{\frac{2E_b}{N_0}}\right) \left[1 - \frac{1}{2}Q\left(\sqrt{\frac{2E_b}{N_0}}\right) \right]$$

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right)$$

DE-M-PSK

$M = 2$, e.g., DE-BPSK (equation 8.36 from [2]):

$$P_s = P_b = 2Q\left(\sqrt{\frac{2E_b}{N_0}}\right) - 2Q^2\left(\sqrt{\frac{2E_b}{N_0}}\right)$$

$M = 4$, e.g., DE-QPSK (equation 8.38 from [2]):

$$P_s = 4Q\left(\sqrt{\frac{2E_b}{N_0}}\right) - 8Q^2\left(\sqrt{\frac{2E_b}{N_0}}\right) + 8Q^3\left(\sqrt{\frac{2E_b}{N_0}}\right) - 4Q^4\left(\sqrt{\frac{2E_b}{N_0}}\right)$$

From equation 5 in [3]:

$$P_b = 2Q\left(\sqrt{\frac{2E_b}{N_0}}\right)\left[1 - Q\left(\sqrt{\frac{2E_b}{N_0}}\right)\right]$$

OQPSK

Same BER/SER as QPSK [2].

DE-OQPSK

Same BER/SER as DE-QPSK [3].

M-DPSK

From equation 8.84 in [2]:

$$P_s = \frac{\sin(\pi/M)}{2\pi} \int_{-\pi/2}^{\pi/2} \frac{\exp(-(kE_b/N_0)(1 - \cos(\pi/M)\cos\theta))}{1 - \cos(\pi/M)\cos\theta} d\theta$$

The following expression is very close, but not strictly equal, to the exact BER [4]:

$$P_b = \frac{1}{k} \left(\sum_{i=1}^{M/2} (w_i) A_i \right)$$

where $w'_i = w_i + w_{M-i}$, $w'_{M/2} = w_{M/2}$, w_i is the Hamming weight of bits assigned to symbol i , and

$$A_i = F\left((2i + 1)\frac{\pi}{M}\right) - F\left((2i - 1)\frac{\pi}{M}\right)$$

$$F(\psi) = -\frac{\sin\psi}{4\pi} \int_{-\pi/2}^{\pi/2} \frac{\exp(-kE_b/N_0(1 - \cos\psi\cos t))}{1 - \cos\psi\cos t} dt$$

Special case of $M = 2$ (equation 8.85 from [2]):

$$P_b = \frac{1}{2} \exp\left(-\frac{E_b}{N_0}\right)$$

M-PAM

From equations 8.3 and 8.7 in [2], and equation 5.2-46 in [1]:

$$P_s = 2\left(\frac{M-1}{M}\right)Q\left(\sqrt{\frac{6}{M^2-1} \frac{kE_b}{N_0}}\right)$$

From [5]:

$$P_b = \frac{2}{M \log_2 M} \times \sum_{k=1}^{\log_2 M} (1-2^{-k})^{M-1} \sum_{i=0}^{M-1} \left\{ (-1)^{\lfloor \frac{i2^k-1}{M} \rfloor} \left(2^{k-1} - \left\lfloor \frac{i2^k-1}{M} + \frac{1}{2} \right\rfloor \right) \right\} Q\left((2i+1) \sqrt{\frac{6 \log_2 M E_b}{M^2-1 N_0}} \right)$$

M-QAM

For square M-QAM, $k = \log_2 M$ is even (equation 8.10 from [2], and equations 5.2-78 and 5.2-79 from [1]):

$$P_s = 4 \frac{\sqrt{M}-1}{\sqrt{M}} Q\left(\sqrt{\frac{3}{M-1} \frac{kE_b}{N_0}}\right) - 4 \left(\frac{\sqrt{M}-1}{\sqrt{M}}\right)^2 Q^2\left(\sqrt{\frac{3}{M-1} \frac{kE_b}{N_0}}\right)$$

From [5]:

$$P_b = \frac{2}{\sqrt{M} \log_2 \sqrt{M}} \times \sum_{k=1}^{\log_2 \sqrt{M}} (1-2^{-k})^{\sqrt{M}-1} \sum_{i=0}^{\sqrt{M}-1} \left\{ (-1)^{\lfloor \frac{i2^k-1}{\sqrt{M}} \rfloor} \left(2^{k-1} - \left\lfloor \frac{i2^k-1}{\sqrt{M}} + \frac{1}{2} \right\rfloor \right) \right\} Q\left((2i+1) \sqrt{\frac{6 \log_2 M E_b}{2(M-1) N_0}} \right)$$

For rectangular (non-square) M-QAM, $k = \log_2 M$ is odd, $M = I \times J$, $I = 2^{\frac{k-1}{2}}$, and

$$J = 2^{\frac{k+1}{2}}:$$

$$P_s = \frac{4IJ - 2I - 2J}{M} \times Q\left(\sqrt{\frac{6\log_2(IJ) E_b}{(I^2 + J^2 - 2)N_0}}\right) - \frac{4}{M}(1 + IJ - I - J)Q^2\left(\sqrt{\frac{6\log_2(IJ) E_b}{(I^2 + J^2 - 2)N_0}}\right)$$

From [5]:

$$P_b = \frac{1}{\log_2(IJ)} \left(\sum_{k=1}^{\log_2 I} P_I(k) + \sum_{l=1}^{\log_2 J} P_J(l) \right)$$

where

$$P_I(k) = \frac{2}{I} \sum_{i=0}^{(1-2^{-k})I-1} \left\{ (-1)^{\lfloor \frac{i2^{k-1}}{I} \rfloor} \left(2^{k-1} - \left\lfloor \frac{i2^{k-1}}{I} + \frac{1}{2} \right\rfloor \right) Q\left((2i+1) \sqrt{\frac{6\log_2(IJ) E_b}{I^2 + J^2 - 2} N_0} \right) \right\}$$

and

$$P_J(k) = \frac{2}{J} \sum_{j=0}^{(1-2^{-k})J-1} \left\{ (-1)^{\lfloor \frac{j2^{k-1}}{J} \rfloor} \left(2^{k-1} - \left\lfloor \frac{j2^{k-1}}{J} + \frac{1}{2} \right\rfloor \right) Q\left((2j+1) \sqrt{\frac{6\log_2(IJ) E_b}{I^2 + J^2 - 2} N_0} \right) \right\}$$

Orthogonal M-FSK with Coherent Detection

From equation 8.40 in [2] and equation 5.2-21 in [1]:

$$P_s = 1 - \int_{-\infty}^{\infty} \left[Q\left(-q - \sqrt{\frac{2kE_b}{N_0}} \right) \right]^{M-1} \frac{1}{\sqrt{2\pi}} \exp\left(-\frac{q^2}{2} \right) dq$$

$$P_b = \frac{2^k - 1}{2^k - 1} P_s$$

Nonorthogonal 2-FSK with Coherent Detection

For $M = 2$ (from equation 5.2-21 in [1] and equation 8.44 in [2]):

$$P_s = P_b = Q\left(\sqrt{\frac{E_b(1 - \text{Re}[\rho])}{N_0}} \right)$$

ρ is the complex correlation coefficient:

$$\rho = \frac{1}{2E_b} \int_0^{T_b} \tilde{s}_1(t) \tilde{s}_2^*(t) dt$$

where $\tilde{s}_1(t)$ and $\tilde{s}_2(t)$ are complex lowpass signals, and

$$E_b = \frac{1}{2} \int_0^{T_b} |\tilde{s}_1(t)|^2 dt = \frac{1}{2} \int_0^{T_b} |\tilde{s}_2(t)|^2 dt$$

For example:

$$\begin{aligned} \tilde{s}_1(t) &= \sqrt{\frac{2E_b}{T_b}} e^{j2\pi f_1 t}, \quad \tilde{s}_2(t) = \sqrt{\frac{2E_b}{T_b}} e^{j2\pi f_2 t} \\ \rho &= \frac{1}{2E_b} \int_0^{T_b} \sqrt{\frac{2E_b}{T_b}} e^{j2\pi f_1 t} \sqrt{\frac{2E_b}{T_b}} e^{-j2\pi f_2 t} dt = \frac{1}{T_b} \int_0^{T_b} e^{j2\pi(f_1 - f_2)t} dt \\ &= \frac{\sin(\pi \Delta f T_b)}{\pi \Delta f T_b} e^{j\pi \Delta f t} \end{aligned}$$

where $\Delta f = f_1 - f_2$.

$$\begin{aligned} \text{Re}[\rho] &= \text{Re} \left[\frac{\sin(\pi \Delta f T_b)}{\pi \Delta f T_b} e^{j\pi \Delta f t} \right] = \frac{\sin(\pi \Delta f T_b)}{\pi \Delta f T_b} \cos(\pi \Delta f T_b) = \frac{\sin(2\pi \Delta f T_b)}{2\pi \Delta f T_b} \\ \Rightarrow P_b &= Q \left(\sqrt{\frac{E_b (1 - \sin(2\pi \Delta f T_b) / (2\pi \Delta f T_b))}{N_0}} \right) \end{aligned}$$

(from equation 8.44 in [2], where $h = \Delta f T_b$)

Orthogonal M-FSK with Noncoherent Detection

From equation 5.4-46 in [1] and equation 8.66 in [2]:

$$\begin{aligned} P_s &= \sum_{m=1}^{M-1} (-1)^{m+1} \binom{M-1}{m} \frac{1}{m+1} \exp \left[-\frac{m}{m+1} \frac{kE_b}{N_0} \right] \\ P_b &= \frac{1}{2} \frac{M}{M-1} P_s \end{aligned}$$

Nonorthogonal 2-FSK with Noncoherent Detection

For $M = 2$ (from equation 5.4-53 in [1] and equation 8.69 in [2]):

$$P_s = P_b = Q(\sqrt{a}, \sqrt{b}) - \frac{1}{2} \exp\left(-\frac{a+b}{2}\right) I_0(\sqrt{ab})$$

where

$$a = \frac{E_b}{2N_0}(1 - \sqrt{1 - |\rho|^2}), \quad b = \frac{E_b}{2N_0}(1 + \sqrt{1 - |\rho|^2})$$

Precoded MSK with Coherent Detection

Same BER/SER as BPSK.

Differentially Encoded MSK with Coherent Detection

Same BER/SER as DE-BPSK.

MSK with Noncoherent Detection (Optimum Block-by-Block)

Upper bound (from equations 10.166 and 10.164 in [6]):

$$\begin{aligned} P_s &= P_b \\ &\leq \frac{1}{2}[1 - Q(\sqrt{b_1}, \sqrt{a_1}) + Q(\sqrt{a_1}, \sqrt{b_1})] + \frac{1}{4}[1 - Q(\sqrt{b_4}, \sqrt{a_4}) + Q(\sqrt{a_4}, \sqrt{b_4})] + \frac{1}{2}e^{-\frac{E_b}{N_0}} \end{aligned}$$

where

$$\begin{aligned} a_1 &= \frac{E_b}{N_0} \left(1 - \sqrt{\frac{3 - 4/\pi^2}{4}}\right), \quad b_1 = \frac{E_b}{N_0} \left(1 + \sqrt{\frac{3 - 4/\pi^2}{4}}\right) \\ a_4 &= \frac{E_b}{N_0} \left(1 - \sqrt{1 - 4/\pi^2}\right), \quad b_4 = \frac{E_b}{N_0} \left(1 + \sqrt{1 - 4/\pi^2}\right) \end{aligned}$$

CPFSK Coherent Detection (Optimum Block-by-Block)

Lower bound (from equation 5.3-17 in [1]):

$$P_s > K\delta_{\min} Q\left(\sqrt{\frac{E_b}{N_0}\delta_{\min}^2}\right)$$

Upper bound:

$$\delta_{\min}^2 > \min_{1 \leq i \leq M-1} \{2i(1 - \text{sinc}(2ih))\}$$

where h is the modulation index, and $K_{\delta_{\min}}$ is the number of paths having the minimum distance.

$$P_b \cong \frac{P_s}{k}$$

Analytical Expressions Used in berfading

- “Notation” on page 15-11
- “M-PSK with MRC” on page 15-12
- “DE-M-PSK with MRC” on page 15-13
- “M-PAM with MRC” on page 15-13
- “M-QAM with MRC” on page 15-14
- “M-DPSK with Postdetection EGC” on page 15-15
- “Orthogonal 2-FSK, Coherent Detection with MRC” on page 15-16
- “Nonorthogonal 2-FSK, Coherent Detection with MRC” on page 15-16
- “Orthogonal M-FSK, Noncoherent Detection with EGC” on page 15-16
- “Nonorthogonal 2-FSK, Noncoherent Detection with No Diversity” on page 15-17

Notation

The following notation is used for the expressions found in berfading.

Value	Notation
Power of the fading amplitude r	$\Omega = E[r^2]$, where $E[\cdot]$ denotes statistical expectation
Number of diversity branches	L

Value	Notation
SNR per symbol per branch	$\bar{\gamma}_l = \left(\Omega_l \frac{E_s}{N_0} \right) / L = \left(\Omega_l \frac{kE_b}{N_0} \right) / L$ <p>For identically-distributed diversity branches:</p> $\bar{\gamma} = \left(\Omega \frac{kE_b}{N_0} \right) / L$
Moment generating functions for each diversity branch	<p>Rayleigh fading:</p> $M_{\gamma_l}(s) = \frac{1}{1 - s\bar{\gamma}_l}$ <p>Rician fading:</p> $M_{\gamma_l}(s) = \frac{1 + K}{1 + K - s\bar{\gamma}_l} e^{\left[\frac{Ks\bar{\gamma}_l}{(1 + K) - s\bar{\gamma}_l} \right]}$ <p>where K is the ratio of energy in the specular component to the energy in the diffuse component (linear scale).</p> <p>For identically-distributed diversity branches: $M_{\gamma_l}(s) = M_{\gamma}(s)$ for all l.</p>

The following acronyms are used:

Acronym	Definition
MRC	maximal-ratio combining
EGC	equal-gain combining

M-PSK with MRC

From equation 9.15 in [2]:

$$P_s = \frac{1}{\pi} \int_0^{(M-1)\pi/M} \prod_{l=1}^L M_{\gamma_l} \left(-\frac{\sin^2(\pi/M)}{\sin^2\theta} \right) d\theta$$

From [4] and [2]:

$$P_b = \frac{1}{k} \left(\sum_{i=1}^{M/2} (w_i) \bar{P}_i \right)$$

where $w'_i = w_i + w_{M-i}$, $w'_{M/2} = w_{M/2}$, w_i is the Hamming weight of bits assigned to symbol i , and

$$\begin{aligned} \bar{P}_i = & \frac{1}{2\pi} \int_0^{\pi(1-(2i-1)/M)} \prod_{l=1}^L M_{\nu_l} \left(-\frac{1}{\sin^2 \theta} \sin^2 \frac{(2i-1)\pi}{M} \right) d\theta \\ & - \frac{1}{2\pi} \int_0^{\pi(1-(2i+1)/M)} \prod_{l=1}^L M_{\nu_l} \left(-\frac{1}{\sin^2 \theta} \sin^2 \frac{(2i+1)\pi}{M} \right) d\theta \end{aligned}$$

For the special case of Rayleigh fading with $M = 2$ (from equations C-18, C-21, and Table C-1 in [6]):

$$P_b = \frac{1}{2} \left[1 - \mu \sum_{i=0}^{L-1} \binom{2i}{i} \left(\frac{1-\mu^2}{4} \right)^i \right]$$

where

$$\mu = \sqrt{\frac{\bar{\gamma}}{\bar{\gamma} + 1}}$$

If $L = 1$:

$$P_b = \frac{1}{2} \left[1 - \sqrt{\frac{\bar{\gamma}}{\bar{\gamma} + 1}} \right]$$

DE-M-PSK with MRC

For $M = 2$ (from equations 8.37 and 9.8-9.11 in [2]):

$$P_s = P_b = \frac{2}{\pi} \int_0^{\pi/2} \prod_{l=1}^L M_{\nu_l} \left(-\frac{1}{\sin^2 \theta} \right) d\theta - \frac{2}{\pi} \int_0^{\pi/4} \prod_{l=1}^L M_{\nu_l} \left(-\frac{1}{\sin^2 \theta} \right) d\theta$$

M-PAM with MRC

From equation 9.19 in [2]:

$$P_s = \frac{2(M-1)}{M\pi} \int_0^{\pi/2} \prod_{l=1}^L M_{\gamma_l} \left(-\frac{3/(M^2-1)}{\sin^2\theta} \right) d\theta$$

From [5] and [2]:

$$P_b = \frac{2}{\pi M \log_2 M} \times \sum_{k=1}^{\log_2 M} (1-2^{-k})^{M-1} \sum_{i=0}^{M-1} \left\{ (-1)^{\lfloor \frac{i2^{k-1}}{M} \rfloor} \left(2^{k-1} - \left\lfloor \frac{i2^{k-1}}{M} + \frac{1}{2} \right\rfloor \right) \int_0^{\pi/2} \prod_{l=1}^L M_{\gamma_l} \left(-\frac{(2i+1)^2 3/(M^2-1)}{\sin^2\theta} \right) d\theta \right\}$$

M-QAM with MRC

For square M-QAM, $k = \log_2 M$ is even (equation 9.21 in [2]):

$$P_s = \frac{4}{\pi} \left(1 - \frac{1}{\sqrt{M}} \right) \int_0^{\pi/2} \prod_{l=1}^L M_{\gamma_l} \left(-\frac{3/(2(M-1))}{\sin^2\theta} \right) d\theta - \frac{4}{\pi} \left(1 - \frac{1}{\sqrt{M}} \right)^2 \int_0^{\pi/4} \prod_{l=1}^L M_{\gamma_l} \left(-\frac{3/(2(M-1))}{\sin^2\theta} \right) d\theta$$

From [5] and [2]:

$$P_b = \frac{2}{\pi \sqrt{M} \log_2 \sqrt{M}} \times \sum_{k=1}^{\log_2 \sqrt{M}} (1-2^{-k})^{\sqrt{M}-1} \sum_{i=0}^{\sqrt{M}-1} \left\{ (-1)^{\lfloor \frac{i2^{k-1}}{\sqrt{M}} \rfloor} \left(2^{k-1} - \left\lfloor \frac{i2^{k-1}}{\sqrt{M}} + \frac{1}{2} \right\rfloor \right) \int_0^{\pi/2} \prod_{l=1}^L M_{\gamma_l} \left(-\frac{(2i+1)^2 3/(2(M-1))}{\sin^2\theta} \right) d\theta \right\}$$

For rectangular (nonsquare) M-QAM, $k = \log_2 M$ is odd, $M = I \times J$, $I = 2^{\frac{k-1}{2}}$, $J = 2^{\frac{k+1}{2}}$,

$$\bar{\gamma}_l = \Omega_l \log_2(IJ) \frac{E_b}{N_0}, \text{ and}$$

$$P_s = \frac{4IJ - 2I - 2J}{M\pi} \int_0^{\pi/2} \prod_{l=1}^L M_{\nu_l} \left(-\frac{3/(I^2 + J^2 - 2)}{\sin^2\theta} \right) d\theta$$

$$- \frac{4}{M\pi} (1 + IJ - I - J) \int_0^{\pi/4} \prod_{l=1}^L M_{\nu_l} \left(-\frac{3/(I^2 + J^2 - 2)}{\sin^2\theta} \right) d\theta$$

From [5] and [2]:

$$P_b = \frac{1}{\log_2(IJ)} \left(\sum_{k=1}^{\log_2 I} P_I(k) + \sum_{l=1}^{\log_2 J} P_J(l) \right)$$

$$P_I(k) = \frac{2}{I\pi} \sum_{i=0}^{(1-2^{-k})I-1} \left\{ (-1)^{\lfloor \frac{i2^k-1}{I} \rfloor} \left(2^{k-1} - \left\lfloor \frac{i2^k-1}{I} + \frac{1}{2} \right\rfloor \right) \int_0^{\pi/2} \prod_{l=1}^L M_{\nu_l} \left(-\frac{(2i+1)^2 3/(I^2 + J^2 - 2)}{\sin^2\theta} \right) d\theta \right.$$

$$P_J(k) = \frac{2}{J\pi} \sum_{j=0}^{(1-2^{-k})J-1} \left\{ (-1)^{\lfloor \frac{j2^k-1}{J} \rfloor} \left(2^{k-1} - \left\lfloor \frac{j2^k-1}{J} + \frac{1}{2} \right\rfloor \right) \int_0^{\pi/2} \prod_{l=1}^L M_{\nu_l} \left(-\frac{(2j+1)^2 3/(I^2 + J^2 - 2)}{\sin^2\theta} \right) d\theta \right.$$

M-DPSK with Postdetection EGC

From equation 8.165 in [2]:

$$P_s = \frac{\sin(\pi/M)}{2\pi} \int_{-\pi/2}^{\pi/2} \frac{1}{[1 - \cos(\pi/M)\cos\theta]} \prod_{l=1}^L M_{\nu_l}(-[1 - \cos(\pi/M)\cos\theta]) d\theta$$

From [4] and [2]:

$$P_b = \frac{1}{k} \left(\sum_{i=1}^{M/2} (w_i) \bar{A}_i \right)$$

where $w'_i = w_i + w_{M-i}$, $w'_{M/2} = w_{M/2}$, w_i is the Hamming weight of bits assigned to symbol i , and

$$\bar{A}_i = \bar{F}\left((2i+1)\frac{\pi}{M}\right) - \bar{F}\left((2i-1)\frac{\pi}{M}\right)$$

$$\bar{F}(\psi) = -\frac{\sin\psi}{4\pi} \int_{-\pi/2}^{\pi/2} \frac{1}{(1 - \cos\psi\cos t)} \prod_{l=1}^L M_{\nu_l}(-(1 - \cos\psi\cos t)) dt$$

For the special case of Rayleigh fading with $M = 2$, and $L = 1$ (equation 8.173 from [2]):

$$P_b = \frac{1}{2(1 + \bar{\gamma})}$$

Orthogonal 2-FSK, Coherent Detection with MRC

From equation 9.11 in [2]:

$$P_s = P_b = \frac{1}{\pi} \int_0^{\pi/2} \prod_{l=1}^L M_{\nu_l} \left(-\frac{1/2}{\sin^2 \theta} \right) d\theta$$

For the special case of Rayleigh fading (equations 14.4-15 and 14.4-21 in [1]):

$$P_s = P_b = \frac{1}{2^L} \left(1 - \sqrt{\frac{\bar{\gamma}}{2 + \bar{\gamma}}} \right)^{L-1} \sum_{k=0}^{L-1} \binom{L-1+k}{k} \frac{1}{2^k} \left(1 + \sqrt{\frac{\bar{\gamma}}{2 + \bar{\gamma}}} \right)^k$$

Nonorthogonal 2-FSK, Coherent Detection with MRC

Equations 9.11 and 8.44 in [2]:

$$P_s = P_b = \frac{1}{\pi} \int_0^{\pi/2} \prod_{l=1}^L M_{\nu_l} \left(-\frac{(1 - \text{Re}[\rho])/2}{\sin^2 \theta} \right) d\theta$$

For the special case of Rayleigh fading with $L = 1$ (equation 20 in [8] and equation 8.130 in [2]):

$$P_s = P_b = \frac{1}{2} \left[1 - \sqrt{\frac{\bar{\gamma}(1 - \text{Re}[\rho])}{2 + \bar{\gamma}(1 - \text{Re}[\rho])}} \right]$$

Orthogonal M-FSK, Noncoherent Detection with EGC

Rayleigh fading (equation 14.4-47 in [1]):

$$P_s = 1 - \int_0^{\infty} \frac{1}{(1 + \bar{\gamma})^L (L-1)!} U^{L-1} e^{-\frac{U}{1 + \bar{\gamma}}} \left(1 - e^{-U} \sum_{k=0}^{L-1} \frac{U^k}{k!} \right)^{M-1} dU$$

$$P_b = \frac{1}{2} \frac{M}{M-1} P_s$$

Rician fading (equation 41 in [8]):

$$P_s = \sum_{r=1}^{M-1} \frac{(-1)^{r+1} e^{-LK\bar{v}_r/(1+\bar{v}_r)} (M-1)^{r(L-1)} \sum_{n=0}^{L-1} \beta_{nr} \frac{\Gamma(L+n)}{\Gamma(L)} \left[\frac{1+\bar{v}_r}{r+1+r\bar{v}_r} \right]^n {}_1F_1\left(L+n, L; \frac{LK\bar{v}_r/(1+\bar{v}_r)}{r(1+\bar{v}_r)+1}\right)}{(r(1+\bar{v}_r)+1)^L} \beta_{nr} \frac{\Gamma(L+n)}{\Gamma(L)} \left[\frac{1+\bar{v}_r}{r+1+r\bar{v}_r} \right]^n {}_1F_1\left(L+n, L; \frac{LK\bar{v}_r/(1+\bar{v}_r)}{r(1+\bar{v}_r)+1}\right)}$$

$$P_b = \frac{1}{2} \frac{M}{M-1} P_s$$

where

$$\bar{v}_r = \frac{1}{1+K} \bar{v}$$

$$\beta_{nr} = \sum_{i=n}^L \sum_{j=0}^{L-1} \frac{\beta_{i(r-1)}}{(n-i)!} I_{[0, (r-1)(L-1)]}(i)$$

$$\beta_{00} = \beta_{0r} = 1$$

$$\beta_{n1} = 1/n!$$

$$\beta_{1r} = r$$

and $I_{[a,b]}(i) = 1$ if $a \leq i \leq b$ and 0 otherwise.

Nonorthogonal 2-FSK, Noncoherent Detection with No Diversity

From equation 8.163 in [2]:

$$P_s = P_b = \frac{1}{4\pi} \int_{-\pi}^{\pi} \frac{1-\zeta^2}{1+2\zeta\sin\theta+\zeta^2} M_V\left(-\frac{1}{4}(1+\sqrt{1-\rho^2})(1+2\zeta\sin\theta+\zeta^2)\right) d\theta$$

where

$$\zeta = \sqrt{\frac{1-\sqrt{1-\rho^2}}{1+\sqrt{1-\rho^2}}}$$

Analytical Expressions Used in bercoding and BERTool

- “Common Notation for This Section” on page 15-18
- “Block Coding” on page 15-18
- “Convolutional Coding” on page 15-20

Common Notation for This Section

Description	Notation
Energy-per-information bit-to-noise power-spectral-density ratio	$\gamma_b = \frac{E_b}{N_0}$
Message length	K
Code length	N
Code rate	$R_c = \frac{K}{N}$

Block Coding

Specific notation for block coding expressions: d_{\min} is the minimum distance of the code.

Soft Decision

BPSK, QPSK, OQPSK, PAM-2, QAM-4, and precoded MSK (equation 8.1-52 in [1]):

$$P_b \leq \frac{1}{2}(2^K - 1)Q(\sqrt{2\gamma_b R_c d_{\min}})$$

DE-BPSK, DE-QPSK, DE-OQPSK, and DE-MSK:

$$P_b \leq \frac{1}{2}(2^K - 1)[2Q(\sqrt{2\gamma_b R_c d_{\min}})[1 - Q(\sqrt{2\gamma_b R_c d_{\min}})]]$$

BFSK, coherent detection (equations 8.1-50 and 8.1-58 in [1]):

$$P_b \leq \frac{1}{2}(2^K - 1)Q(\sqrt{\gamma_b R_c d_{\min}})$$

BFSK, noncoherent square-law detection (equations 8.1-65 and 8.1-64 in [1]):

$$P_b \leq \frac{1}{2} \frac{2^K - 1}{2^{2d_{\min} - 1}} \exp\left(-\frac{1}{2}\gamma_b R_c d_{\min}\right) \sum_{i=0}^{d_{\min} - 1} \left(\frac{1}{2}\gamma_b R_c d_{\min}\right)^i \frac{1}{i!} \sum_{r=0}^{d_{\min} - 1 - i} \binom{2d_{\min} - 1}{r}$$

DPSK:

$$P_b \leq \frac{1}{2} \frac{2^K - 1}{2^{2d_{\min} - 1}} \exp(-\gamma_b R_c d_{\min}) \sum_{i=0}^{d_{\min} - 1} (\gamma_b R_c d_{\min})^i \frac{1}{i!} \sum_{r=0}^{d_{\min} - 1 - i} \binom{2d_{\min} - 1}{r}$$

Hard Decision

General linear block code (equations 4.3, 4.4 in [9], and 12.136 in [6]):

$$P_b \leq \frac{1}{N} \sum_{m=t+1}^N (m+t) \binom{N}{m} p^m (1-p)^{N-m}$$

$$t = \left\lfloor \frac{1}{2}(d_{\min} - 1) \right\rfloor$$

Hamming code (equations 4.11, 4.12 in [9], and 6.72, 6.73 in [7]):

$$P_b \approx \frac{1}{N} \sum_{m=2}^N m \binom{N}{m} p^m (1-p)^{N-m} = p - p(1-p)^{N-1}$$

(24, 12) extended Golay code (equation 4.17 in [9], and 12.139 in [6]):

$$P_b \leq \frac{1}{24} \sum_{m=4}^{24} \beta_m \binom{24}{m} p^m (1-p)^{24-m}$$

where β_m is the average number of channel symbol errors that remain in corrected N -tuple when the channel caused m symbol errors (table 4.2 in [9]).

Reed-Solomon code with $N = Q - 1 = 2^q - 1$:

$$P_b \approx \frac{2^q - 1}{2^q - 1} \frac{1}{N} \sum_{m=t+1}^N m \binom{N}{m} (P_s)^m (1 - P_s)^{N-m}$$

for FSK (equations 4.25, 4.27 in [9], 8.1-115, 8.1-116 in [1], 8.7, 8.8 in [7], and 12.142, 12.143 in [6]), and

$$P_b \approx \frac{1}{q} \frac{1}{N} \sum_{m=t+1}^N m \binom{N}{m} (P_s)^m (1 - P_s)^{N-m}$$

otherwise.

If $\log_2 Q / \log_2 M = q/k = h$ where h is an integer (equation 1 in [10]):

$$P_s = 1 - (1 - s)^h$$

where s is the symbol error rate (SER) in an uncoded AWGN channel.

For example, for BPSK, $M = 2$ and $P_s = 1 - (1 - s)^q$

Otherwise, P_s is given by table 1 and equation 2 in [10].

Convolutional Coding

Specific notation for convolutional coding expressions: d_{free} is the free distance of the code, and a_d is the number of paths of distance d from the all-zero path that merge with the all-zero path for the first time.

Soft Decision

From equations 8.2-26, 8.2-24, and 8.2-25 in [1], and equations 13.28 and 13.27 in [6]:

$$P_b < \sum_{d=d_{free}}^{\infty} a_d f(d) P_2(d)$$

with transfer function

$$T(D, N) = \sum_{d=d_{free}}^{\infty} a_d D^d N^{f(d)}$$

$$\left. \frac{dT(D, N)}{dN} \right|_{N=1} = \sum_{d=d_{free}}^{\infty} a_d f(d) D^d$$

where $f(d)$ is the exponent of N as a function of d .

Results for BPSK, QPSK, OQPSK, PAM-2, QAM-4, precoded MSK, DE-BPSK, DE-QPSK, DE-OQPSK, DE-MSK, DPSK, and BFSK are obtained as:

$$P_2(d) = P_b \left| \frac{E_b}{N_0} \right| = \gamma_b R_c d$$

where P_b is the BER in the corresponding uncoded AWGN channel. For example, for BPSK (equation 8.2-20 in [1]):

$$P_2(d) = Q(\sqrt{2\gamma_b R_c d})$$

Hard Decision

From equations 8.2-33, 8.2-28, and 8.2-29 in [1], and equations 13.28, 13.24, and 13.25 in [6]:

$$P_b < \sum_{d=d_{free}}^{\infty} a_d f(d) P_2(d)$$

where

$$P_2(d) = \sum_{k=(d+1)/2}^d \binom{d}{k} p^k (1-p)^{d-k}$$

when d is odd, and

$$P_2(d) = \sum_{k=d/2+1}^d \binom{d}{k} p^k (1-p)^{d-k} + \frac{1}{2} \binom{d}{d/2} p^{d/2} (1-p)^{d/2}$$

when d is even (p is the bit error rate (BER) in an uncoded AWGN channel).

Performance Results via Simulation

- “Section Overview” on page 15-21
- “Using Simulated Data to Compute Bit and Symbol Error Rates” on page 15-21
- “Example: Computing Error Rates” on page 15-22
- “Comparing Symbol Error Rate and Bit Error Rate” on page 15-23

Section Overview

One way to compute the bit error rate or symbol error rate for a communication system is to simulate the transmission of data messages and compare all messages before and after transmission. The simulation of the communication system components using Communications Toolbox is covered in other parts of this guide. This section describes how to compare the data messages that enter and leave the simulation.

Another example of computing performance results via simulation is in “Curve Fitting for Error Rate Plots” on page 15-34 in the discussion of curve fitting.

Using Simulated Data to Compute Bit and Symbol Error Rates

The `biterr` function compares two sets of data and computes the number of bit errors and the bit error rate. The `symerr` function compares two sets of data and computes the

number of symbol errors and the symbol error rate. An error is a discrepancy between corresponding points in the two sets of data.

Of the two sets of data, typically one represents messages entering a transmitter and the other represents recovered messages leaving a receiver. You might also compare data entering and leaving other parts of your communication system, for example, data entering an encoder and data leaving a decoder.

If your communication system uses several bits to represent one symbol, counting bit errors is different from counting symbol errors. In either the bit- or symbol-counting case, the error rate is the number of errors divided by the total number (of bits or symbols) transmitted.

Note To ensure an accurate error rate, you should typically simulate enough data to produce at least 100 errors.

If the error rate is very small (for example, 10^{-6} or smaller), the semianalytic technique might compute the result more quickly than a simulation-only approach. See “Performance Results via the Semianalytic Technique” on page 15-24 for more information on how to use this technique.

Example: Computing Error Rates

The script below uses the `symerr` function to compute the symbol error rates for a noisy linear block code. After artificially adding noise to the encoded message, it compares the resulting noisy code to the original code. Then it decodes and compares the decoded message to the original one.

```
m = 3; n = 2^m-1; k = n-m; % Prepare to use Hamming code.
msg = randi([0 1],k*200,1); % 200 messages of k bits each
code = encode(msg,n,k,'hamming');
codenoisy = rem(code+(rand(n*200,1)>.95),2); % Add noise.
% Decode and correct some errors.
newmsg = decode(codenoisy,n,k,'hamming');
% Compute and display symbol error rates.
noisyVec = step(comm.ErrorRate,code,codenoisy);
decodedVec = step(comm.ErrorRate,msg,newmsg);
disp(['Error rate in the received code: ',num2str(noisyVec(1))])
disp(['Error rate after decoding: ',num2str(decodedVec(1))])
```

The output is below. The error rate decreases after decoding because the Hamming decoder corrects some of the errors. Your results might vary because this example uses random numbers.

Error rate in the received code: 0.054286
 Error rate after decoding: 0.03

Comparing Symbol Error Rate and Bit Error Rate

In the example above, the symbol errors and bit errors are the same because each symbol is a bit. The commands below illustrate the difference between symbol errors and bit errors in other situations.

```
a = [1 2 3]'; b = [1 4 4]';
format rat % Display fractions instead of decimals.
% Create ErrorRate Calculator System object
serVec = step(comm.ErrorRate,a,b);
srate = serVec(1)
snum = serVec(2)
% Convert integers to bits
hIntToBit = comm.IntegerToBit(3);
a_bit = step(hIntToBit, a);
b_bit = step(hIntToBit, b);
% Calculate BER
berVec = step(comm.ErrorRate,a_bit,b_bit);
brate = berVec(1)
bnum = berVec(2)
```

The output is below.

```
snum =
      2

srate =
      2/3

bnum =
      5
```

`brate =`

`5/9`

`brnum` is 5 because the second entries differ in two bits and the third entries differ in three bits. `brate` is 5/9 because the total number of bits is 9. The total number of bits is, by definition, the number of entries in `a` or `b` times the maximum number of bits among all entries of `a` and `b`.

Performance Results via the Semianalytic Technique

The technique described in “Performance Results via Simulation” on page 15-21 works well for a large variety of communication systems, but can be prohibitively time-consuming if the system's error rate is very small (for example, 10^{-6} or smaller). This section describes how to use the semianalytic technique as an alternative way to compute error rates. For certain types of systems, the semianalytic technique can produce results much more quickly than a nonanalytic method that uses only simulated data.

The semianalytic technique uses a combination of simulation and analysis to determine the error rate of a communication system. The `semianalytic` function in Communications Toolbox helps you implement the semianalytic technique by performing some of the analysis.

When to Use the Semianalytic Technique

The semianalytic technique works well for certain types of communication systems, but not for others. The semianalytic technique is applicable if a system has all of these characteristics:

- Any effects of multipath fading, quantization, and amplifier nonlinearities must *precede* the effects of noise in the actual channel being modeled.
- The receiver is perfectly synchronized with the carrier, and timing jitter is negligible. Because phase noise and timing jitter are slow processes, they reduce the applicability of the semianalytic technique to a communication system.
- The noiseless simulation has no errors in the received signal constellation. Distortions from sources other than noise should be mild enough to keep each signal point in its correct decision region. If this is not the case, the calculated BER is too low. For instance, if the modeled system has a phase rotation that places the received signal points outside their proper decision regions, the semianalytic technique is not suitable to predict system performance.

Furthermore, the `semianalytic` function assumes that the noise in the actual channel being modeled is Gaussian. For details on how to adapt the semianalytic technique for non-Gaussian noise, see the discussion of generalized exponential distributions in [11].

Procedure for the Semianalytic Technique

The procedure below describes how you would typically implement the semianalytic technique using the `semianalytic` function:

- 1 Generate a message signal containing *at least* M^L symbols, where M is the alphabet size of the modulation and L is the length of the impulse response of the channel in symbols. A common approach is to start with an augmented binary pseudonoise (PN) sequence of total length $(\log_2 M)M^L$. An *augmented* PN sequence is a PN sequence with an extra zero appended, which makes the distribution of ones and zeros equal.
- 2 Modulate a carrier with the message signal using baseband modulation. Supported modulation types are listed on the reference page for `semianalytic`. Shape the resultant signal with rectangular pulse shaping, using the oversampling factor that you will later use to filter the modulated signal. Store the result of this step as `txsig` for later use.
- 3 Filter the modulated signal with a transmit filter. This filter is often a square-root raised cosine filter, but you can also use a Butterworth, Bessel, Chebyshev type 1 or 2, elliptic, or more general FIR or IIR filter. If you use a square-root raised cosine filter, use it on the nonoversampled modulated signal and specify the oversampling factor in the filtering function. If you use another filter type, you can apply it to the rectangularly pulse shaped signal.
- 4 Run the filtered signal through a *noiseless* channel. This channel can include multipath fading effects, phase shifts, amplifier nonlinearities, quantization, and additional filtering, but it must not include noise. Store the result of this step as `rxsig` for later use.
- 5 Invoke the `semianalytic` function using the `txsig` and `rxsig` data from earlier steps. Specify a receive filter as a pair of input arguments, unless you want to use the function's default filter. The function filters `rxsig` and then determines the error probability of each received signal point by analytically applying the Gaussian noise distribution to each point. The function averages the error probabilities over the entire received signal to determine the overall error probability. If the error probability calculated in this way is a symbol error probability, the function converts it to a bit error rate, typically by assuming Gray coding. The function returns the bit error rate (or, in the case of DQPSK modulation, an upper bound on the bit error rate).

Using Semianalytic Technique

The example below illustrates the procedure for the semianalytic technique, using 16-QAM modulation. It also compares the error rates obtained from the semianalytic technique with the theoretical error rates obtained from published formulas and computed using the `berawgn` function. The resulting plot shows that the error rates obtained using the two methods are nearly identical. The discrepancies between the theoretical and computed error rates are largely due to the phase offset inserted through a noiseless channel model.

Generate message signal of length $\geq M^L$.

```
M = 16; % Alphabet size of modulation
L = 1; % Length of impulse response of channel
msg = [0:M-1 0]; % M-ary message sequence of length  $> M^L$ 
```

Modulate the message signal using baseband modulation.

```
modsig = qammod(msg',M); % Modulate data
Nsamp = 16;
modsig = rectpulse(modsig,Nsamp); % Use rectangular pulse shaping.
```

Apply a transmit filter.

```
txsig = modsig; % No filter in this example
```

Run `txsig` through a noiseless channel.

```
rxsig = txsig*exp(1i*pi/180); % Static phase offset of 1 degree
```

Use the semianalytic function.

Specify the receive filter as a pair of input arguments. In this case, `num` and `den` describe an ideal integrator.

```
num = ones(Nsamp,1)/Nsamp;
den = 1;
EbNo = 0:20; % Range of Eb/No values under study
ber = semianalytic(txsig,rxsig,'qam',M,Nsamp,num,den,EbNo);
```

```
% For comparison, calculate theoretical BER.
```

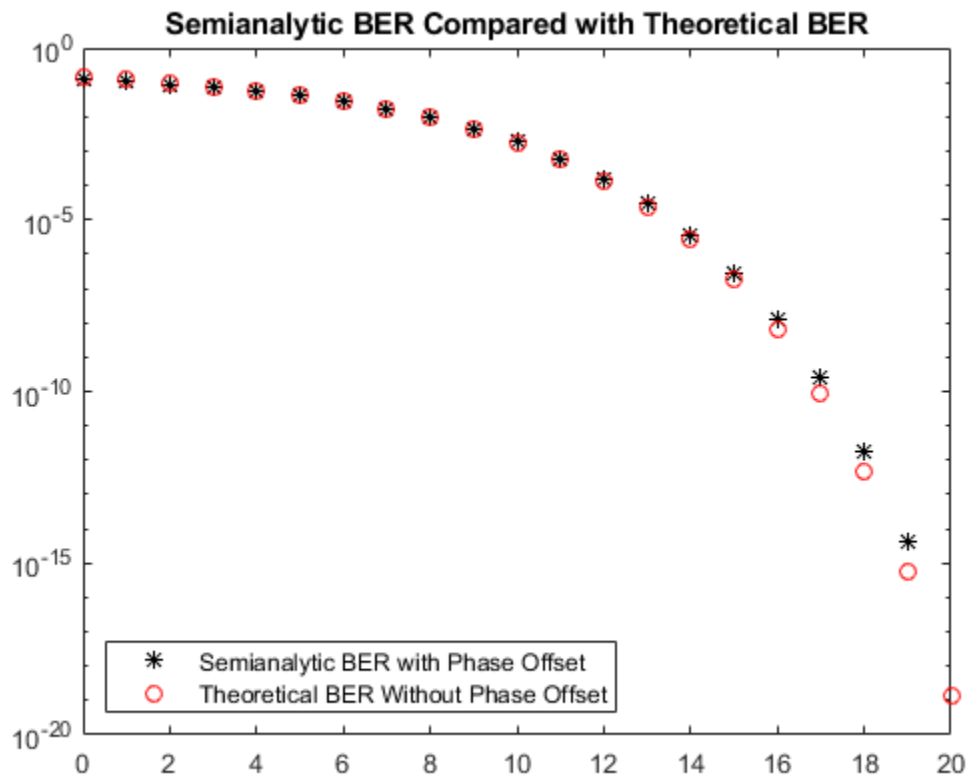
```
bertheory = berawgn(EbNo,'qam',M);
```

```
% Plot computed BER and theoretical BER.
```

```

figure; semilogy(EbNo,ber,'k*');
hold on; semilogy(EbNo,bertheory,'ro');
title('Semianalytic BER Compared with Theoretical BER');
legend('Semianalytic BER with Phase Offset',...
'Theoretical BER Without Phase Offset','Location','SouthWest');
hold off;

```



Theoretical Performance Results

- “Computing Theoretical Error Statistics” on page 15-28
- “Plotting Theoretical Error Rates” on page 15-28

- “Comparing Theoretical and Empirical Error Rates” on page 15-30

Computing Theoretical Error Statistics

While the `biterr` function discussed above can help you gather empirical error statistics, you might also compare those results to theoretical error statistics. Certain types of communication systems are associated with closed-form expressions for the bit error rate or a bound on it. The functions listed in the table below compute the closed-form expressions for some types of communication systems, where such expressions exist.

Type of Communication System	Function
Uncoded AWGN channel	<code>berawgn</code>
Coded AWGN channel	<code>bercoding</code>
Uncoded Rayleigh and Rician fading channel	<code>berfading</code>
Uncoded AWGN channel with imperfect synchronization	<code>bersync</code>

Each function's reference page lists one or more books containing the closed-form expressions that the function implements.

Plotting Theoretical Error Rates

This example uses the `bercoding` function to compute upper bounds on bit error rates for convolutional coding with a soft-decision decoder.

```
coderate = 1/4; % Code rate
```

Create a structure `dspec` with information about distance spectrum. Define the `EbNo` sweep range and generate the theoretical bound results.

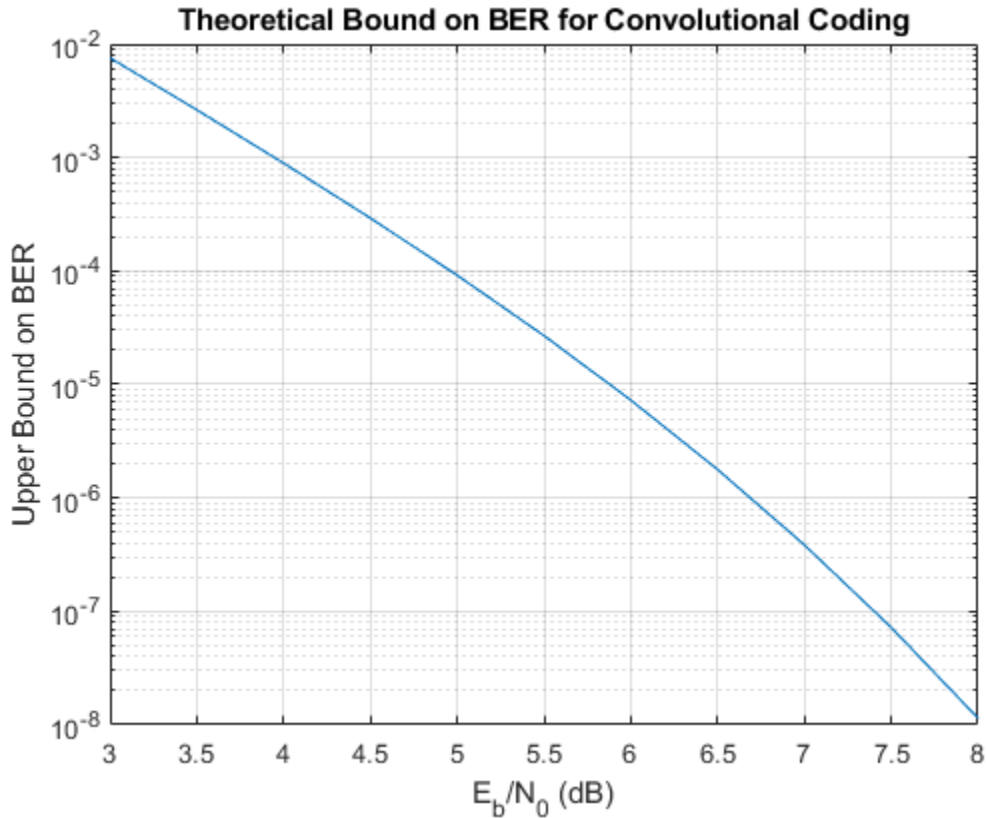
```
dspec.dfree = 10; % Minimum free distance of code
dspec.weight = [1 0 4 0 12 0 32 0 80 0 192 0 448 0 1024 ...
    0 2304 0 5120 0]; % Distance spectrum of code
EbNo = 3:0.5:8;
berbound = bercoding(EbNo, 'conv', 'soft', coderate, dspec);
```

Plot the theoretical bound results.

```
semilogy(EbNo, berbound)
xlabel('E_b/N_0 (dB)');
```



```
ylabel('Upper Bound on BER');  
title('Theoretical Bound on BER for Convolutional Coding');  
grid on;
```



References

- [1] Proakis, J. G., *Digital Communications*, 4th Ed., McGraw-Hill, 2001.
- [2] Frenger, Pal, Pal Orten, and Tony Ottosson, "Convolutional Codes with Optimum Distance Spectrum," *IEEE Communications Letters*, Vol. 3, #11, Nov. 1999, pp. 317-319

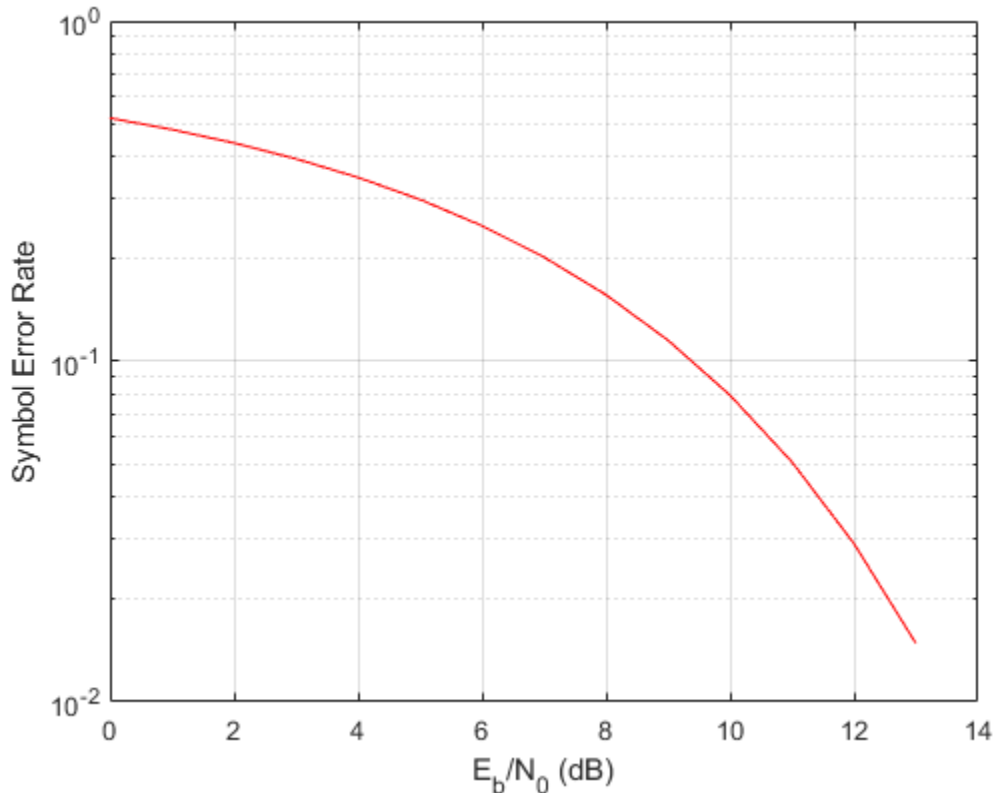
Comparing Theoretical and Empirical Error Rates

This example uses the `berawgn` function to compute symbol error rates for pulse amplitude modulation (PAM) with a series of E_b/N_0 values. For comparison, the code simulates 8-PAM with an AWGN channel and computes empirical symbol error rates. The code also plots the theoretical and empirical symbol error rates on the same set of axes.

Compute and plot theoretical error rate using `berawgn`.

```
rng('default') % Set random number seed for repeatability
M = 8;
EbNo = 0:13;
[ber, ser] = berawgn(EbNo, 'pam', M);

semilogy(EbNo, ser, 'r');
xlabel('E_b/N_0 (dB)');
ylabel('Symbol Error Rate');
grid on;
```



Compute empirical error rate by simulating. Define simulation parameters and preallocate variables to save time. Convert from E_b/N_0 to SNR. Since $N_0 = 2 \cdot \text{noiseVariance}^2$, we must add 3 dB to get SNR. For details, see Proakis' book listed in "Selected Bibliography for Performance Evaluation."

```
n = 10000; % Number of symbols to process
k = log2(M); % Number of bits per symbol
snr = EbNo+3+10*log10(k);
ynoisyy = zeros(n,length(snr));
z = zeros(n,length(snr));
berVec = zeros(3,length(EbNo));
```

Create objects for an AWGN channel error rate calculator. The error rate calculator is used to compare decoded symbols to the original transmitted symbols.

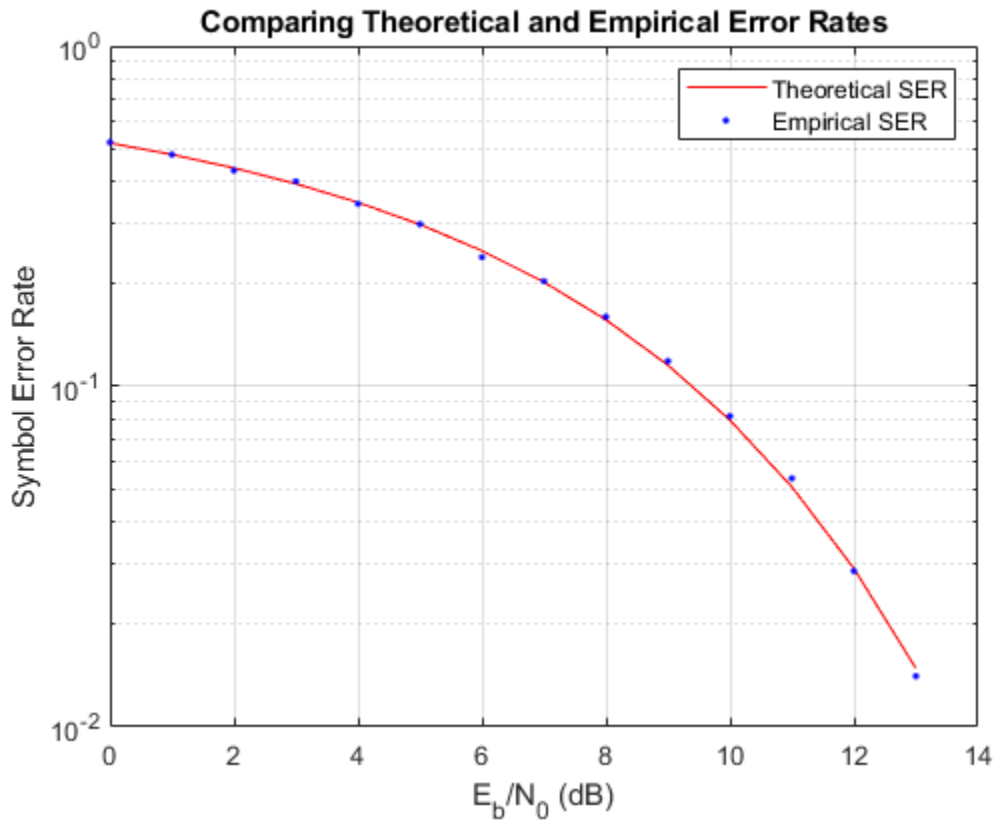
```
awgnchan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)');  
errrcalc = comm.ErrorRate;
```

Generate random data message and apply PAM modulation. Normalize channel to signal power. Loop simulation to generate BERs over the range of SNR values.

```
x = randi([0 M-1],n,1); % Create message signal.  
y = pammod(x,M); % Modulate.  
awgnchan.SignalPower = (real(y)' * real(y))/ length(real(y));  
  
for jj = 1:length(snr)  
    reset(errrcalc)  
    awgnchan.SNR = snr(jj); % Assign Channel SNR  
    ynoisy(:,jj) = awgnchan(real(y)); % Add AWGN  
    z(:,jj) = pamdemod(complex(ynoisyl(:,jj)),M); % Demodulate.  
  
    % Compute symbol error rate from simulation.  
    berVec(:,jj) = errrcalc(x,z(:,jj));  
end
```

Compare theoretical and empirical results.

```
hold on;  
semilogy(EbNo,berVec(1,:), 'b. ');  
legend('Theoretical SER', 'Empirical SER');  
title('Comparing Theoretical and Empirical Error Rates');  
hold off;
```



Error Rate Plots

- “Section Overview” on page 15-33
- “Creating Error Rate Plots Using semilogy” on page 15-34
- “Curve Fitting for Error Rate Plots” on page 15-34
- “Curve Fitting An Error Rate Plot” on page 15-35

Section Overview

Error rate plots provide a visual way to examine the performance of a communication system, and they are often included in publications. This section mentions some of the

tools you can use to create error rate plots, modify them to suit your needs, and do curve fitting on error rate data on page 15-34. It also provides an example of curve fitting on page 15-35. For more detailed discussions about the more general plotting capabilities in MATLAB, see the MATLAB documentation set.

Creating Error Rate Plots Using `semilogy`

In many error rate plots, the horizontal axis indicates E_b/N_0 values in dB and the vertical axis indicates the error rate using a logarithmic (base 10) scale. To see an example of such a plot, as well as the code that creates it, see “Comparing Theoretical and Empirical Error Rates” on page 15-30. The part of that example that creates the plot uses the `semilogy` function to produce a logarithmic scale on the vertical axis and a linear scale on the horizontal axis.

Other examples that illustrate the use of `semilogy` are in these sections:

- “Using Semianalytic Technique” on page 15-26, which also illustrates
 - Plotting two sets of data on one pair of axes
 - Adding a title
 - Adding a legend
- “Plotting Theoretical Error Rates” on page 15-28, which also illustrates
 - Adding axis labels
 - Adding grid lines

Curve Fitting for Error Rate Plots

Curve fitting is useful when you have a small or imperfect data set but want to plot a smooth curve for presentation purposes. The `berfit` function in Communications Toolbox offers curve-fitting capabilities that are well suited to the situation when the empirical data describes error rates at different E_b/N_0 values. This function enables you to

- Customize various relevant aspects of the curve-fitting process, such as the type of closed-form function (from a list of preset choices) used to generate the fit.
- Plot empirical data along with a curve that `berfit` fits to the data.
- Interpolate points on the fitted curve between E_b/N_0 values in your empirical data set to make the plot smoother looking.
- Collect relevant information about the fit, such as the numerical values of points along the fitted curve and the coefficients of the fit expression.

Note The `berfit` function is intended for curve fitting or interpolation, *not* extrapolation. Extrapolating BER data beyond an order of magnitude below the smallest empirical BER value is inherently unreliable.

For a full list of inputs and outputs for `berfit`, see its reference page.

Curve Fitting An Error Rate Plot

This example simulates a simple differential binary phase shift keying (DBPSK) communication system and plots error rate data for a series of E_b/N_0 values. It uses the `berfit` function to fit a curve to the somewhat rough set of empirical error rates.

Initialize Simulation Parameters

Specify the input signal message length, modulation order, range of E_b/N_0 values to consider, and the minimum number of errors that must occur before the simulation computes an error rate for that E_b/N_0 value. Preallocate variables for results and interim results.

For statistically accurate error rate results, the minimum number of errors should be on the order of 100 but this simulation uses a small number of errors to shorten run time and to illustrate how curve fitting can smooth out a rough data set.

```
siglen = 100000; % Number of bits in each trial
M = 2; % DBPSK is binary.
EbN0vec = 0:5; % Vector of EbN0 values
minnumerr = 5; % Compute BER only after 5 errors occur.
numEbN0 = length(EbN0vec); % Number of EbN0 values

ber = zeros(1,numEbN0); % final BER values
berVec = zeros(3,numEbN0); % Updated BER values
intv = cell(1,numEbN0); % Cell array of confidence intervals
```

Create an error rate calculator System object™.

```
errorCalc = comm.ErrorRate;
```

Simulation Loop

Simulate the DBPSK modulated communication system and compute the BER using a `for` loop to vary the E_b/N_0 value. The inner `while` loop ensures that a minimum number of bit errors occur for each E_b/N_0 value. Error rate statistics are saved for each E_b/N_0 value to use when curve fitting and plotting.

```

% Loop over the vector of EbN0 values.
for jj = 1:numEbN0
    EbN0 = EbN0vec(jj);
    snr = EbN0; % Because of binary modulation
    reset(errorCalc)
    awgnChan.SNR = snr; % Assign Channel SNR

    while (berVec(2,jj) < minnumerr)
        msg = randi([0,M-1],siglen,1); % Generate message sequence
        % txsig = step(hMod,msg); % Modulate
        txsig = dpskmod(msg,M); % Modulate
        awgnChan.SignalPower = (txsig'*txsig)/length(txsig); % Calculate and assign signal power
        % rxsig = awgnChan(txsig); % Add noise.
        rxsig = awgn(txsig,snr,'measured'); % Add noise.
        % decodmsg = step(hDemod, rxsig); % Demodulate.
        decodmsg = dpskdemod(rxsig,M); % Demodulate.
        berVec(:,jj) = errorCalc(msg,decodmsg); % Calculate BER
    end
end

```

Use `berconfint` to compute error rate at a 98% confidence interval for the E_b/N_0 value.

```

[ber(jj), intv1] = berconfint(berVec(2,jj),berVec(3,jj),0.98);
intv{jj} = intv1;
disp(['EbN0 = ' num2str(EbN0) ' dB, ' num2str(berVec(2,jj)) ...
    ' errors, BER = ' num2str(ber(jj))])
end

```

```

EbN0 = 0 dB, 18392 errors, BER = 0.18392
EbN0 = 1 dB, 14307 errors, BER = 0.14307
EbN0 = 2 dB, 10190 errors, BER = 0.1019
EbN0 = 3 dB, 6940 errors, BER = 0.0694
EbN0 = 4 dB, 4151 errors, BER = 0.04151
EbN0 = 5 dB, 2098 errors, BER = 0.02098

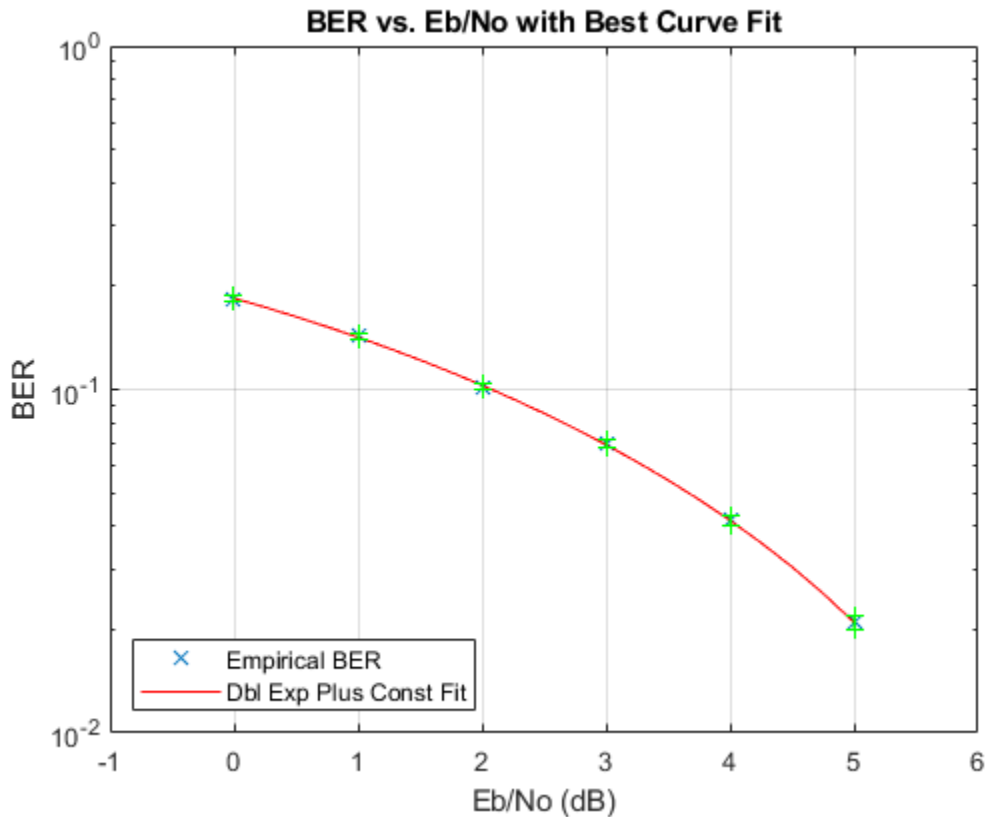
```

Use `berfit` to plot the best fitted curve, interpolating to get a smooth plot. Add confidence intervals to the plot.

```

fitEbN0 = EbN0vec(1):0.25:EbN0vec(end); % Interpolation values
berfit(EbN0vec,ber,fitEbN0);
hold on;
for jj=1:numEbN0
    semilogy([EbN0vec(jj) EbN0vec(jj)],intv{jj},'g-+');
end
hold off;

```

BERTool

The command **bertool** launches the Bit Error Rate Analysis Tool (BERTool) application.

The application enables you to analyze the bit error rate (BER) performance of communications systems. BERTool computes the BER as a function of signal-to-noise ratio. It analyzes performance either with Monte-Carlo simulations of MATLAB functions and Simulink models or with theoretical closed-form expressions for selected types of communication systems.

Using BERTool you can:

- Generate BER data for a communication system using
 - Closed-form expressions for theoretical BER performance of selected types of communication systems.
 - The semianalytic technique.
 - Simulations contained in MATLAB simulation functions or Simulink models. After you create a function or model that simulates the system, BERTool iterates over your choice of E_b/N_0 values and collects the results.
- Plot one or more BER data sets on a single set of axes. For example, you can graphically compare simulation data with theoretical results or simulation data from a series of similar models of a communication system.
- Fit a curve to a set of simulation data.
- Send BER data to the MATLAB workspace or to a file for any further processing you might want to perform.

Note BERTool is designed for analyzing bit error rates only, not symbol error rates, word error rates, or other types of error rates. If, for example, your simulation computes a symbol error rate (SER), convert the SER to a BER before using the simulation with BERTool.

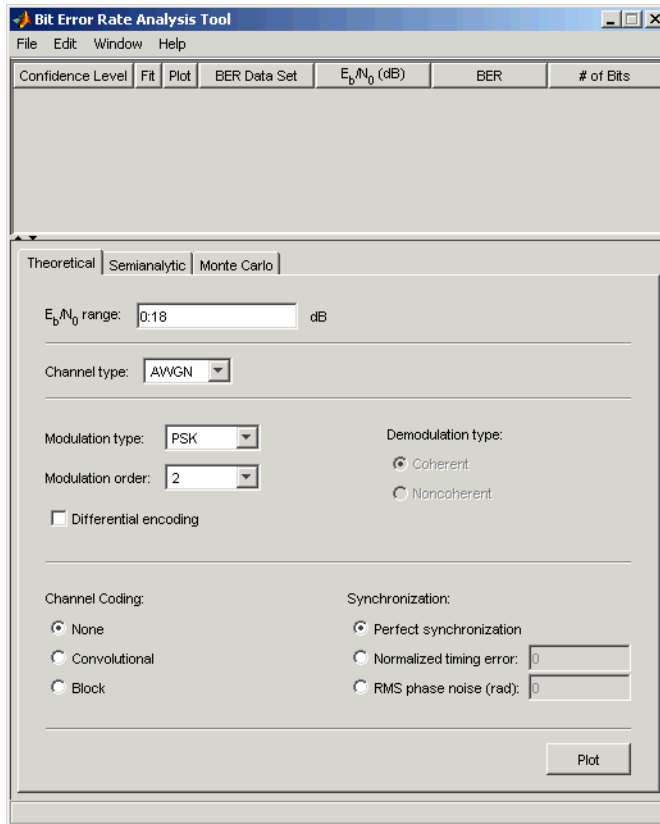
The following sections describe the Bit Error Rate Analysis Tool (BERTool) and provide examples showing how to use its GUI.

- “Start BERTool” on page 15-38
- “The BERTool Environment” on page 15-39
- “Computing Theoretical BERs” on page 15-42
- “Using the Semianalytic Technique to Compute BERs” on page 15-48
- “Run MATLAB Simulations” on page 15-53
- “Use Simulation Functions with BERTool” on page 15-59
- “Run Simulink Simulations” on page 15-66
- “Use Simulink Models with BERTool” on page 15-71
- “Manage BER Data” on page 15-79

Start BERTool

To open BERTool, type

bertool



The BERTool Environment

- “Components of BERTool” on page 15-39
- “Interaction Among BERTool Components” on page 15-41

Components of BERTool

- A data viewer at the top. It is initially empty.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits

After you instruct BERTool to generate one or more BER data sets, they appear in the data viewer. An example that shows how data sets look in the data viewer is in “Example: Using a MATLAB Simulation with BERTool” on page 15-53.

- A set of tabs on the bottom. Labeled **Theoretical**, **Semianalytic**, and **Monte Carlo**, the tabs correspond to the different methods by which BERTool can generate BER data.

Theoretical | Semianalytic | Monte Carlo

E_b/N_0 range: 0.18 dB

Channel type: AWGN

Modulation type: PSK

Modulation order: 2

Demodulation type: Coherent Noncoherent

Differential encoding

Channel Coding: None Convolutional Block

Synchronization: Perfect synchronization Normalized timing error: 0 RMS phase noise (rad): 0

Plot

Note When using BERTool to compare theoretical results and Monte Carlo results, the Simulink model provided must model exactly the system defined by the parameters on the **Theoretical** tab.

To learn more about each of the methods, see

- “Computing Theoretical BERs” on page 15-42

- “Using the Semianalytic Technique to Compute BERs” on page 15-48
- “Run MATLAB Simulations” on page 15-53 or “Run Simulink Simulations” on page 15-66
- A separate BER Figure window, which displays some or all of the BER data sets that are listed in the data viewer. BERTool opens the BER Figure window after it has at least one data set to display, so you do not see the BER Figure window when you first open BERTool. For an example of how the BER Figure window looks, see “Example: Using the Theoretical Tab in BERTool” on page 15-43.

Interaction Among BERTool Components

The components of BERTool act as one integrated tool. These behaviors reflect their integration:

- If you select a data set in the data viewer, BERTool reconfigures the tabs to reflect the parameters associated with that data set and also highlights the corresponding data in the BER Figure window. This is useful if the data viewer displays multiple data sets and you want to recall the meaning and origin of each data set.
- If you click data plotted in the BER Figure window, BERTool reconfigures the tabs to reflect the parameters associated with that data and also highlights the corresponding data set in the data viewer.

Note You cannot click on a data point while BERTool is generating Monte Carlo simulation results. You must wait until the tool generates all data points before clicking for more information.

- If you configure the **Semianalytic** or **Theoretical** tab in a way that is already reflected in an existing data set, BERTool highlights that data set in the data viewer. This prevents BERTool from duplicating its computations and its entries in the data viewer, while still showing you the results that you requested.
- If you close the BER Figure window, then you can reopen it by choosing **BER Figure** from the **Window** menu in BERTool.
- If you select options in the data viewer that affect the BER plot, the BER Figure window reflects your selections immediately. Such options relate to data set names, confidence intervals, curve fitting, and the presence or absence of specific data sets in the BER plot.

Note If you want to observe the integration yourself but do not yet have any data sets in BERTool, then first try the procedure in “Example: Using the Theoretical Tab in BERTool” on page 15-43.

Note If you save the BER Figure window using the window's **File** menu, the resulting file contains the contents of the window but not the BERTool data that led to the plot. To save an entire BERTool session, see “Saving a BERTool Session” on page 15-82.

Computing Theoretical BERs

- “Section Overview” on page 15-42
- “Example: Using the Theoretical Tab in BERTool” on page 15-43
- “Available Sets of Theoretical BER Data” on page 15-46

Section Overview

You can use BERTool to generate and analyze theoretical BER data. Theoretical data is useful for comparison with your simulation results. However, closed-form BER expressions exist only for certain kinds of communication systems.

To access the capabilities of BERTool related to theoretical BER data, use the following procedure:

- 1** Open BERTool, and go to the **Theoretical** tab.

The screenshot shows the 'Theoretical' tab of the BERTool interface. The parameters are configured as follows:

- Tab Selection:** Theoretical (selected), Semianalytic, Monte Carlo
- E_b/N_0 range:** 0.18 dB
- Channel type:** AWGN
- Modulation type:** PSK
- Modulation order:** 2
- Demodulation type:** Coherent (selected), Noncoherent
- Differential encoding:**
- Channel Coding:** None (selected), Convolutional, Block
- Synchronization:** Perfect synchronization (selected), Normalized timing error: 0, RMS phase noise (rad): 0
- Plot:** A button is located at the bottom right of the configuration area.

- 2 Set the parameters to reflect the system whose performance you want to analyze. Some parameters are visible and active only when other parameters have specific values. See “Available Sets of Theoretical BER Data” on page 15-46 for details.
- 3 Click **Plot**.

For an example that shows how to generate and analyze theoretical BER data via BERTool, see “Example: Using the Theoretical Tab in BERTool” on page 15-43.

Also, “Available Sets of Theoretical BER Data” on page 15-46 indicates which combinations of parameters are available on the **Theoretical** tab and which underlying functions perform computations.

Example: Using the Theoretical Tab in BERTool

This example illustrates how to use BERTool to generate and plot theoretical BER data. In particular, the example compares the performance of a communication system that uses an AWGN channel and QAM modulation of different orders.

Running the Theoretical Example

- 1 Open BERTool, and go to the **Theoretical** tab.
- 2 Set the parameters as shown in the following figure.

Theoretical | Semianalytic | Monte Carlo

E_b/N_0 range: 0.18 dB

Channel type: AWGN

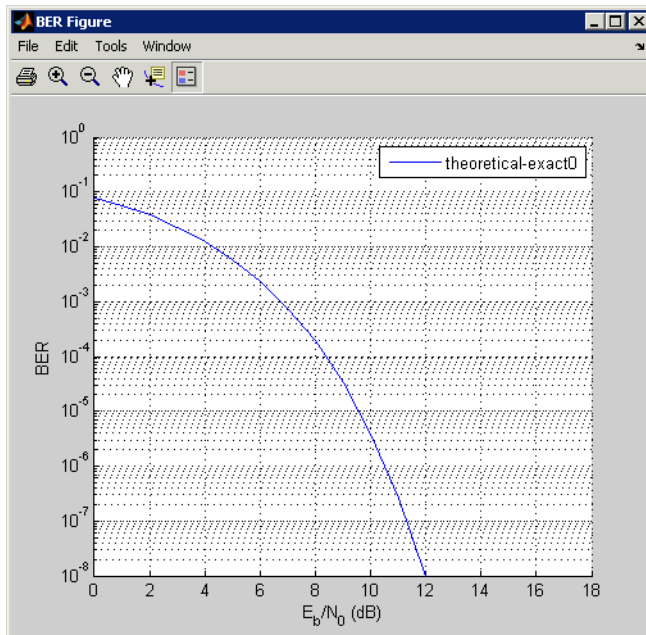
Modulation type: QAM

Modulation order: 4

3 Click **Plot**.

BERTool creates an entry in the data viewer and plots the data in the BER Figure window. Even though the parameters request that E_b/N_0 go up to 18, BERTool plots only those BER values that are at least 10^{-8} . The following figures illustrate this step.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits
		<input checked="" type="checkbox"/>	theoretical-exact0	[0 1 2 3 4 5 6...	[0.0786 0.0...	N/A



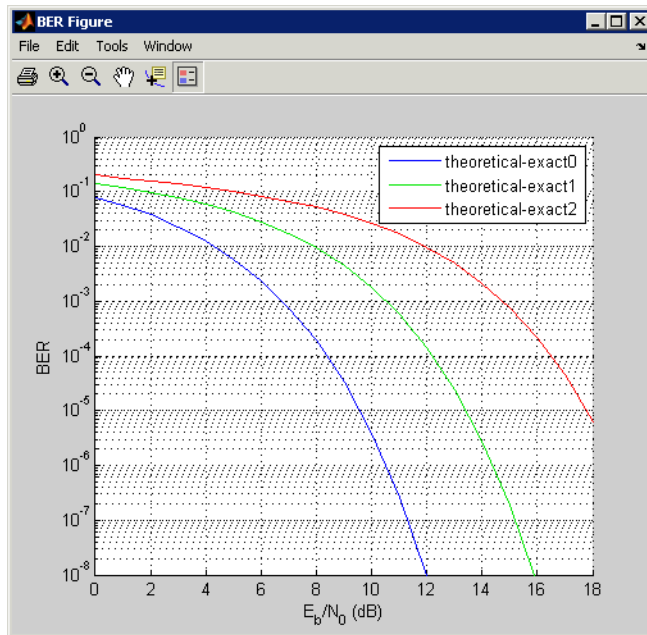
4 Change the **Modulation order** parameter to 16, and click **Plot**.

BERTool creates another entry in the data viewer and plots the new data in the same BER Figure window (not pictured).

- Change the **Modulation order** parameter to 64, and click **Plot**.

BERTool creates another entry in the data viewer and plots the new data in the same BER Figure window, as shown in the following figures.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	theoretical-exact0	[0 1 2 3 4 5 6...	[0.0786 0.0...	N/A
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	theoretical-exact1	[0 1 2 3 4 5 6...	[0.1409 0.1...	N/A
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	theoretical-exact2	[0 1 2 3 4 5 6...	[0.1998 0.1...	N/A



- To recall which value of **Modulation order** corresponds to a given curve, click the curve. BERTool responds by adjusting the parameters in the **Theoretical** tab to reflect the values that correspond to that curve.
- To remove the last curve from the plot (but not from the data viewer), clear the check box in the last entry of the data viewer in the **Plot** column. To restore the curve to the plot, select the check box again.

Available Sets of Theoretical BER Data

BERTool can generate a large set of theoretical bit-error rates, but not all combinations of parameters are currently supported. The **Theoretical** tab adjusts itself to your choices, so that the combination of parameters is always valid. You can set the **Modulation order** parameter by selecting a choice from the menu or by typing a value in the field. The **Normalized timing error** must be between 0 and 0.5.

BERTool assumes that Gray coding is used for all modulations.

For QAM, when $\log_2 M$ is odd (M being the modulation order), a rectangular constellation is assumed.

Combinations of Parameters for AWGN Channel Systems

The following table lists the available sets of theoretical BER data for systems that use an AWGN channel.

Modulation	Modulation Order	Other Choices
PSK	2, 4	Differential or nondifferential encoding.
	8, 16, 32, 64, or a higher power of 2	
OQPSK	4	Differential or nondifferential encoding.
DPSK	2, 4, 8, 16, 32, 64, or a higher power of 2	
PAM	2, 4, 8, 16, 32, 64, or a higher power of 2	
QAM	4, 8, 16, 32, 64, 128, 256, 512, 1024, or a higher power of 2	
FSK	2	Orthogonal or nonorthogonal; Coherent or Noncoherent demodulation.
	4, 8, 16, 32, or a higher power of 2	Orthogonal; Coherent demodulation.
	4, 8, 16, 32, or 64	Orthogonal; Noncoherent demodulation.
MSK	2	Coherent conventional or precoded MSK; Noncoherent precoded MSK.

Modulation	Modulation Order	Other Choices
CPFSK	2, 4, 8, 16, or a higher power of 2	Modulation index > 0.

BER results are also available for the following:

- block and convolutional coding with hard-decision decoding for all modulations except CPFSK
- block coding with soft-decision decoding for all binary modulations (including 4-PSK and 4-QAM) except CPFSK, noncoherent non-orthogonal FSK, and noncoherent MSK
- convolutional coding with soft-decision decoding for all binary modulations (including 4-PSK and 4-QAM) except CPFSK
- uncoded nondifferentially-encoded 2-PSK with synchronization errors

For more information about specific combinations of parameters, including bibliographic references that contain closed-form expressions, see the reference pages for the following functions:

- `berawgn` — For systems with no coding and perfect synchronization
- `bercoding` — For systems with channel coding
- `bersync` — For systems with BPSK modulation, no coding, and imperfect synchronization

Combinations of Parameters for Rayleigh and Rician Channel Systems

The following table lists the available sets of theoretical BER data for systems that use a Rayleigh or Rician channel.

When diversity is used, the SNR on each diversity branch is derived from the SNR at the input of the channel (E_b/N_0) divided by the diversity order.

Modulation	Modulation Order	Other Choices
PSK	2	Differential or nondifferential encoding Diversity order ≥ 1 In the case of nondifferential encoding, diversity order being 1, and Rician fading, a value for RMS phase noise (in radians) can be specified.
	4, 8, 16, 32, 64, or a higher power of 2	Diversity order ≥ 1
OQPSK	4	Diversity order ≥ 1
DPSK	2, 4, 8, 16, 32, 64, or a higher power of 2	Diversity order ≥ 1
PAM	2, 4, 8, 16, 32, 64, or a higher power of 2	Diversity order ≥ 1
QAM	4, 8, 16, 32, 64, 128, 256, 512, 1024, or a higher power of 2	Diversity order ≥ 1
FSK	2	Correlation coefficient $\in [-1, 1]$. Coherent or Noncoherent demodulation Diversity order ≥ 1 In the case of a nonzero correlation coefficient and noncoherent demodulation, the diversity order is 1 only.
	4, 8, 16, 32, or a higher power of 2	Noncoherent demodulation only. Diversity order ≥ 1

For more information about specific combinations of parameters, including bibliographic references that contain closed-form expressions, see the reference page for the BER fading function.

Using the Semianalytic Technique to Compute BERs

- “Section Overview” on page 15-49

- “Example: Using the Semianalytic Tab in BERTool” on page 15-49
- “Procedure for Using the Semianalytic Tab in BERTool” on page 15-51

Section Overview

You can use BERTool to generate and analyze BER data via the semianalytic technique. The semianalytic technique is discussed in “Performance Results via the Semianalytic Technique” on page 15-24, and “When to Use the Semianalytic Technique” on page 15-24 is particularly relevant as background material.

To access the semianalytic capabilities of BERTool, open the **Semianalytic** tab.

The screenshot shows the BERTool interface with the Semianalytic tab selected. The parameters are as follows:

- Tab:** Semianalytic
- E_b/N_0 range:** 0:18 dB
- Channel type:** AWGN
- Modulation type:** PSK
- Modulation order:** 2
- Differential encoding:**
- Samples per symbol:** 16
- Transmitted signal:** `rectpulse(pskmod(randint(16, 1, 2, 9973), 2), 16)`
- Received signal:** `rectpulse(pskmod(randint(16, 1, 2, 9973), 2), 16)`
- Receiver filter coefficients:**
 - Numerator:** `ones(16, 1) / 16`
 - Denominator:** 1
- Plot button:** Present at the bottom right.

For further details about how BERTool applies the semianalytic technique, see the reference page for the `semianalytic` function, which BERTool uses to perform computations.

Example: Using the Semianalytic Tab in BERTool

This example illustrates how BERTool applies the semianalytic technique, using 16-QAM modulation. This example is a variation on the example in “Using Semianalytic Technique” on page 15-26, but it is tailored to use BERTool instead of using the semianalytic function directly.

Running the Semianalytic Example

- 1 Set up the transmitted and received signals. Generate message signal of length $\geq M^L$. Modulate the message signal using baseband modulation. Apply pulse shaping with no further transmit filtering. Pass the transmission signal through a noiseless channel.

```
M = 16; % Alphabet size of modulation
L = 1; % Length of impulse response of channel
msg = [0:M-1 0]; % M-ary message sequence of length > M^L
```

```
modsig = qammod(msg,M); % Use 16-QAM.
```

```
Nsamp = 16;
modsig = rectpulse(modsig,Nsamp); % Use rectangular pulse shaping.
txsig = modsig; % No filter in this example
```

```
rxsig = txsig*exp(j*pi/180); % Static phase offset of 1 degree
```

- 2 Open BERTool and go to the **Semianalytic** tab.
- 3 Set parameters as shown in the following figure.

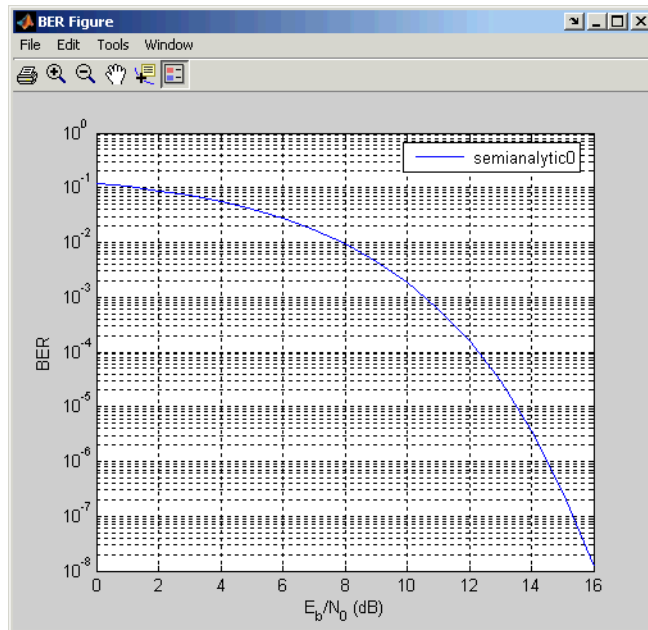
- 4 Click **Plot**.

Visible Results of the Semianalytic Example

After you click **Plot**, BERTool creates a listing for the resulting data in the data viewer.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits
	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	semianalytic0	[0 1 2 3 4 5 6...]	[0.1199 0.1...]	[272]

BERTool plots the data in the BER Figure window.



Procedure for Using the Semianalytic Tab in BERTool

The procedure below describes how you typically implement the semianalytic technique using BERTool:

- 1 Generate a message signal containing *at least* M^L symbols, where M is the alphabet size of the modulation and L is the length of the impulse response of the channel in symbols. A common approach is to start with an augmented binary pseudonoise (PN) sequence of total length $(\log_2 M)M^L$. An *augmented* PN sequence is a PN sequence with an extra zero appended, which makes the distribution of ones and zeros equal.
- 2 Modulate a carrier with the message signal using baseband modulation. Supported modulation types are listed on the reference page for semianalytic. Shape the resultant signal with rectangular pulse shaping, using the oversampling factor that you will later use to filter the modulated signal. Store the result of this step as `txsig` for later use.
- 3 Filter the modulated signal with a transmit filter. This filter is often a square-root raised cosine filter, but you can also use a Butterworth, Bessel, Chebyshev type 1 or 2, elliptic, or more general FIR or IIR filter. If you use a square-root raised cosine

filter, use it on the nonoversampled modulated signal and specify the oversampling factor in the filtering function. If you use another filter type, you can apply it to the rectangularly pulse shaped signal.

- 4 Run the filtered signal through a *noiseless* channel. This channel can include multipath fading effects, phase shifts, amplifier nonlinearities, quantization, and additional filtering, but it must not include noise. Store the result of this step as `rxsig` for later use.
- 5 On the **Semianalytic** tab of BERTool, enter parameters as in the table below.

Parameter Name	Meaning
Eb/No range	A vector that lists the values of E_b/N_0 for which you want to collect BER data. The value in this field can be a MATLAB expression or the name of a variable in the MATLAB workspace.
Modulation type	These parameters describe the modulation scheme you used earlier in this procedure.
Modulation order	
Differential encoding	This check box, which is visible and active for MSK and PSK modulation, enables you to choose between differential and nondifferential encoding.
Samples per symbol	The number of samples per symbol in the transmitted signal. This value is also the sampling rate of the transmitted and received signals, in Hz.
Transmitted signal	The <code>txsig</code> signal that you generated earlier in this procedure
Received signal	The <code>rxsig</code> signal that you generated earlier in this procedure
Numerator	Coefficients of the receiver filter that BERTool applies to the received signal
Denominator	

Note Consistency among the values in the GUI is important. For example, if the signal referenced in the **Transmitted signal** field was generated using DPSK and you set **Modulation type** to MSK, the results might not be meaningful.

- 6 Click **Plot**.

Semianalytic Computations and Results

After you click **Plot**, BERTool performs these tasks:

- Filters `rxsig` and then determines the error probability of each received signal point by analytically applying the Gaussian noise distribution to each point. BERTool averages the error probabilities over the entire received signal to determine the overall error probability. If the error probability calculated in this way is a symbol error probability, BERTool converts it to a bit error rate, typically by assuming Gray coding. (If the modulation type is DQPSK or cross QAM, the result is an upper bound on the bit error rate rather than the bit error rate itself.)
- Enters the resulting BER data in the data viewer of the BERTool window.
- Plots the resulting BER data in the BER Figure window.

Run MATLAB Simulations

- “Section Overview” on page 15-53
- “Example: Using a MATLAB Simulation with BERTool” on page 15-53
- “Varying the Stopping Criteria” on page 15-56
- “Plotting Confidence Intervals” on page 15-57
- “Fitting BER Points to a Curve” on page 15-58

Section Overview

You can use BERTool in conjunction with your own MATLAB simulation functions to generate and analyze BER data. The MATLAB function simulates the communication system whose performance you want to study. BERTool invokes the simulation for E_b/N_0 values that you specify, collects the BER data from the simulation, and creates a plot. BERTool also enables you to easily change the E_b/N_0 range and stopping criteria for the simulation.

To learn how to make your own simulation functions compatible with BERTool, see “Use Simulation Functions with BERTool” on page 15-59.

Example: Using a MATLAB Simulation with BERTool

This example illustrates how BERTool can run a MATLAB simulation function. The function is `viterbisim`, one of the demonstration files included with Communications Toolbox software.

To run this example, follow these steps:

- 1 Open BERTool and go to the **Monte Carlo** tab. (The default parameters depend on whether you have Communications Toolbox software installed. Also note that the **BER variable name** field applies only to Simulink models.)
- 2 Set parameters as shown in the following figure.

The screenshot shows the BERTool GUI with the Monte Carlo tab selected. The parameters are as follows:

- E_b/N₀ range:** 0:5 dB
- Simulation M-file or model:** viterbisim.m
- BER variable name:** (empty)
- Simulation limits:**
 - Number of errors: 100
 - or
 - Number of bits: 1e8

Buttons for 'Run' and 'Stop' are located at the bottom right of the window.

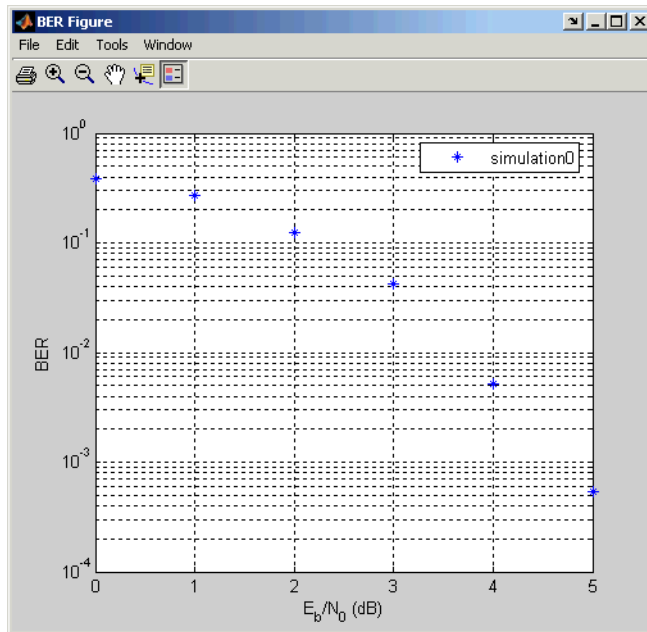
- 3 Click **Run**.

BERTool runs the simulation function once for each specified value of E_b/N_0 and gathers BER data. (While BERTool is busy with this task, it cannot process certain other tasks, including plotting data from the other tabs of the GUI.)

Then BERTool creates a listing in the data viewer.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits
off	<input type="checkbox"/>	<input checked="" type="checkbox"/>	simulation0	[0 1 2 3 4 5]	[0.3743 0.2...	[10000 1000...

BERTool plots the data in the BER Figure window.

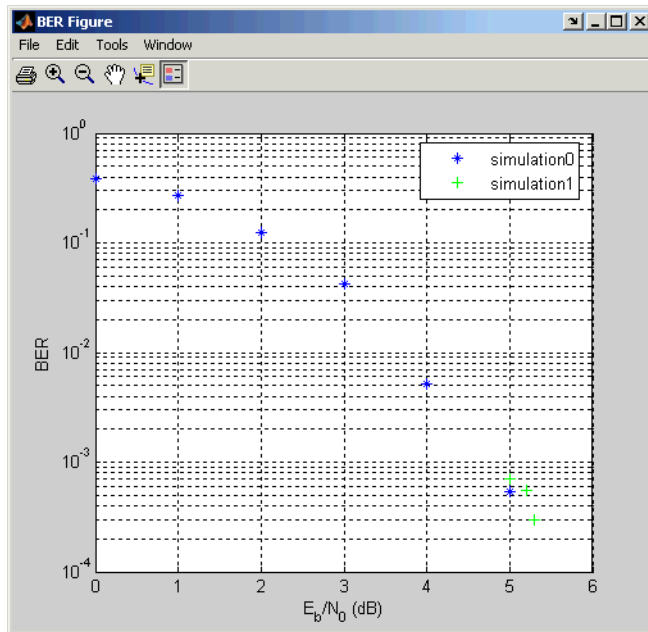


- 4 To change the range of E_b/N_0 while reducing the number of bits processed in each case, type [5 5.2 5.3] in the **Eb/No range** field, type 1e5 in the **Number of bits** field, and click **Run**.

BERTool runs the simulation function again for each new value of E_b/N_0 and gathers new BER data. Then BERTool creates another listing in the data viewer.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits
off	<input type="checkbox"/>	<input checked="" type="checkbox"/>	simulation0	[0 1 2 3 4 5]	[0.3739 0.2...	[10000 1000...
off	<input type="checkbox"/>	<input checked="" type="checkbox"/>	simulation1	[5 5.2 5.3]	[3.37E-4 5.4...	[109680 109...

BERTool plots the data in the BER Figure window, adjusting the horizontal axis to accommodate the new data.



The two points corresponding to 5 dB from the two data sets are different because the smaller value of **Number of bits** in the second simulation caused the simulation to end before observing many errors. To learn more about the criteria that BERTool uses for ending simulations, see “Varying the Stopping Criteria” on page 15-56.

For another example that uses BERTool to run a MATLAB simulation function, see “Example: Prepare a Simulation Function for Use with BERTool” on page 15-62.

Varying the Stopping Criteria

When you create a MATLAB simulation function for use with BERTool, you must control the flow so that the simulation ends when it either detects a target number of errors or processes a maximum number of bits, whichever occurs first. To learn more about this requirement, see “Requirements for Functions” on page 15-59; for an example, see “Example: Prepare a Simulation Function for Use with BERTool” on page 15-62.

After creating your function, set the target number of errors and the maximum number of bits in the **Monte Carlo** tab of BERTool.

Simulation limits:

Number of errors:

or

Number of bits:

Typically, a **Number of errors** value of at least 100 produces an accurate error rate. The **Number of bits** value prevents the simulation from running too long, especially at large values of E_b/N_0 . However, if the **Number of bits** value is so small that the simulation collects very few errors, the error rate might not be accurate. You can use confidence intervals to gauge the accuracy of the error rates that your simulation produces; the larger the confidence interval, the less accurate the computed error rate.

As an example, follow the procedure described in “Example: Using a MATLAB Simulation with BERTool” on page 15-53 and set **Confidence Level** to 95 for each of the two data sets. The confidence intervals for the second data set are larger than those for the first data set. This is because the second data set uses a small value for **Number of bits** relative to the communication system properties and the values in **Eb/No range**, resulting in BER values based on only a small number of observed errors.

Note You can also use the **Stop** button in BERTool to stop a series of simulations prematurely, as long as your function is set up to detect and react to the button press.

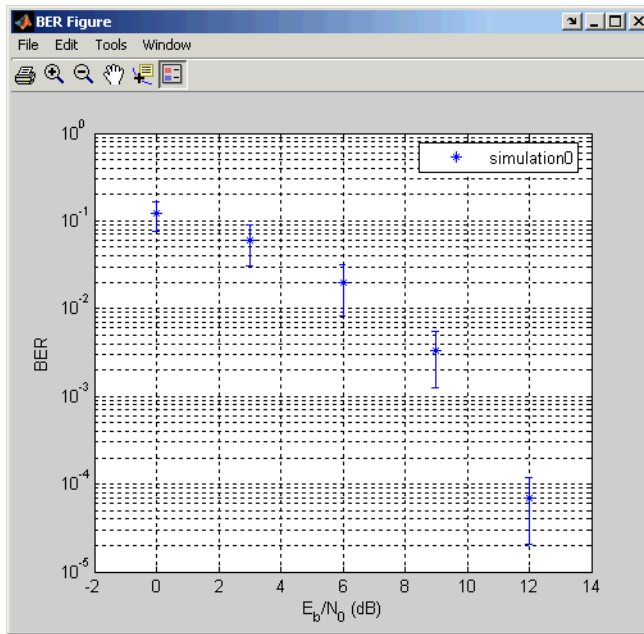
Plotting Confidence Intervals

After you run a simulation with BERTool, the resulting data set in the data viewer has an active menu in the **Confidence Level** column. The default value is **off**, so that the simulation data in the BER Figure window does not show confidence intervals.

To show confidence intervals in the BER Figure window, set **Confidence Level** to a numerical value: 90%, 95%, or 99%.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits
off	<input type="checkbox"/>	<input checked="" type="checkbox"/>	simulation0	0.3:12	[0.12 0.06 0.02...	[300 300 600 ...
off						
90%						
95%						
99%						

The plot in the BER Figure window responds immediately to your choice. A sample plot is below.



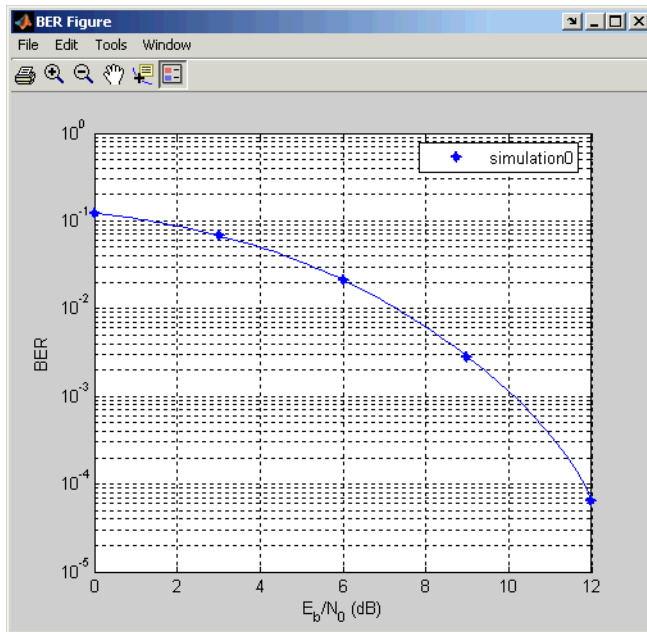
For an example that plots confidence intervals for a Simulink simulation, see “Example: Using a Simulink Model with BERTool” on page 15-67.

To find confidence intervals for levels not listed in the **Confidence Level** menu, use the `berconfint` function.

Fitting BER Points to a Curve

After you run a simulation with BERTool, the BER Figure window plots individual BER data points. To fit a curve to a data set that contains at least four points, select the box in the **Fit** column of the data viewer.

The plot in the BER Figure window responds immediately to your choice. A sample plot is below.



For an example that performs curve fitting for data from a Simulink simulation and generates the plot shown above, see “Example: Using a Simulink Model with BERTool” on page 15-67. For an example that performs curve fitting for data from a MATLAB simulation function, see “Example: Prepare a Simulation Function for Use with BERTool” on page 15-62.

For greater flexibility in the process of fitting a curve to BER data, use the `berfit` function.

Use Simulation Functions with BERTool

- “Requirements for Functions” on page 15-59
- “Template for a Simulation Function” on page 15-60
- “Example: Prepare a Simulation Function for Use with BERTool” on page 15-62

Requirements for Functions

When you create a MATLAB function for use with BERTool, ensure the function interacts properly with the GUI. This section describes the inputs, outputs, and basic operation of a BERTool-compatible function.

Input Arguments

BERTool evaluates your entries in fields of the GUI and passes data to the function as these input arguments, in sequence:

- One value from the **Eb/No range** vector each time BERTool invokes the simulation function
- The **Number of errors** value
- The **Number of bits** value

Output Arguments

Your simulation function must compute and return these output arguments, in sequence:

- Bit error rate of the simulation
- Number of bits processed when computing the BER

BERTool uses these output arguments when reporting and plotting results.

Simulation Operation

Your simulation function must perform these tasks:

- Simulate the communication system for the E_b/N_0 value specified in the first input argument.
- Stop simulating when the number of errors or the number of processed bits equals or exceeds the corresponding threshold specified in the second or third input argument, respectively.
- Detect whether you click **Stop** in BERTool and abort the simulation in that case.

Template for a Simulation Function

Use the following template when adapting your code to work with BERTool. You can open it in an editor by entering `edit bertooltemplate` in the MATLAB Command Window. “Understanding the Template” on page 15-61 explains the template’s key sections, while “Using the Template” on page 15-62 indicates how to use the template with your own simulation code. Alternatively, you can develop your simulation function without using the template, but be sure it satisfies the requirements described in “Requirements for Functions” on page 15-59.

Note The template is not yet ready for use with BERTool. You must insert your own simulation code in the places marked **INSERT YOUR CODE HERE**. For a complete example based on this template, see “Example: Prepare a Simulation Function for Use with BERTool” on page 15-62.

```
function [ber, numBits] = bertooltemplate(EbNo, maxNumErrs, maxNumBits)
% Import Java class for BERTool.
import com.mathworks.toolbox.comm.BERTool;

% Initialize variables related to exit criteria.
berVec = zeros(3,1); % Updated BER values

% --- Set up parameters. ---
% --- INSERT YOUR CODE HERE.
% Simulate until number of errors exceeds maxNumErrs
% or number of bits processed exceeds maxNumBits.
while((berVec(2) < maxNumErrs) && (berVec(3) < maxNumBits))

    % Check if the user clicked the Stop button of BERTool.
    if (BERTool.getSimulationStop)
        break;
    end

    % --- Proceed with simulation.
    % --- Be sure to update totErr and numBits.
    % --- INSERT YOUR CODE HERE.
end % End of loop

% Assign values to the output variables.
ber = berVec(1);
numBits = berVec(3);
```

Understanding the Template

From studying the code in the function template, observe how the function either satisfies the requirements listed in “Requirements for Functions” on page 15-59 or indicates where your own insertions of code should do so. In particular,

- The function has appropriate input and output arguments.
- The function includes a placeholder for code that simulates a system for the given E_b/N_0 value.
- The function uses a loop structure to stop simulating when the number of errors exceeds `maxNumErrs` or the number of bits exceeds `maxNumBits`, whichever occurs first.

Note Although the `while` statement of the loop describes the exit criteria, your own code inserted into the section marked `Proceed with simulation` must compute

the number of errors and the number of bits. If you do not perform these computations in your own code, clicking **Stop** is the only way to terminate the loop.

- In each iteration of the loop, the function detects when the user clicks **Stop** in BERTool.

Using the Template

Here is a procedure for using the template with your own simulation code:

- 1 Determine the setup tasks you must perform. For example, you might want to initialize variables containing the modulation alphabet size, filter coefficients, a convolutional coding trellis, or the states of a convolutional interleaver. Place the code for these setup tasks in the template section marked `Set up parameters`.
- 2 Determine the core simulation tasks, assuming that all setup work has already been performed. For example, these tasks might include error-control coding, modulation/demodulation, and channel modeling. Place the code for these core simulation tasks in the template section marked `Proceed with simulation`.
- 3 Also in the template section marked `Proceed with simulation`, include code that updates the values of `totErr` and `numBits`. The quantity `totErr` represents the number of errors observed so far. The quantity `numBits` represents the number of bits processed so far. The computations to update these variables depend on how your core simulation tasks work.

Note Updating the numbers of errors and bits is important for ensuring that the loop terminates. However, if you accidentally create an infinite loop early in your development work using the function template, click **Stop** in BERTool to abort the simulation.

- 4 Omit any setup code that initializes `EbNo`, `maxNumErrs`, or `maxNumBits`, because BERTool passes these quantities to the function as input arguments after evaluating the data entered in the GUI.
- 5 Adjust your code or the template's code as necessary to use consistent variable names and meanings. For example, if your original code uses a variable called `ebn0` and the template's function declaration (first line) uses the variable name `EbNo`, you must change one of the names so they match. As another example, if your original code uses SNR instead of E_b/N_0 , you must convert quantities appropriately.

Example: Prepare a Simulation Function for Use with BERTool

This section adapts the function template given in “Template for a Simulation Function” on page 15-60.

Preparing the Function

To prepare the function for use with BERTool, follow these steps:

- 1 Copy the template from “Template for a Simulation Function” on page 15-60 into a new MATLAB file in the MATLAB Editor. Save it in a folder on your MATLAB path using the file name `bertool_simfcn`.
- 2 From the original example, the following lines are setup tasks. They are modified from the original example to rely on the input arguments that BERTool provides to the function, instead of defining variables such as `EbNovec` and `numerrmin` directly.

```
% Set up initial parameters.
siglen = 1000; % Number of bits in each trial
M = 2; % DBPSK is binary.
snr = EbNo; % Because of binary modulation
% ErrorRate calculator System object to compare decoded symbols to the
% original transmitted symbols.
errorCalc = comm.ErrorRate;
```

Place these lines of code in the template section marked `Set up parameters`.

- 3 From the original example, the following lines are the core simulation tasks, after all setup work has been performed.

```
msg = randi([0,M-1], siglen, 1); % Generate message sequence.
txsig = dpskmod(msg,M); % Modulate.
hChan.SignalPower = (txsig'*txsig)/length(txsig); % Calculate and
% assign signal power
rxsig = awgn(txsig,snr,'measured'); % Add noise.
decodmsg = dpskdemod(rxsig,M); % Demodulate.
berVec = errorCalc(msg,decodmsg); % Calculate BER
```

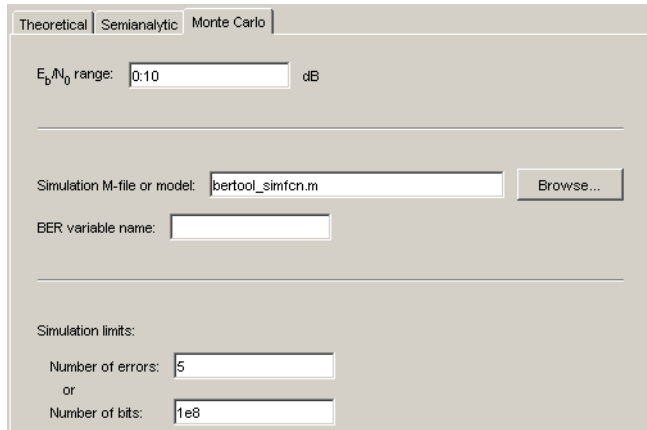
Place the code for these core simulation tasks in the template section marked `Proceed with simulation`.

The `bertool_simfcn` function is now compatible with BERTool. Note that unlike the original example, the function here does *not* initialize `EbNovec`, define `EbNo` as a scalar, or use `numerrmin` as the target number of errors; this is because BERTool provides input arguments for similar quantities. The `bertool_simfcn` function also excludes code related to plotting, curve fitting, and confidence intervals in the original example because BERTool enables you to do similar tasks interactively without writing code.

Using the Prepared Function

To use `bertool_simfcn` in conjunction with BERTool, continue the example by following these steps:

- 1 Open BERTool and go to the **Monte Carlo** tab.
- 2 Set parameters on the **Monte Carlo** tab as shown in the following figure.

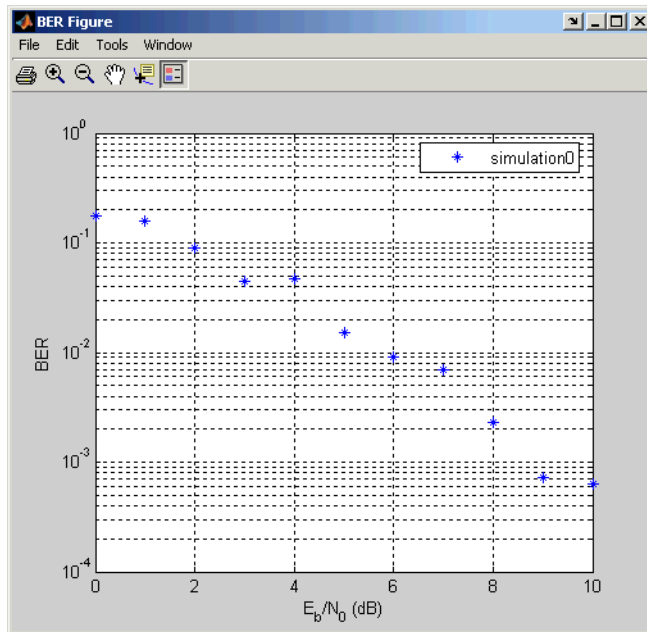


The image shows a screenshot of the BERTool software interface, specifically the Monte Carlo tab. The window has three tabs: 'Theoretical', 'Semianalytic', and 'Monte Carlo', with 'Monte Carlo' being the active tab. The interface contains several input fields and a button:

- E_b/N₀ range:** A text box containing '0:10' followed by 'dB'.
- Simulation M-file or model:** A text box containing 'bertool_simfcn.m' and a 'Browse...' button to its right.
- BER variable name:** An empty text box.
- Simulation limits:** A section with two options:
 - Number of errors:** A text box containing '5'.
 - or**
 - Number of bits:** A text box containing '1e8'.

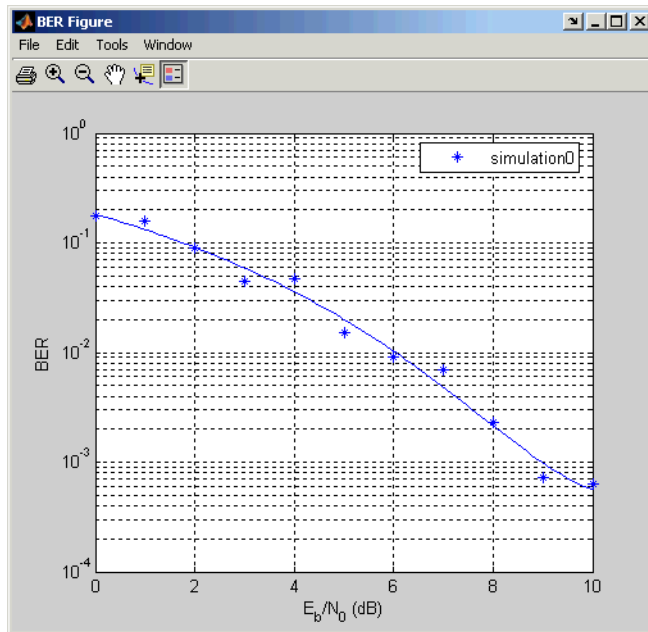
- 3 Click **Run**.

BERTool spends some time computing results and then plots them. They do not appear to fall along a smooth curve because the simulation required only five errors for each value in $E_b N_0$.



- 4 To fit a curve to the series of points in the BER Figure window, select the box next to **Fit** in the data viewer.

BERTool plots the curve, as shown in the following figure.



Run Simulink Simulations

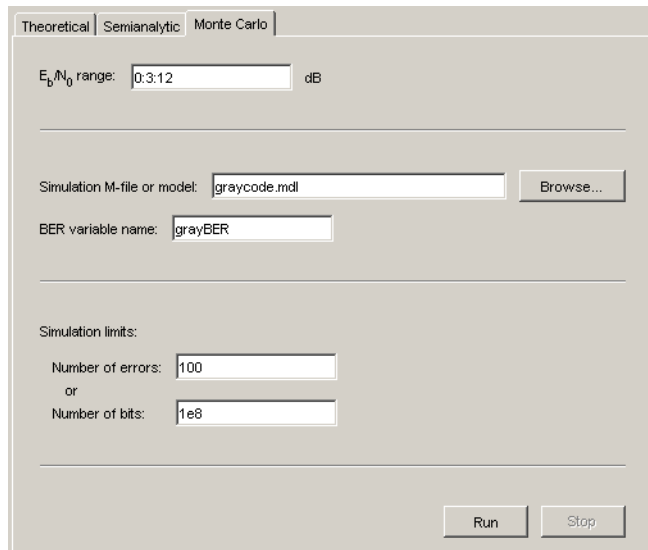
- “Section Overview” on page 15-66
- “Example: Using a Simulink Model with BERTool” on page 15-67
- “Varying the Stopping Criteria” on page 15-70

Section Overview

You can use BERTool in conjunction with Simulink models to generate and analyze BER data. The Simulink model simulates the communication system whose performance you want to study, while BERTool manages a series of simulations using the model and collects the BER data.

Note To use Simulink models within BERTool, you must have a Simulink license. Communications Toolbox software is highly recommended. The rest of this section assumes you have a license for both Simulink and Communications Toolbox applications.

To access the capabilities of BERTool related to Simulink models, open the **Monte Carlo** tab.



For further details about confidence intervals and curve fitting for simulation data, see “Plotting Confidence Intervals” on page 15-57 and “Fitting BER Points to a Curve” on page 15-58, respectively.

Example: Using a Simulink Model with BERTool

This example illustrates how BERTool can manage a series of simulations of a Simulink model, and how you can vary the plot. The model is `commgraycode`, one of the demonstration models included with Communications Toolbox software. The example assumes that you have Communications Toolbox software installed.

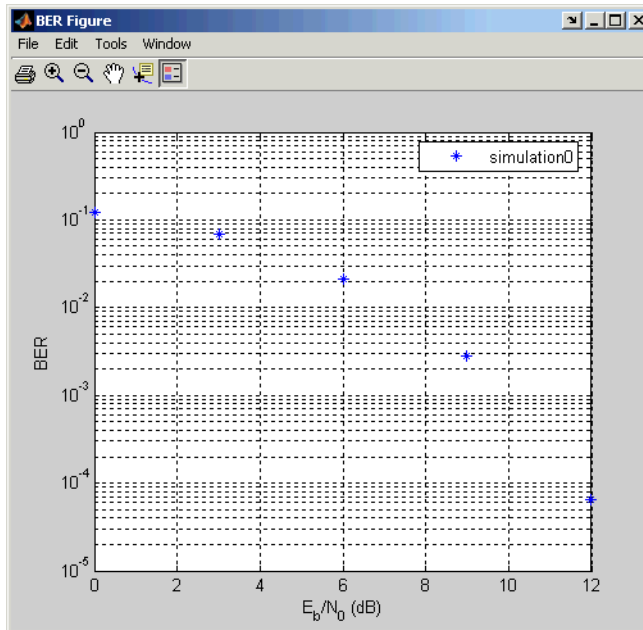
To run this example, follow these steps:

- 1 Open BERTool and go to the **Monte Carlo** tab. The model's file name, `commgraycode.mdl`, appears as the **Simulation MATLAB file or Simulink model** parameter. (If `viterbisim.m` appears there, select to indicate that Communications Toolbox software is installed.)
- 2 Click **Run**.

BERTool loads the model into memory (which in turn initializes several variables in the MATLAB workspace), runs the simulation once for each value of E_b/N_0 , and gathers BER data. BERTool creates a listing in the data viewer.

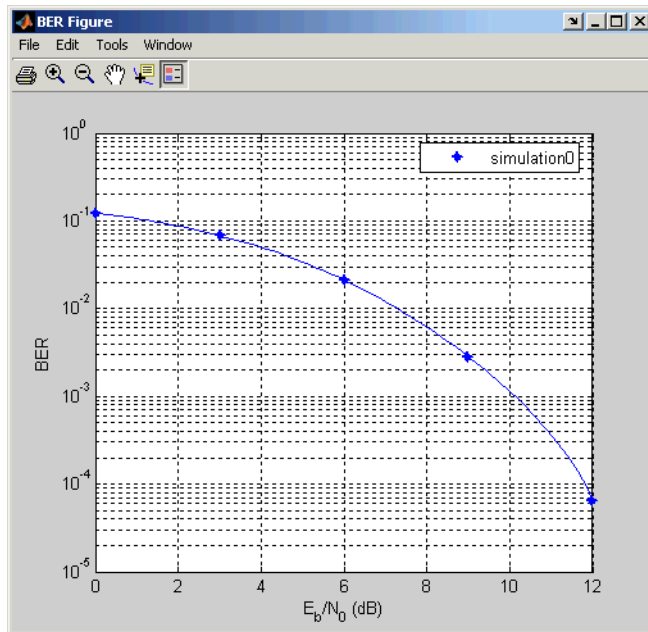
Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits
off	<input type="checkbox"/>	<input checked="" type="checkbox"/>	simulation0	0:3:12	[0.1233 0.0...	[900 1500 4...

BERTool plots the data in the BER Figure window.



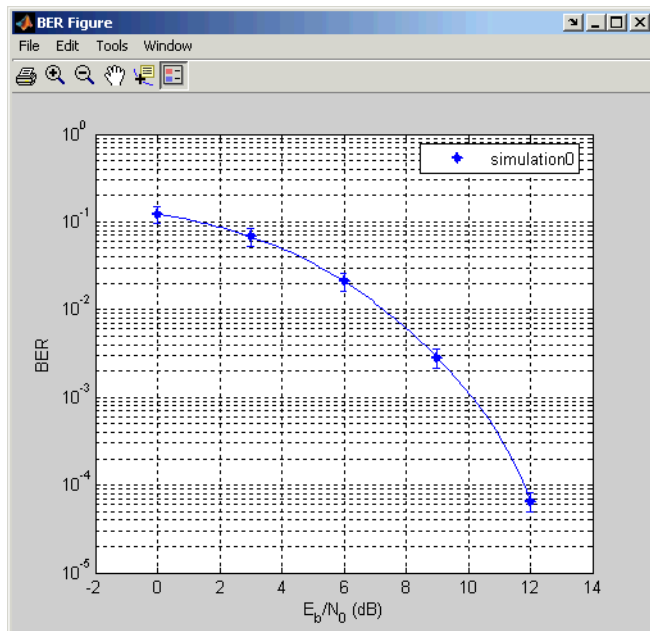
- 3 To fit a curve to the series of points in the BER Figure window, select the box next to **Fit** in the data viewer.

BERTool plots the curve, as below.



- 4 To indicate the 99% confidence interval around each point in the simulation data, set **Confidence Level** to 99% in the data viewer.

BERTool displays error bars to represent the confidence intervals, as below.



Another example that uses BERTool to manage a series of Simulink simulations is in “Example: Prepare a Model for Use with BERTool” on page 15-74.

Varying the Stopping Criteria

When you create a Simulink model for use with BERTool, you must set it up so that the simulation ends when it either detects a target number of errors or processes a maximum number of bits, whichever occurs first. To learn more about this requirement, see “Requirements for Models” on page 15-71; for an example, see “Example: Prepare a Model for Use with BERTool” on page 15-74.

After creating your Simulink model, set the target number of errors and the maximum number of bits in the **Monte Carlo** tab of BERTool.

Simulation limits:

Number of errors:

or

Number of bits:

Typically, a **Number of errors** value of at least 100 produces an accurate error rate. The **Number of bits** value prevents the simulation from running too long, especially at large

values of E_b/N_0 . However, if the **Number of bits** value is so small that the simulation collects very few errors, the error rate might not be accurate. You can use confidence intervals to gauge the accuracy of the error rates that your simulation produces; the larger the confidence interval, the less accurate the computed error rate.

You can also click **Stop** in BERTool to stop a series of simulations prematurely.

Use Simulink Models with BERTool

- “Requirements for Models” on page 15-71
- “Tips for Preparing Models” on page 15-71
- “Example: Prepare a Model for Use with BERTool” on page 15-74

Requirements for Models

A Simulink model must satisfy these requirements before you can use it with BERTool, where the case-sensitive variable names must be exactly as shown below:

- The channel block must use the variable `EbNo` rather than a hard-coded value for E_b/N_0 .
- The simulation must stop when the error count reaches the value of the variable `maxNumErrs` or when the number of processed bits reaches the value of the variable `maxNumBits`, whichever occurs first.

You can configure the Error Rate Calculation block in Communications Toolbox software to stop the simulation based on such criteria.

- The simulation must send the final error rate data to the MATLAB workspace as a variable whose name you enter in the **BER variable name** field in BERTool. The variable must be a three-element vector that lists the BER, the number of bit errors, and the number of processed bits.

This three-element vector format is supported by the Error Rate Calculation block.

Tips for Preparing Models

Here are some tips for preparing a Simulink model for use with BERTool:

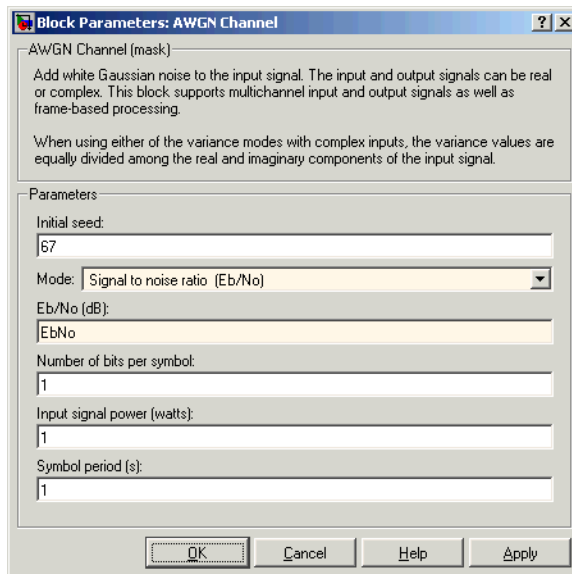
- To avoid using an undefined variable name in the dialog box for a Simulink block in the steps that follow, set up variables in the MATLAB workspace using a command such as the one below.

```
EbNo = 0; maxNumErrs = 100; maxNumBits = 1e8;
```

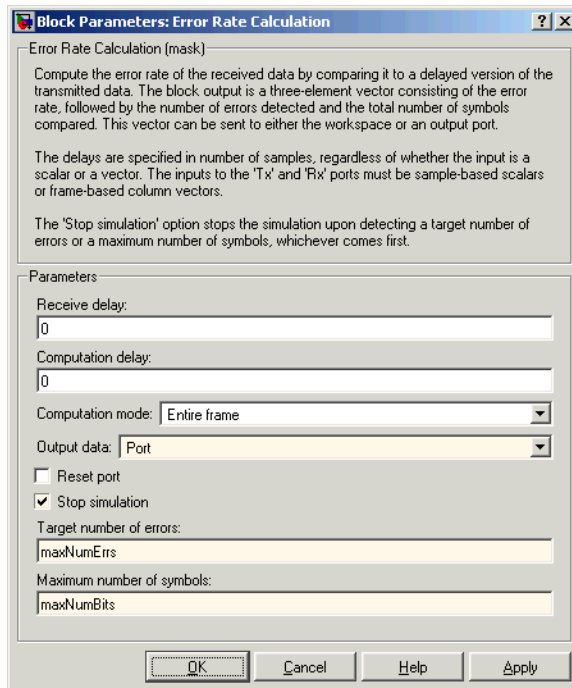
You might also want to put the same command in the model's preload function callback, to initialize the variables if you reopen the model in a future MATLAB session.

When you use BERTool, it provides the actual values based on what you enter in the GUI, so the initial values above are somewhat arbitrary.

- To model the channel, use the AWGN Channel block in Communications Toolbox software with these parameters:
 - **Mode** = Signal to noise ratio (Eb/No)
 - **Eb/No** = EbNo



- To compute the error rate, use the Error Rate Calculation block in Communications Toolbox software with these parameters:
 - Check **Stop simulation**.
 - **Target number of errors** = maxNumErrs
 - **Maximum number of symbols** = maxNumBits



- To send data from the Error Rate Calculation block to the MATLAB workspace, set **Output data** to **Port**, attach a To Workspace block, and set the latter block's **Limit data points to last** parameter to 1. The **Variable name** parameter in the To Workspace block must match the value you enter in the **BER variable name** field of BERTool.

Tip More than one To Workspace block is available. Be sure to select To Workspace from the DSP System Toolbox / Sinks sublibrary.

- If your model computes a symbol error rate instead of a bit error rate, use the Integer to Bit Converter block in Communications Toolbox software to convert symbols to bits.
- Frame-based simulations often run faster than sample-based simulations for the same number of bits processed. The number of errors or number of processed bits might exceed the values you enter in BERTool, because the simulation always processes a fixed amount of data in each frame.
- If you have an existing model that uses the AWGN Channel block using a **Mode** parameter other than **Signal to noise ratio (Eb/No)**, you can adapt the block

to use the Eb/No mode instead. To learn about how the block's different modes are related to each other, press the AWGN Channel block's **Help** button to view the online reference page.

- If your model uses a preload function or other callback to initialize variables in the MATLAB workspace upon loading, make sure before you use the **Run** button in BERTool that one of these conditions is met:
 - The model is not currently in memory. In this case, BERTool loads the model into memory and runs the callback functions.
 - The model is in memory (whether in a window or not), and the variables are intact.

If you clear or overwrite the model's variables and want to restore their values before using the **Run** button in BERTool, you can use the `bdclose` function in the MATLAB Command Window to clear the model from memory. This causes BERTool to reload the model after you click **Run**. Similarly, if you refresh your workspace by issuing a `clear all` or `clear variables` command, you should also clear the model from memory by using `bdclose all`.

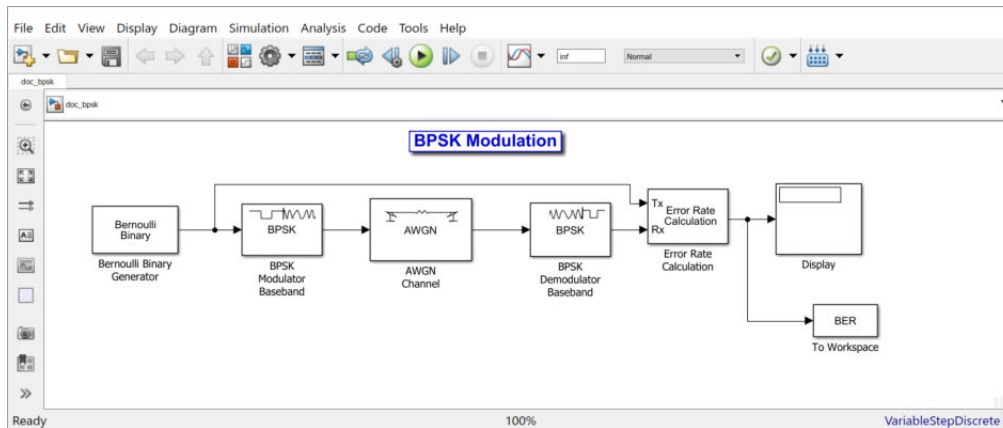
Example: Prepare a Model for Use with BERTool

This example uses a Simulink model that is set up for use with BERTool. The example also illustrates how to compare the BER performance of a Simulink simulation with theoretical BER results. The example assumes that you have Communications Toolbox software installed.

To prepare the model for use with BERTool, follow these steps, using the exact case-sensitive variable names as shown:

- 1** Open the model by entering the following command in the MATLAB Command Window.

```
doc_bpsk
```



- 2 To initialize parameters in the MATLAB workspace and avoid using undefined variables as block parameters, enter the following command in the MATLAB Command Window.

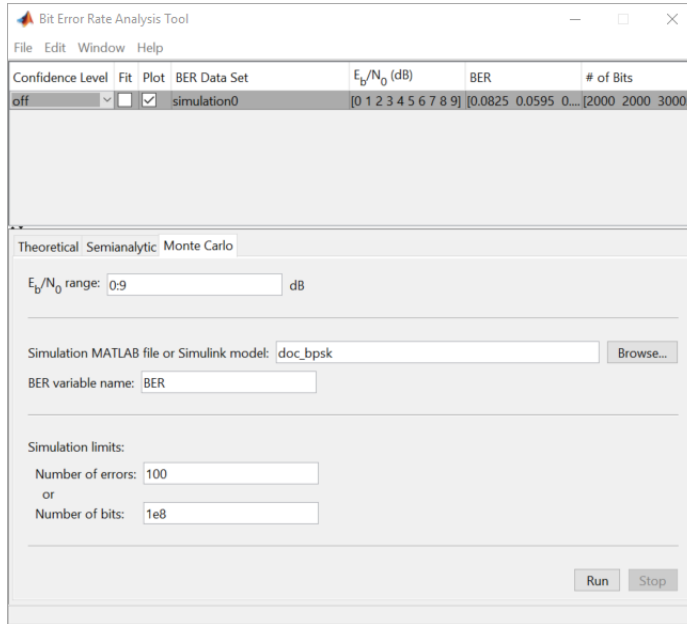

```
EbNo = 0; maxNumErrs = 100; maxNumBits = 1e8;
```
- 3 To ensure that BERTool uses the correct amount of noise each time it runs the simulation, open the dialog box for the AWGN Channel block by double-clicking the block. Verify that **Es/No** is set to **EbNo** and click **OK**. In this particular model, E_s/N_0 is equivalent to E_b/N_0 because the modulation type is BPSK.
- 4 To ensure that BERTool uses the correct stopping criteria for each iteration,
 - Open the dialog box for the Error Rate Calculation block. Verify that **Target number of errors** is set to **maxNumErrs**, and that **Maximum number of symbols** is set to **maxNumBits**. Click **OK**.
 - The simulation stop time must be set to **Inf**.
- 5 To enable BERTool to access the BER results that the Error Rate Calculation block computes, the To Workspace block, BER, is connected to the output of the Error Rate Calculation block.

Tip More than one To Workspace block is available. Be sure to select To Workspace from the DSP System Toolbox / Sinks sublibrary.

To use the doc_bpsk model with BERTool, follow these steps:

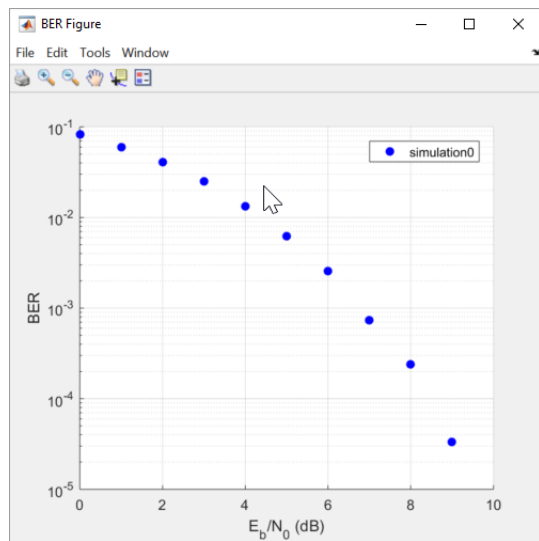
- 1 Open BERTool and go to the **Monte Carlo** tab.

- 2 Set parameters on the **Monte Carlo** tab as shown in the following figure.

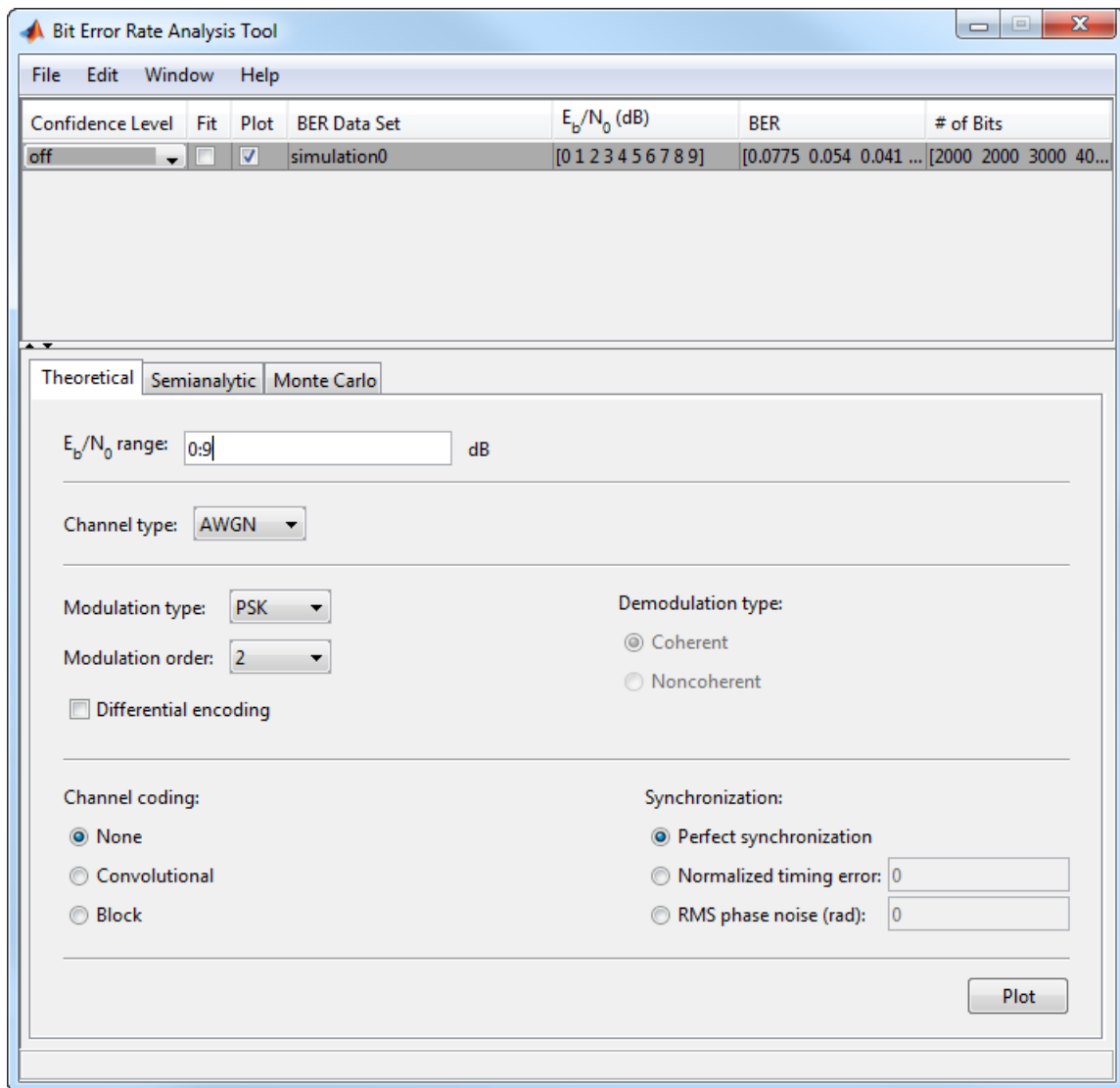


- 3 Click **Run**.

BERTool spends some time computing results and then plots them.

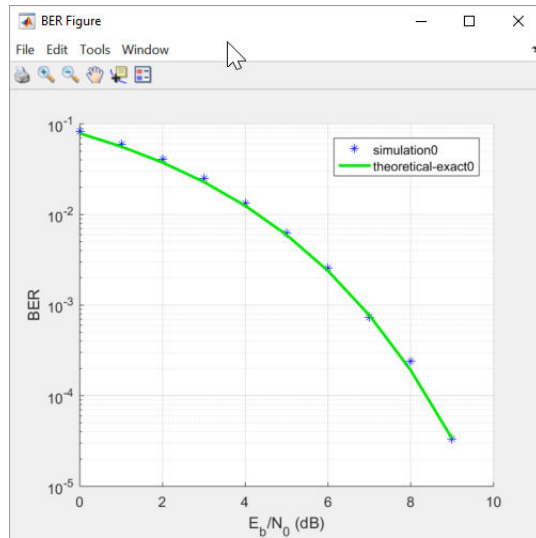


- 4 To compare these simulation results with theoretical results, go to the **Theoretical** tab in BERTool and set parameters as shown below.



5 Click **Plot**.

BERTool plots the theoretical curve in the BER Figure window along with the earlier simulation results.



Manage BER Data

- “Exporting Data Sets or BERTool Sessions” on page 15-79
- “Importing Data Sets or BERTool Sessions” on page 15-82
- “Managing Data in the Data Viewer” on page 15-83

Exporting Data Sets or BERTool Sessions

BERTool enables you to export individual data sets to the MATLAB workspace or to MAT-files. One option for exporting is convenient for processing the data outside BERTool. For example, to create a highly customized plot using data from BERTool, export the BERTool data set to the MATLAB workspace and use any of the plotting commands in MATLAB. Another option for exporting enables you to reimport the data into BERTool later.

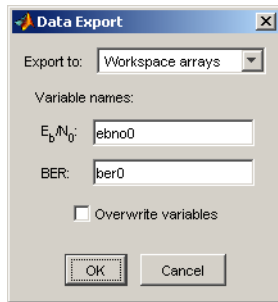
BERTool also enables you to save an entire session, which is useful if your session contains multiple data sets that you want to return to in a later session.

This section describes these capabilities:

Exporting Data Sets

To export an individual data set, follow these steps:

- 1 In the data viewer, select the data set you want to export.
- 2 Choose **File > Export Data**.



- 3 Set **Export to** to indicate the format and destination of the data.
 - a If you want to reimport the data into BERTool later, you *must* choose either **Workspace structure** or **MAT-file structure** to create a structure in the MATLAB workspace or a MAT-file, respectively.

A new field called **Structure name** appears. Set it to the name that you want BERTool to use for the structure it creates.

If you selected **Workspace structure** and you want BERTool to use your chosen variable name, even if a variable by that name already exists in the workspace, select **Overwrite variables**.

- b If you do *not* need to reimport the data into BERTool later, a convenient way to access the data outside BERTool is to have BERTool create a pair of arrays in the MATLAB workspace. One array contains E_b/N_0 values, while the other array contains BER values. To choose this option, set **Export to** to **Workspace arrays**.

Then type two variable names in the fields under **Variable names**.

If you want BERTool to use your chosen variable names even if variables by those names already exist in the workspace, select **Overwrite variables**.

- 4 Click **OK**. If you selected **MAT-file structure**, BERTool prompts you for the path to the MAT-file that you want to create.

To reimport a structure later, see “Importing Data Sets” on page 15-82.

Examining an Exported Structure

This section briefly describes the contents of the structure that BERTool exports to the workspace or to a MAT-file. The structure's fields are indicated in the table below. The fields that are most relevant for you when you want to manipulate exported data are `paramsEvaled` and `data`.

Name of Field	Significance
params	The parameter values in the BERTool GUI, some of which might be invisible and hence irrelevant for computations.
paramsEvaled	The parameter values that BERTool uses when computing the data set.
data	The E_b/N_0 , BER, and number of bits processed.
dataView	Information about the appearance in the data viewer. Used by BERTool for data reimport.
cellEditabilities	Indicates whether the data viewer has an active Confidence Level or Fit entry. Used by BERTool for data reimport.

Parameter Fields

The `params` and `paramsEvaled` fields are similar to each other, except that `params` describes the exact state of the GUI whereas `paramsEvaled` indicates the values that are actually used for computations. As an example of the difference, for a theoretical system with an AWGN channel, `params` records but `paramsEvaled` omits a diversity order parameter. The diversity order is not used in the computations because it is relevant only for systems with Rayleigh channels. As another example, if you type `[0:3]+1` in the GUI as the range of E_b/N_0 values, `params` indicates `[0:3]+1` while `paramsEvaled` indicates `1 2 3 4`.

The length and exact contents of `paramsEvaled` depend on the data set because only relevant information appears. If the meaning of the contents of `paramsEvaled` is not clear upon inspection, one way to learn more is to reimport the data set into BERTool and inspect the parameter values that appear in the GUI. To reimport the structure, follow the instructions in "Importing Data Sets or BERTool Sessions" on page 15-82.

Data Field

If your exported workspace variable is called `ber0`, the field `ber0.data` is a cell array that contains the numerical results in these vectors:

- `ber0.data{1}` lists the E_b/N_0 values.
- `ber0.data{2}` lists the BER values corresponding to each of the E_b/N_0 values.
- `ber0.data{3}` indicates, for simulation or semianalytic results, how many bits BERTool processed when computing each of the corresponding BER values.

Saving a BERTool Session

To save an entire BERTool session, follow these steps:

- 1 Choose **File > Save Session**.
- 2 When BERTool prompts you, enter the path to the file that you want to create.

BERTool creates a text file that records all data sets currently in the data viewer, along with the GUI parameters associated with the data sets.

Note If your BERTool session requires particular workspace variables (such as `txsig` or `rxsig` for the **Semianalytic** tab), save those separately in a MAT-file using the `save` command in MATLAB.

Importing Data Sets or BERTool Sessions

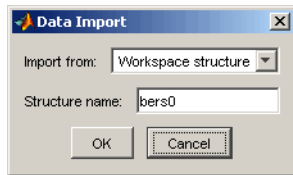
BERTool enables you to reimport individual data sets that you previously exported to a structure, or to reload entire sessions that you previously saved. This section describes these capabilities:

To learn more about exporting data sets or saving sessions from BERTool, see “Exporting Data Sets or BERTool Sessions” on page 15-79.

Importing Data Sets

To import an individual data set that you previously exported from BERTool to a structure, follow these steps:

- 1 Choose **File > Import Data**.



- 2 Set **Import from** to either `Workspace structure` or `MAT-file structure`. If you select `Workspace structure`, type the name of the workspace variable in the **Structure name** field.
- 3 Click **OK**. If you select `MAT-file`, BERTool prompts you to select the file that contains the structure you want to import.

After you dismiss the **Data Import** dialog box (and the file selection dialog box, in the case of a MAT-file), the data viewer shows the newly imported data set and the BER Figure window plots it.

Opening a Previous BERTool Session

To replace the data sets in the data viewer with data sets from a previous BERTool session, follow these steps:

- 1 Choose **File > Open Session**.

Note If BERTool already contains data sets, it asks you whether you want to save the current session. If you answer no and continue with the loading process, BERTool discards the current session upon opening the new session from the file.

- 2 When BERTool prompts you, enter the path to the file you want to open. It must be a file that you previously created using the **Save Session** option in BERTool.

After BERTool reads the session file, the data viewer shows the data sets from the file.

If your BERTool session requires particular workspace variables (such as `txsig` or `rxsig` for the **Semianalytic** tab) that you saved separately in a MAT-file, you can retrieve them using the `load` command in MATLAB.

Managing Data in the Data Viewer

The data viewer gives you flexibility to rename and delete data sets, and to reorder columns in the data viewer.

- To rename a data set in the data viewer, double-click its name in the **BER Data Set** column and type a new name.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0 (dB)	BER	# of Bits
	<input type="checkbox"/>	<input checked="" type="checkbox"/>	theoretical0	[0.0 1.0 2.0 3....]	[0.0755 0.0546 ...]	
off	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	simulation0	[0.0 3.0 6.0]	[0.12 0.06 0.02]	[300 300 600]

- To delete a data set from the data viewer, select it and choose **Edit > Delete**.

Note If the data set originated from the **Semianalytic** or **Theoretical** tab, BERTool deletes the data without asking for confirmation. You cannot undo this operation.

- To move a column in the data viewer, drag the column's heading to the left or right with the mouse. For example, the image below shows the mouse dragging the **BER** column to the left of its default position. When you release the mouse button, the columns snap into place.

Confidence Level	Fit	Plot	BER Data Set	E_b/N_0	BER	# of Bits
	<input type="checkbox"/>	<input checked="" type="checkbox"/>	theoretical0	[0.0 1.0	[0.0755 0.0546 ...]	
off	<input type="checkbox"/>	<input checked="" type="checkbox"/>	simulation0	[0.0 3.0	[0.12 0.06 0.02]	[300 300 600]

Error Vector Magnitude (EVM)

Error Vector Magnitude (EVM) is a measurement of modulator or demodulator performance in the presence of impairments. Essentially, EVM is the vector difference at a given time between the ideal (transmitted) signal and the measured (received) signal. If used correctly, these measurements can help in identifying sources of signal degradation, such as: phase noise, I-Q imbalance, amplitude non-linearity and filter distortion

These types of measurements are useful for determining system performance in communications applications. For example, determining if an EDGE system conforms to the 3GPP radio transmission standards requires accurate RMS, EVM, Peak EVM, and 95th percentile for the EVM measurements.

Users can create the EVM object in two ways: using a default object or by defining parameter-value pairs. As defined by the 3GPP standard, the unit of measure for RMS, Maximum, and Percentile EVM measurements is a percentile (%). For more information, see the EVM Measurement or `comm.EVM` help page.

Measuring Modulator Accuracy

- “Overview” on page 15-85
- “Structure” on page 15-87
- “References” on page 15-89

Overview

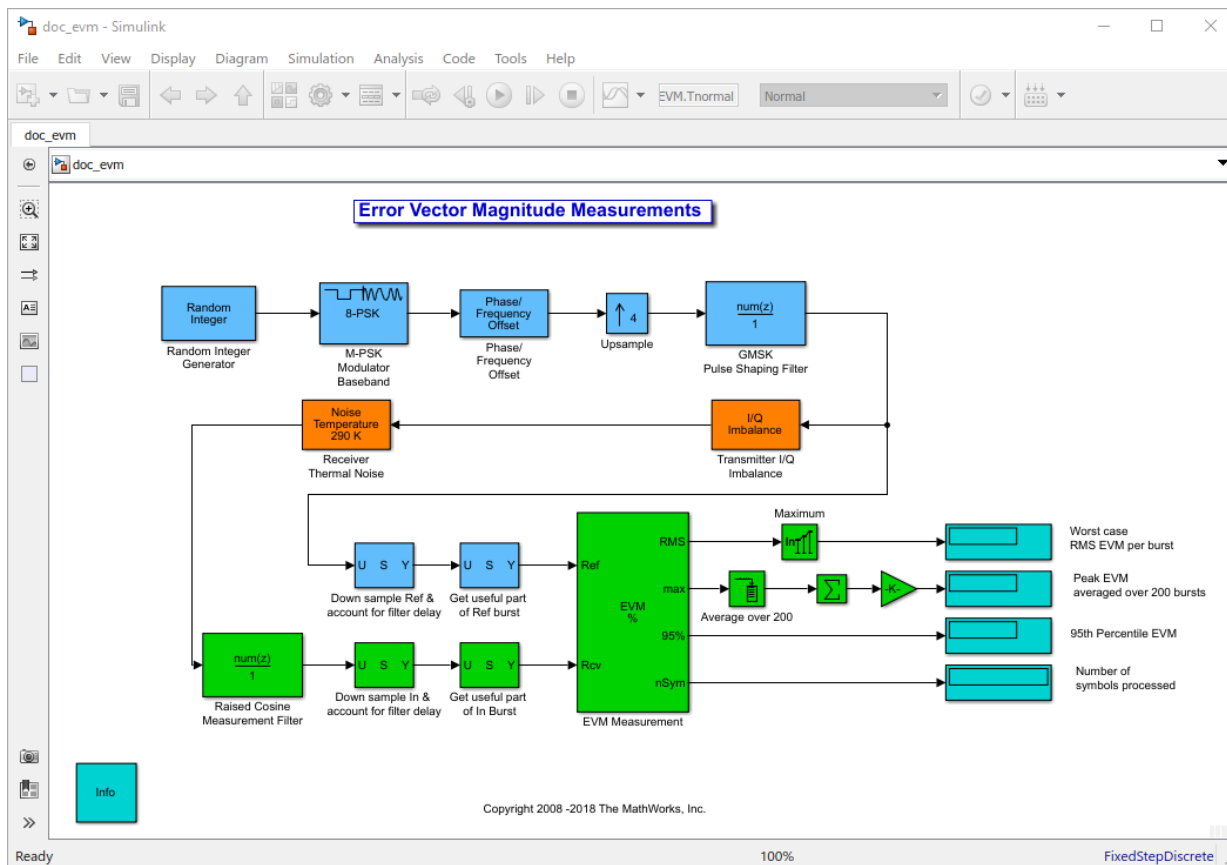
The Communications Toolbox provides two blocks you can use for measuring modulator accuracy: EVM Measurement and MER Measurement.

This example tests an EDGE transmitter for system design impairments using EVM measurements. In this example, the EVM Measurements block compares an ideal reference signal to a measured signal, and then computes RMS EVM, maximum EVM, and percentile EVM values. According to the EDGE standard [1], the error vector magnitude of the received signal, calculated relative to the transmitted waveform, should not exceed the following values:

EDGE Standard Measurement Specifications [2]

Measurement	Mobile Station		Base Transceiver Station	
	Normal	Extreme	Normal	Extreme
RMS	9%	10%	7%	8%
Peak EVM	30%	30%	22%	22%
95th Percentile EVM	15%	15%	11%	11%

This example uses the following model.



You can open this model by typing `doc_evm` at the MATLAB command line.

Structure

The model essentially contains three parts:

- Transmitter
- Receiver impairments
- EVM calculation

The following sections of the tutorial contain descriptions for each part of the model.

Transmitter

The following blocks comprise the transmitter:

- Random Integer Generator
- M-PSK Modulator Baseband
- Phase/Frequency Offset
- Upsample
- Discrete FIR Filter
- I/Q Imbalance

The Random Integer Generator block simulates random data generation. The EDGE standard specifies that the transmitter performs measurements during the useful part of the burst - excluding tail bits - over at least 200 bursts. In this mode, the transmitter produces 435 symbols per burst (9 additional symbols account for filter delays). The Phase Offset block provides continuous $3\pi/8$ phase rotation to the signal. For synchronization purposes, the Upsample block oversamples the signal by a factor of 4.

The Discrete FIR Filter block provides a GMSK pulse linearization, the main component in a Laurent decomposition of the GMSK modulation [3]. A helper function computes the filter coefficients and uses a direct-form FIR digital filter to create the pulse shaping effect. The filter normalization provides unity gain at the main tap.

The I/Q Imbalance block simulates transmitter impairments. This block adds rotation to the signal, simulating a defect in the transmitter under test. The **I/Q amplitude imbalance** is 0.5 dB, and **I/Q phase imbalance** is 1° .

Receiver Impairments

In this model, the Receiver Thermal Noise block represents receiver impairments. This model assumes 290 K of thermal noise, representing imperfections of the hardware under test.

EVM Calculation

The EVM calculation relies upon the following blocks:

- Discrete FIR Filter
- Selector
- EVM Measurement
- Display

The EVM measurement block computes the vector difference between an ideal reference signal and an impaired signal. The output of the FIR filter provides the Reference input for the EVM block. The output of the Noise Temperature block provides the impaired signal at the Input port of the EVM block.

While the block has different normalization options available, the EDGE standard requires normalizing by the Average reference signal power. For illustration purposes in this example, the EVM block outputs RMS, maximum, and percentile measurement values.

Experimenting with the Model

- 1** Run the model by clicking the play button in the Simulink model window.
- 2** Examine the output of the EVM block and compare the measurements to the limits in the EDGE Standard Measurement Specifications table.

In this example, the EVM Measurement block computes the following:

- Worst case RMS EVM per burst: 9.77%
- Peak EVM: 18.95%
- 95th Percentile EVM:14.76%

As a result, this simulated EDGE transmitter passes the EVM test for a Mobile Station under extreme conditions.

- 3** Double-click the I/Q Imbalance block.

- 4 Enter 2 into **I/Q Imbalance (dB)** and click **OK**.
- 5 Click the Play button in the Simulink model window.
- 6 Examine the output of the EVM block. Then, compare the measurements to the limits in the EDGE Standard Measurement Specifications table.

In this example, the EVM Measurement block computes the following results:

- Worst case RMS EVM per burst: 15.15%
- Peak EVM: 29.73%
- 95th Percentile EVM: 22.55%.

These EVM values are clearly unacceptable according to the EDGE standard. You can experiment with the other I/Q imbalance values, examine the impact on calculations, and compare them to the values provided in the table.

References

- [1] 3GPP TS 45.004, "Radio Access Networks; Modulation," Release 7, v7.2.0, 2008-02.
- [2] 3GPP TS 45.005, "Radio Access Network; Radio transmission and reception," Release 8, v8.1.0, 2008-05.
- [3] Laurent, Pierre. "Exact and approximate construction of digital phase modulation by superposition of amplitude modulated pulses (AMP)." *IEEE Transactions on Communications*. Vol. COM-34, #2, Feb. 1986, pp. 150-160.

Modulation Error Ratio (MER)

Communications Toolbox can perform Modulation Error Ratio (MER) measurements. MER is a measure of the signal-to-noise ratio (SNR) in a digital modulation applications. These types of measurements are useful for determining system performance in communications applications. For example, determining if an EDGE system conforms to the 3GPP radio transmission standards requires accurate RMS, EVM, Peak EVM, and 95th percentile for the EVM measurements.

As defined by the DVB standard, the unit of measure for MER is decibels (dB). For consistency, the unit of measure for Minimum MER and Percentile MER measurements is also in decibels. For more information, see the `comm.MER` help page.

Adjacent Channel Power Ratio (ACPR)

Adjacent channel power ratio (ACPR) calculations (also known as adjacent channel leakage ratio (ACLR)), characterize spectral regrowth in a communications system component, such as a modulator or an analog front end. Amplifier nonlinearity causes spectral regrowth. ACPR calculations determine the likelihood that a given system causes interference with an adjacent channel.

Many transmission standards, such as IS-95, CDMA, WCDMA, 802.11, and Bluetooth, contain a definition for ACPR measurements. Most standards define ACPR measurements as the ratio of the average power in the main channel and any adjacent channels. The offset frequencies and measurement bandwidths (BWs) you use when obtaining measurements depends on which specific industry standard you are using. For instance, measurements for CDMA amplifiers involve two offsets (from the carrier frequency) of 885 kHz and 1.98 MHz, and a measurement BW of 30 KHz.

For more information, see the `comm.ACPR` help page.

Obtain ACPR Measurements

Communications Toolbox contains the `comm.ACPR` System object. In this tutorial, you obtain ACPR measurements using a WCDMA communications signal, according to the 3GPP™ TS 125.104 standard.

This example uses baseband WCDMA sample signals at the input and output of a nonlinear amplifier. The `WCDMASignal.mat` file contains sample data for use with the tutorial. This file divides the data into 25 signal snapshots of 7e3 samples each and stores them in the columns of data matrices, `dataBeforeAmplifier` and `dataAfterAmplifier`.

The WCDMA specification requires that you obtain all measurements using a 3.84 MHz sampling frequency.

Create `comm.ACPR` System Object and Set Up Measurements

- 1 Define the sample rate, load the WCDMA file, and get the data by entering the following at the MATLAB command line:

```
% System sampling frequency, 3.84 MHz chip rate, 8 samples per chip
SampleRate = 3.84e6*8;
load WCDMASignal.mat
```

```
% Use the first signal snapshot
txSignalBeforeAmplifier = dataBeforeAmplifier(:,1);
txSignalAfterAmplifier = dataAfterAmplifier(:,1);
```

- 2 Create the `comm.ACPR` System object and specify the sampling frequency.

```
hACPR = comm.ACPR('SampleRate', SampleRate)
```

The System object presents the following information:

```
NormalizedFrequency: false
SampleRate: 30720000
MainChannelFrequency: 0
MainMeasurementBandwidth: 50000
AdjacentChannelOffset: [-100000 100000]
AdjacentMeasurementBandwidth: 50000
MeasurementFilterSource: 'None'
SpectralEstimation: 'Auto'
FFTLength: 'Next power of 2'
MaxHold: false
PowerUnits: 'dBm'
MainChannelPowerOutputPort: false
AdjacentChannelPowerOutputPort: false
```

- 3 Specify the *main channel* center frequency and measurement bandwidth.

Specify the main channel center frequency using the `MainChannelFrequency` property. Then, specify the main channel measurement bandwidth using the `MainMeasurementBandwidth` property.

For the baseband data you are using, the main channel center frequency is at 0 Hz. The WCDMA standard specifies that you obtain main channel power using a 3.84-MHz measurement bandwidth. Specify these by typing the following.

```
hACPR.MainChannelFrequency = 0;
hACPR.MainMeasurementBandwidth = 3.84e6;
```

- 4 Specify *adjacent channel* offsets and measurement bandwidths.

The WCDMA standard specifies ACPR limits for four adjacent channels, located at 5, -5, 10, -10 MHz away from the main channel center frequency. In all cases, you obtain adjacent channel power using a 3.84-MHz bandwidth. Specify the adjacent channel offsets and measurement bandwidths using the `AdjacentChannelOffset` and `AdjacentMeasurementBandwidth` properties.


```
hACPR.AdjacentChannelOffset = [-10 -5 5 10]*1e6;
hACPR.AdjacentMeasurementBandwidth = 3.84e6;
```

Notice that if the measurement bandwidths for all the adjacent channels are equal, you specify a scalar value. If measurement bandwidths are different, you specify a vector of measurement bandwidths with a length equal to the length of the offset vector.

- 5 Set the `MainChannelPowerOutputPort` and `AdjacentChannelPowerOutputPort` properties to `true` by entering the following at the MATLAB command line:

```
hACPR.MainChannelPowerOutputPort = true
hACPR.AdjacentChannelPowerOutputPort = true
```

- 6 Create a `comm.ACPR System` object to measure the amplifier output.

```
hACPRoutput = clone(hACPR);
```

Obtain the ACPR Measurements

You obtain ACPR measurements by calling the `step` method of `comm.ACPR`. You can also obtain the power measurements for the main and adjacent channels. The `PowerUnits` property specifies the unit of measure. The property value defaults to `dBm` (power ratio referenced to one milliwatt (mW)).

- 1 Obtain the ACPR measurements at the amplifier input:

```
[ACPR mainChannelPower adjChannelPower] = ...,
step(hACPR,txSignalBeforeAmplifier)
```

The `comm.ACPR System` object produces the following output measurement data:

ACPR =

```
-68.6668 -54.9002 -55.0653 -68.4604
```

mainChannelPower =

```
29.5190
```

adjChannelPower =

```
-39.1477 -25.3812 -25.5463 -38.9414
```

- 2 Obtain the ACPR measurements at the amplifier output:

```
[ACPR mainChannelPower adjChannelPower] = ...,  
step(hACPRoutput,txSignalAfterAmplifier)
```

The `comm.ACPR System` object produces the following input measurement data:

ACPR =

```
-42.1625 -27.0912 -26.8785 -42.4915
```

mainChannelPower =

```
40.6725
```

adjChannelPower =

```
-1.4899 13.5813 13.7941 -1.8190
```

Notice the increase in ACPR values at the output of the amplifier. This increase reflects distortion due to amplifier nonlinearity. The WCDMA standard specifies that ACPR values be below -45 dB at +/- 5 MHz offsets, and below -50 dB at +/- 10 MHz offsets. In this example, the signal at the amplifier input meets the specifications while the signal at the amplifier output does not.

Specifying a Measurement Filter

The WCDMA standard specifies that you obtain ACPR measurements using a root-raised-cosine filter. It also states that you measure *both* the main channel power and adjacent channel powers using a matched root-raised-cosine (RRC) filter with rolloff factor 0.22. You specify the measurement filter using the `MeasurementFilter` property. This property value defaults to an all-pass filter with unity gain.

The filter must be an FIR filter, and its response must center at 0 Hz. The ACPR object automatically shifts and applies the filter at each of the specified main and adjacent channel bands. (The power measurement still falls within the bands specified by the `MainMeasurementBandwidth`, and `AdjacentMeasurementBandwidth` properties.)

The `WCDMASignal.mat` file contains data that was obtained using a 96 tap filter with a rolloff factor of 0.22.

- 1 Create the filter (using `rcosdesign`, from the Signal Processing Toolbox software) and obtain measurements by entering the following at the MATLAB command line:

```
% Scale for 0 dB passband gain
measFilt = rcosdesign(0.22,16,8)/sqrt(8);
```

- 2 Set the filter you created in the previous step as the measurement filter for the ACPR object.

```
release(hACPR);
hACPR.MeasurementFilterSource = 'Property';
hACPR.MeasurementFilter = measFilt;
```

- 3 Implement the same filter at the amplifier output by cloning the comm.ACPR System object.

```
hACPRoutput = clone(hACPR)
```

- 4 Obtain the ACPR power measurements at the amplifier input.

```
ACPR = step(hACPR,txSignalBeforeAmplifier)
```

The comm.ACPR System object produces the following measurement data:

```
ACPR =
   -71.4648   -55.5514   -55.9476   -71.3909
```

- 5 Obtain the ACPR power measurements at the amplifier output.

```
ACPRoutput = step(hACPRoutput,txSignalAfterAmplifier)
```

The comm.ACPR System object produces the following measurement data:

```
ACPR =
   -42.2364   -27.2242   -27.0748   -42.5810
```

Control the Power Spectral Estimator

By default, the ACPR object measures power uses a Welch power spectral estimator with a Hamming window and zero percent overlap. The object uses a rectangle approximation of the integral for the power spectral density estimates in the measurement bandwidth of interest. If you set SpectralEstimatorOption to 'User defined' several properties become available, providing you control of the resolution, variance, and dynamic range of the spectral estimates.

- 1 Enable the SegmentLength, OverlapPercentage, and WindowOption properties by entering the following at the MATLAB command line:

```
release(hACPRoutput)
hACPRoutput.SpectralEstimation = 'Specify window parameters'
```

This change allows you to customize the spectral estimates for obtaining power measurements. For example, you can set the spectral estimator segment length to 1024 and increase the overlap percentage to 50%, reducing the consequent variance increase. You can also choose a window with larger side lobe attenuation (compared to the default Hamming window).

- 2 Create a spectral estimator with a 'Chebyshev' window and a side lobe attenuation of 200 dB.

```
hACPRoutput.SegmentLength = 1024;
hACPRoutput.OverlapPercentage = 50;
% Choosing a Chebyshev window enables a SidelobeAtten property
% you can use to set the side lobe attenuation of the window.
hACPRoutput.Window = 'Chebyshev';
hACPRoutput.SidelobeAttenuation = 200;
```

- 3 Call the step method to obtain the ACPR power measurements at the amplifier output.

```
ACPRoutput = step(hACPRoutput,txSignalAfterAmplifier)
```

The ACPR object produces the following measurement data:

```
ACPR =
   -44.9399   -30.7136   -30.7670   -44.4450
```

Measure Power Using the Max-Hold Option.

Some communications standards specify using max-hold spectrum power measurements when computing ACPR values. Such calculations compare the current power spectral density vector estimation to the previous max-hold accumulated power spectral density vector estimation. When obtaining max-hold measurements, the object obtains the power spectral density vector estimation using the current input data. It obtains the previous max-hold accumulated power spectral density vector from the previous call to the step method. The object uses the maximum values at each frequency bin for calculating average power measurements. A call to the reset method clears the max-hold spectrum.

- 1 Accumulate max-hold spectra for 25 amplifier output data snapshots and get ACPR measurements by typing the following at the MATLAB command line:

```
for idx = 1:24
    step(hACPRoutput,dataAfterAmplifier(:,idx));
end
ACPRoutput = step(hACPRoutput,dataAfterAmplifier(:,25))
```

The ACPR object produces the following output data:

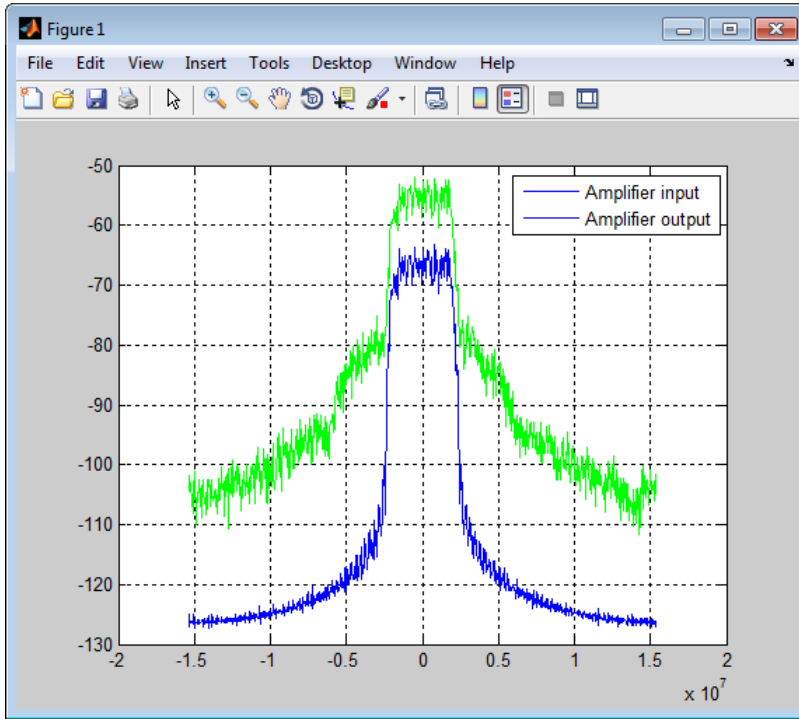
ACPR =

-43.1123 -26.6964 -27.0009 -42.4803

Plotting the Signal Spectrum

Use the MATLAB software to plot the power spectral density of the WCDMA signals at the input and output of the nonlinear amplifier. The plot allows you to visualize the spectral regrowth effects intrinsic to amplifier nonlinearity. Notice how the measurements reflect the spectral regrowth. (Note: the following code is just for visualizing signal spectra; it has nothing to do with obtaining the ACPR measurements).

```
win = hamming(1024);
[PSD1,F] = pwelch(txSignalBeforeAmplifier,win,50,1024,SampleRate,'centered');
[PSD2,F] = pwelch(txSignalAfterAmplifier,win,50,1024,SampleRate,'centered');
plot(F,10*log10(PSD1))
hold on
grid on
plot(F,10*log10(PSD2),'g')
legend('Amplifier input', 'Amplifier output')
```



Complementary Cumulative Distribution Function CCDF

Using the `comm.CCDF` System object you can measure the probability that the instantaneous power of a signal is above its specified level average power.

Selected Bibliography for Measurements

- [1] Proakis, J. G., *Digital Communications*, 4th Ed., McGraw-Hill, 2001.
- [2] Simon, M. K., and Alouini, M. S., *Digital Communication over Fading Channels - A Unified Approach to Performance Analysis*, 1st Ed., Wiley, 2000.
- [3] Simon, M. K., "On the bit-error probability of differentially encoded QPSK and offset QPSK in the presence of carrier synchronization", *IEEE Trans. Commun.*, vol. 54, May 2006, pp. 806-812.
- [4] Lee, P. J., "Computation of the bit error rate of coherent M-ary PSK with Gray code bit mapping", *IEEE Trans. Commun.*, Vol. COM-34, Number 5, pp. 488-491, 1986.
- [5] Cho, K., and Yoon, D., "On the general BER expression of one- and two-dimensional amplitude modulations", *IEEE Trans. Commun.*, vol. 50, no. 7, July 2002, pp. 1074-1080.
- [6] Simon, M. K., Hinedi, S. M., and Lindsey, W. C., *Digital Communication Techniques - Signal Design and Detection*, Prentice-Hall, 1995.
- [7] Sklar, B., *Digital Communications*, 2nd Ed., Prentice-Hall, 2001.
- [8] Lindsey, W. C., "Error probabilities for Rician fading multichannel reception of binary and N-ary signals", *IEEE Trans. Inform. Theory*, Vol. IT-10, pp. 339-350, 1964.
- [9] Odenwalder, J. P., *Error Control Coding Handbook* (Final report), Linkabit Corp., 15 July 1976.
- [10] Gulliver, T. A., "Matching Q-ary Reed-Solomon codes with M-ary modulation," *IEEE Trans. Commun.*, vol. 45, no. 11, Nov. 1997, pp. 1349-1353.
- [11] Jeruchim, Michel C., Philip Balaban, and K. Sam Shanmugan, *Simulation of Communication Systems*, New York, Plenum Press, 1992.
- [12] Frenger, Pal, Pal Orten, and Tony Ottosson, "Convolutional Codes with Optimum Distance Spectrum," *IEEE Communications Letters*, Vol. 3, #11, Nov. 1999, pp. 317-319

Filtering Section

- “Filtering” on page 16-2
- “Group Delay” on page 16-5
- “Pulse Shaping Using a Raised Cosine Filter” on page 16-7
- “Design Raised Cosine Filters Using MATLAB Functions” on page 16-13
- “Filter Using Simulink Raised Cosine Filter Blocks” on page 16-15
- “Design Raised Cosine Filters in Simulink” on page 16-21
- “Reduce ISI Using Raised Cosine Filtering” on page 16-24
- “Find Delay for Encoded and Filtered Signal” on page 16-29

Filtering

In this section...
“Filter Features” on page 16-2
“Selected Bibliography Filtering” on page 16-4

The Communications Toolbox software includes several functions, objects, and blocks that can help you design and use filters. Other filtering capabilities are in the Signal Processing Toolbox and the DSP System Toolbox. The sections of this chapter are as follows:

For an example involving raised cosine filters, type `showdemo rcosdemo`.

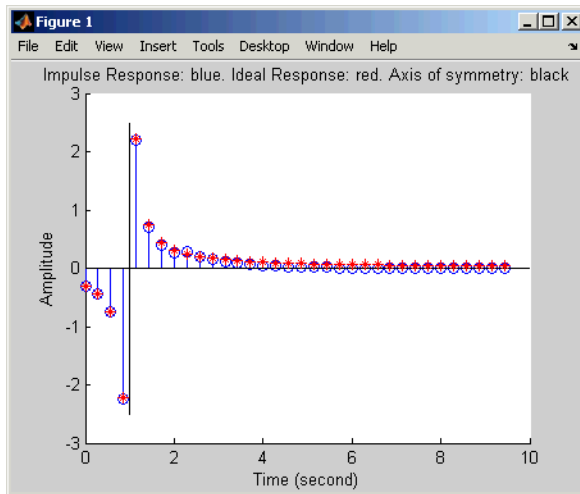
Filter Features

Without propagation delays, both Hilbert filters and raised cosine filters are noncausal. This means that the current output depends on the system's future input. In order to design only *realizable* filters, the `hilbair` function delays the input signal before producing an output. This delay, known as the filter's *group delay*, is the time between the filter's initial response and its peak response. The group delay is defined as

$$-\frac{d}{d\omega}\theta(\omega)$$

where θ represents the phase of the filter and ω represents the frequency in radians per second. This delay is set so that the impulse response before time zero is negligible and can safely be ignored by the function.

For example, the Hilbert filter whose impulse is shown below uses a group delay of one second. In the figure, the impulse response near time 0 is small and the large impulse response values occur near time 1.



Filtering tasks that blocks in the Communications Toolbox support include:

- “Filter Using Simulink Raised Cosine Filter Blocks” on page 16-15. Raised cosine filters are very commonly used for pulse shaping and matched filtering. The following block diagram illustrates a typical use of raised cosine filters.



- Shaping a signal using ideal rectangular pulses.
- Implementing an integrate-and-dump operation or a windowed integrator. An integrate-and-dump operation is often used in a receiver model when the system's transmitter uses an ideal rectangular-pulse model. Integrate-and-dump can also be used in fiber optics and in spread-spectrum communication systems such as CDMA (code division multiple access) applications.

Additional filtering capabilities exist in the Filter Designs and Multirate Filters libraries of the DSP System Toolbox product.

For more background information about filters and pulse shaping, see the works listed in the “Selected Bibliography Filtering” on page 16-4.

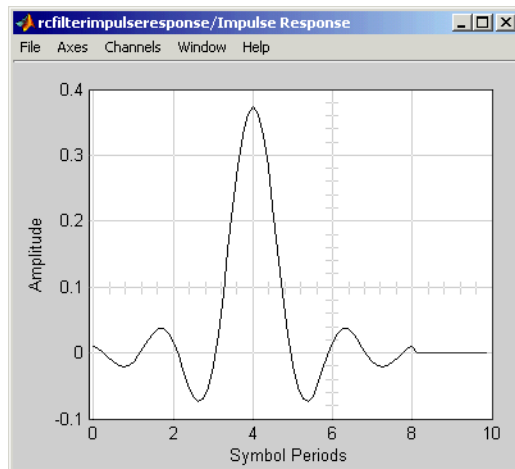
Selected Bibliography Filtering

- [1] Korn, Israel, *Digital Communications*, New York, Van Nostrand Reinhold, 1985.
- [2] Oppenheim, Alan V., and Ronald W. Schaffer, *Discrete-Time Signal Processing*, Englewood Cliffs, NJ, Prentice Hall, 1989.
- [3] Proakis, John G., *Digital Communications*, 3rd ed., New York, McGraw-Hill, 1995.
- [4] Rappaport, Theodore S., *Wireless Communications: Principles and Practice*, Upper Saddle River, NJ, Prentice Hall, 1996.
- [5] Sklar, Bernard, *Digital Communications: Fundamentals and Applications*, Englewood Cliffs, NJ, Prentice Hall, 1988.

Group Delay

The raised cosine filter blocks in the `commfilt2` library implement realizable filters by delaying the peak response. This delay, known as the filter's *group delay*, is the length of time between the filter's initial response and its peak response. The filter blocks in this library have a **Filter span in symbols** parameter, which is twice the group delay in symbols.

For example, the square-root raised cosine filter whose impulse response shown in the following figure uses a **Filter span in symbols** parameter of 8 in the filter block. In the figure, the initial impulse response is small and the peak impulse response occurs at the fourth symbol.

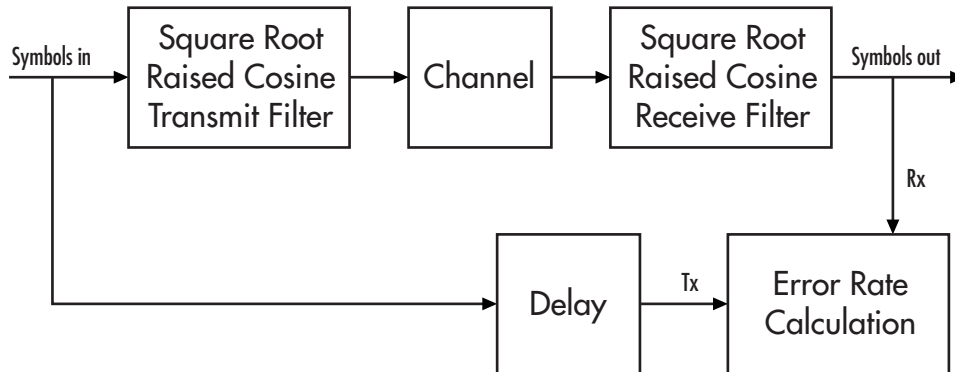


Implications of Delay for Simulations

A filter block's group delay has implications for other parts of your model. For example, suppose you compare the symbol streams marked Symbols In and Symbols Out in the schematics shown on the "Filtering" on page 16-2 page by plotting or computing an error rate. Use one of these methods to make sure you are comparing symbols that truly correspond to each other:

- Use the Delay block to delay the Symbols In signal, thus aligning it with the Symbols Out signal. Set the **Delay** parameter equal to the filter's group delay (or the sum of

both values, if your model uses a pair of square root raised cosine filter blocks). The following figure illustrates this usage.



- Use the Find Delay block to find the delay between the two signals and add that delay using the Delay block.
- When using the Error Rate Calculation block to compare the two signals, increase the **Receive delay** parameter by the group delay value (or the sum of both values, if your model uses a pair of square-root raised cosine filter blocks). The **Receive delay** parameter might include other delays as well, depending on the contents of your model.

For more information about how to manage delays in a model, see “Delays” on page 2-6.

Pulse Shaping Using a Raised Cosine Filter

Filter a 16-QAM signal using a pair of square root raised cosine matched filters. Plot the eye diagram and scatter plot of the signal. After passing the signal through an AWGN channel, calculate the number of bit errors.

Set the simulation parameters.

```
M = 16; % Modulation order
k = log2(M); % Bits/symbol
n = 20000; % Transmitted bits
nSamp = 4; % Samples per symbol
EbNo = 10; % Eb/No (dB)
```

Set the filter parameters.

```
span = 10; % Filter span in symbols
rolloff = 0.25; % Rolloff factor
```

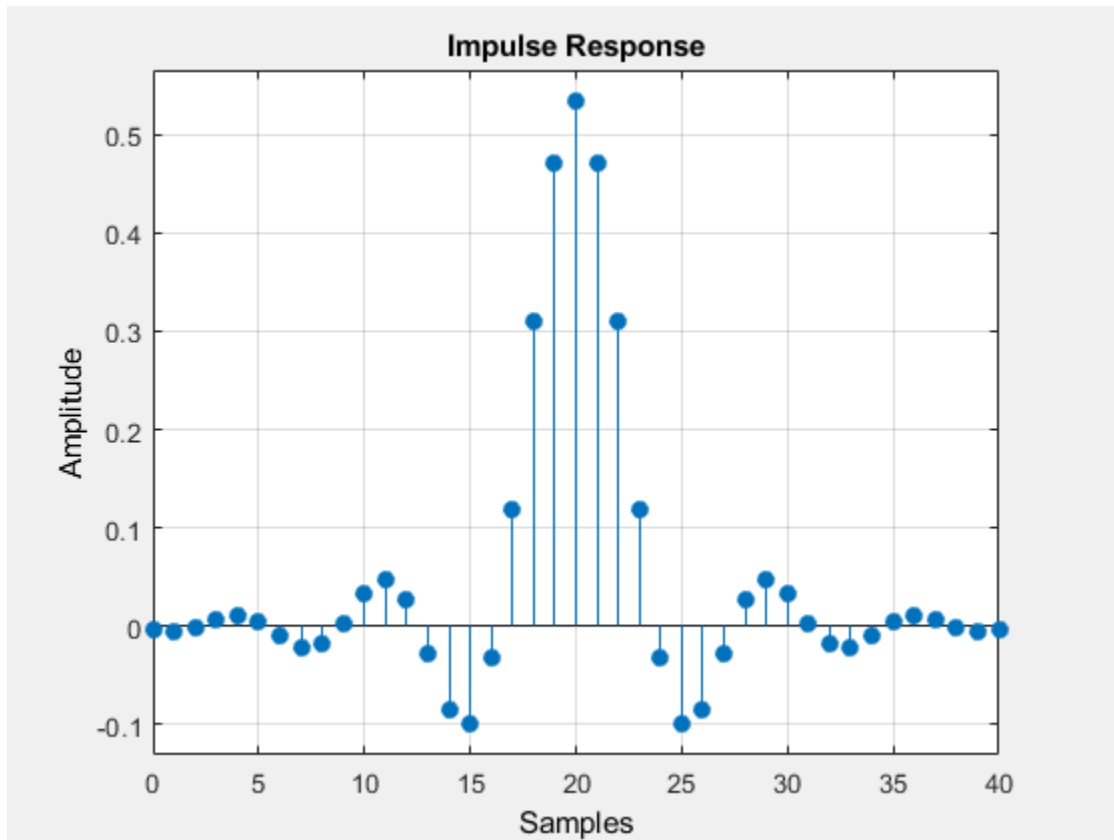
Create the raised cosine transmit and receive filters using the previously defined parameters.

```
txfilter = comm.RaisedCosineTransmitFilter('RolloffFactor',rolloff, ...
    'FilterSpanInSymbols',span,'OutputSamplesPerSymbol',nSamp);

rxfilter = comm.RaisedCosineReceiveFilter('RolloffFactor',rolloff, ...
    'FilterSpanInSymbols',span,'InputSamplesPerSymbol',nSamp, ...
    'DecimationFactor',nSamp);
```

Plot the impulse response of hTxFilter.

```
fvtool(txfilter,'impulse')
```



Calculate the delay through the matched filters. The group delay is half of the filter span through one filter and is, therefore, equal to the filter span for both filters. Multiply by the number of bits per symbol to get the delay in bits.

```
filtDelay = k*span;
```

Create an error rate counter System object. Set the ReceiveDelay property to account for the delay through the matched filters.

```
errorRate = comm.ErrorRate('ReceiveDelay',filtDelay);
```

Generate binary data.

```
x = randi([0 1],n,1);
```


Modulate the data.

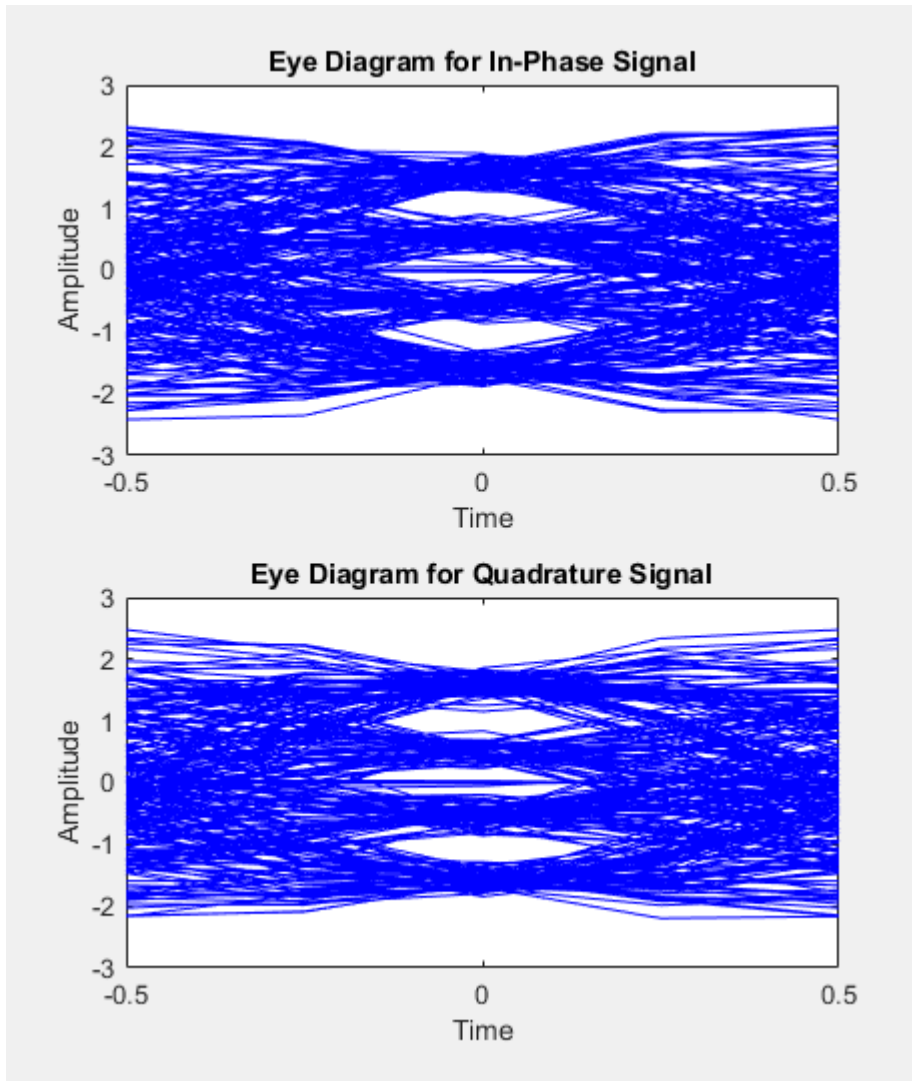
```
modSig = qammod(x,M, 'InputType', 'bit');
```

Filter the modulated signal.

```
txSig = txfilter(modSig);
```

Plot the eye diagram of the first 1000 samples.

```
eyediagram(txSig(1:1000), nSamp)
```

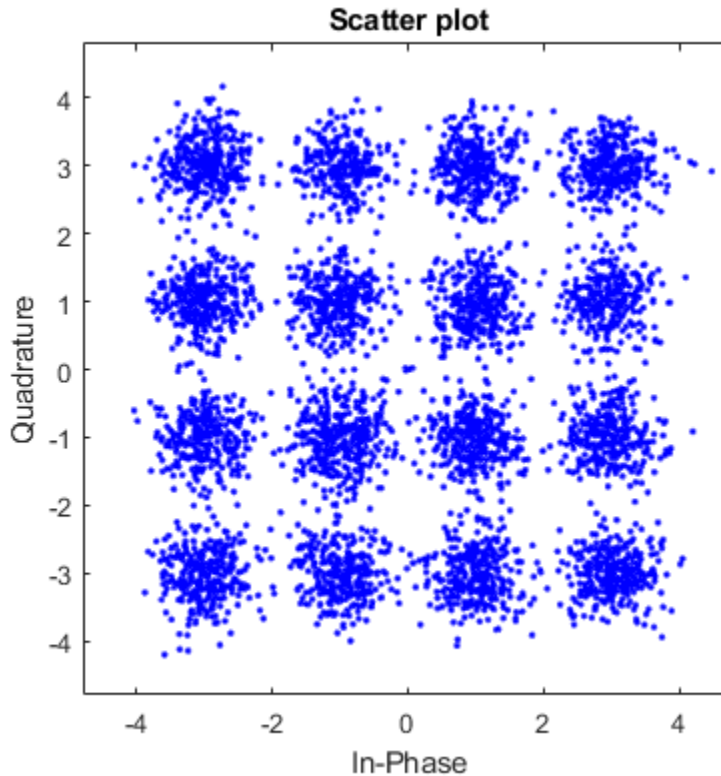


Calculate the signal-to-noise ratio (SNR) in dB given E_b/N_0 . Pass the transmitted signal through the AWGN channel using the `awgn` function.

```
SNR = EbNo + 10*log10(k) - 10*log10(nSamp);  
noisySig = awgn(txSig,SNR,'measured');
```

Filter the noisy signal and display its scatter plot.

```
rxSig = rxfilter(noisySig);  
scatterplot(rxSig)
```



Demodulate the filtered signal and calculate the error statistics. The delay through the filters is accounted for by the ReceiveDelay property in errorRate .

```
z = qamdemod(rxSig,M, 'OutputType', 'bit');  
  
errStat = errorRate(x,z);  
fprintf('\nBER = %5.2e\nBit Errors = %d\nBits Transmitted = %d\n',...  
        errStat)
```

BER = 1.85e-03
Bit Errors = 37
Bits Transmitted = 19960

Design Raised Cosine Filters Using MATLAB Functions

In this section...

“Section Overview” on page 16-13

“Example Designing a Square-Root Raised Cosine Filter” on page 16-13

Section Overview

The `rcosdesign` function designs (but does not apply) filters of these types:

- Finite impulse response (FIR) raised cosine filter
- FIR square-root raised cosine filter

The function returns the FIR coefficients as output.

Example Designing a Square-Root Raised Cosine Filter

For example, the command below designs a square-root raised cosine FIR filter with a rolloff of 0.25, a filter span of 6 symbols, and an oversampling factor of 2.

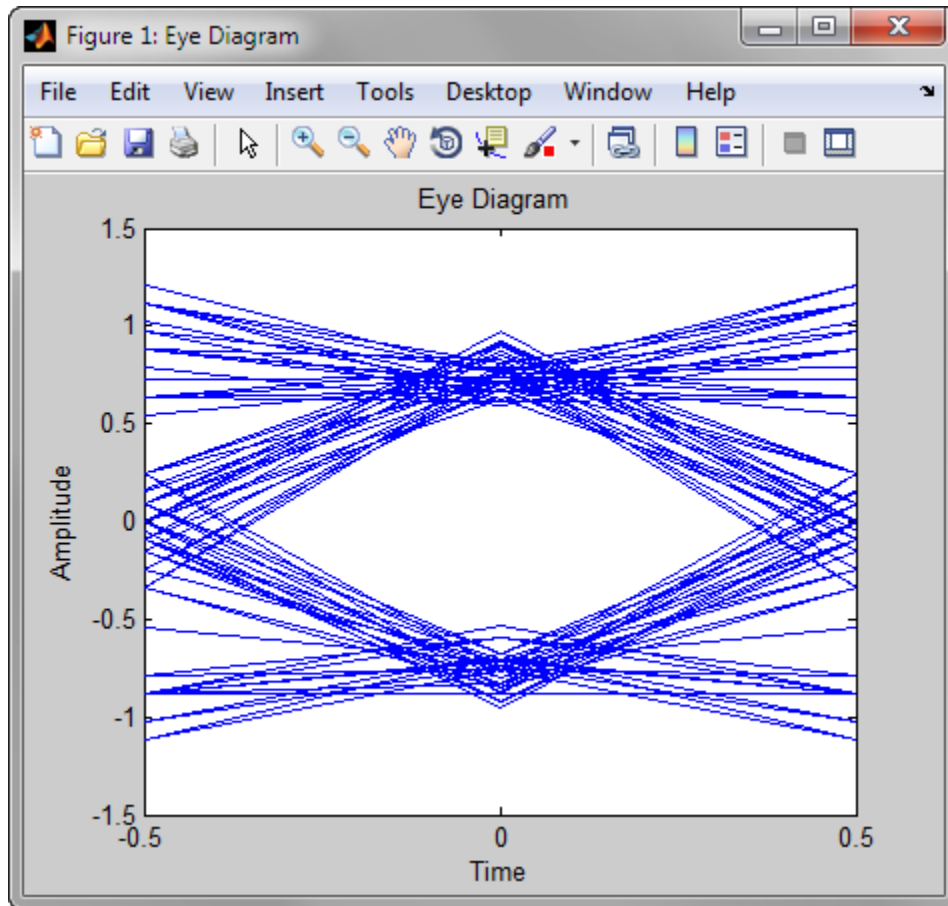
```
sps = 2;
num = rcosdesign(0.25, 6, sps)

num =
  Columns 1 through 7
  -0.0265    0.0462    0.0375   -0.1205   -0.0454    0.4399    0.7558
  Columns 8 through 13
  0.4399   -0.0454   -0.1205    0.0375    0.0462   -0.0265
```

Here, the vector `num` contains the coefficients of the filter, in ascending order of powers of z^{-1} .

You can use the `upfirdn` function to filter data with a raised cosine filter generated by `rcosdesign`. The following code illustrates this usage:

```
d = 2*randi([0 1], 100, 1)-1;
f = upfirdn(d, num, sps);
eyediagram(f(7:200), sps)
```



The eye diagram shows an imperfect eye because num characterizes a square-root filter.

Filter Using Simulink Raised Cosine Filter Blocks

The Raised Cosine Transmit Filter and Raised Cosine Receive Filter blocks are designed for raised cosine filtering. Each block can apply a square-root raised cosine filter or a normal raised cosine filter to a signal. You can vary the rolloff factor and span of the filter.

The Raised Cosine Transmit Filter and Raised Cosine Receive Filter blocks are tailored for use at the transmitter and receiver, respectively. The transmit filter outputs an upsampled (interpolated) signal, while the receive filter expects its input signal to be upsampled. The receive filter lets you choose whether to have the block downsample (decimate) the filtered signal before sending it to the output port.

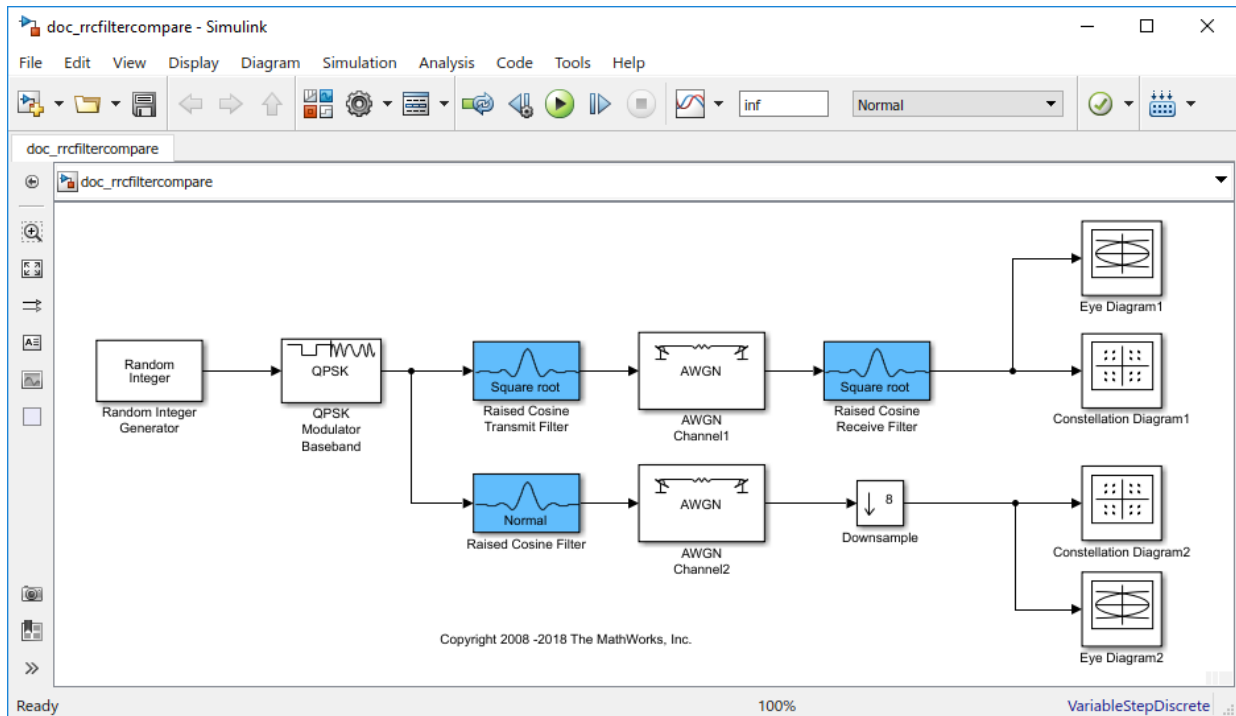
Both raised cosine filter blocks introduce a propagation delay, as described in “Group Delay” on page 16-5.

Combining Two Square-Root Raised Cosine Filters

This model shows how to split the filtering equally between the transmitter's filter and the receiver's filter by using a pair of square root raised cosine filters.

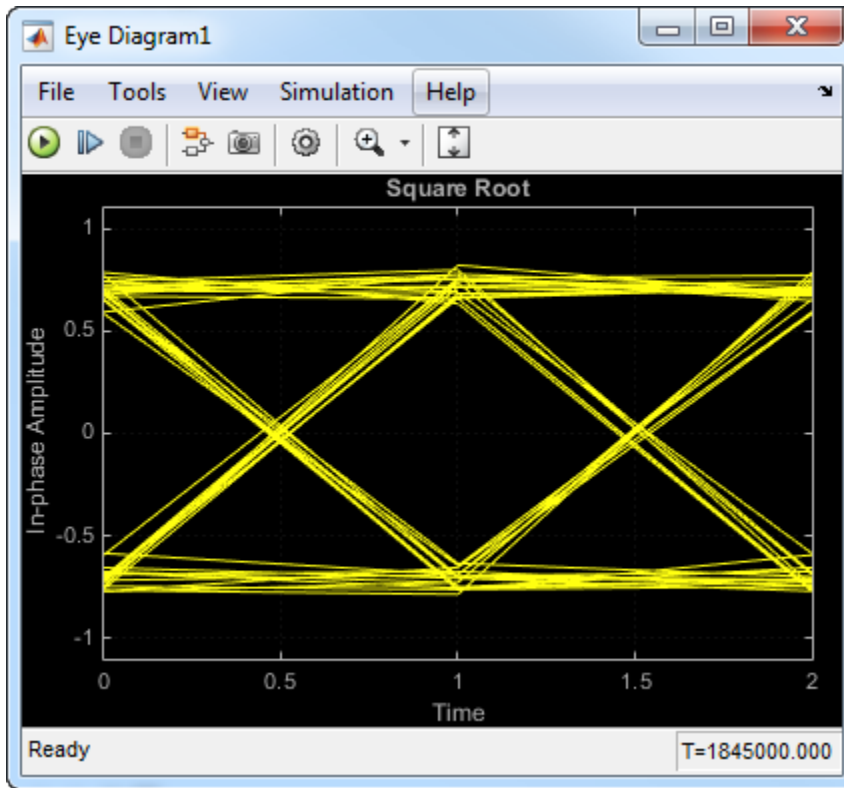
The use of two matched square root raised cosine filters is equivalent to a single normal raised cosine filter. To see this illustrated, load the model `doc_rrcfiltercompare` by typing the following at the MATLAB command line.

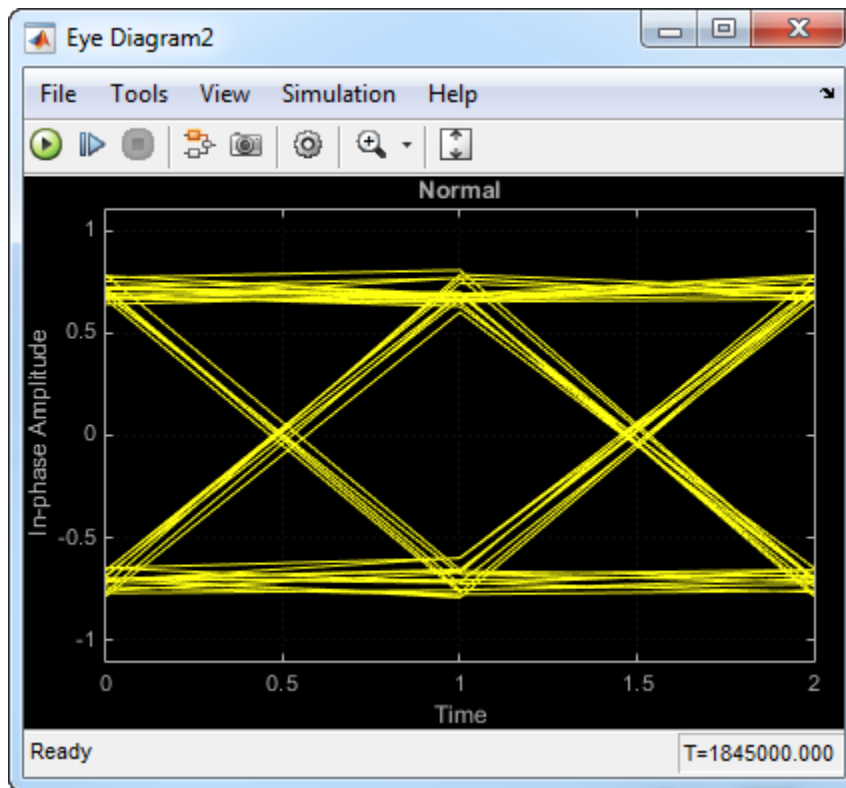
```
doc_rrcfiltercompare
```

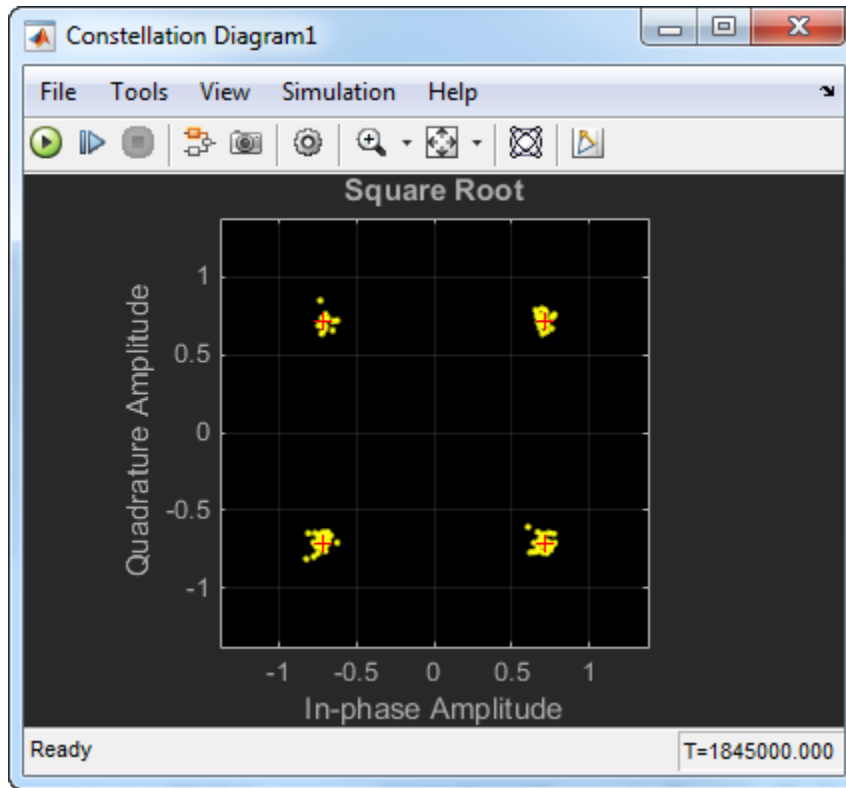


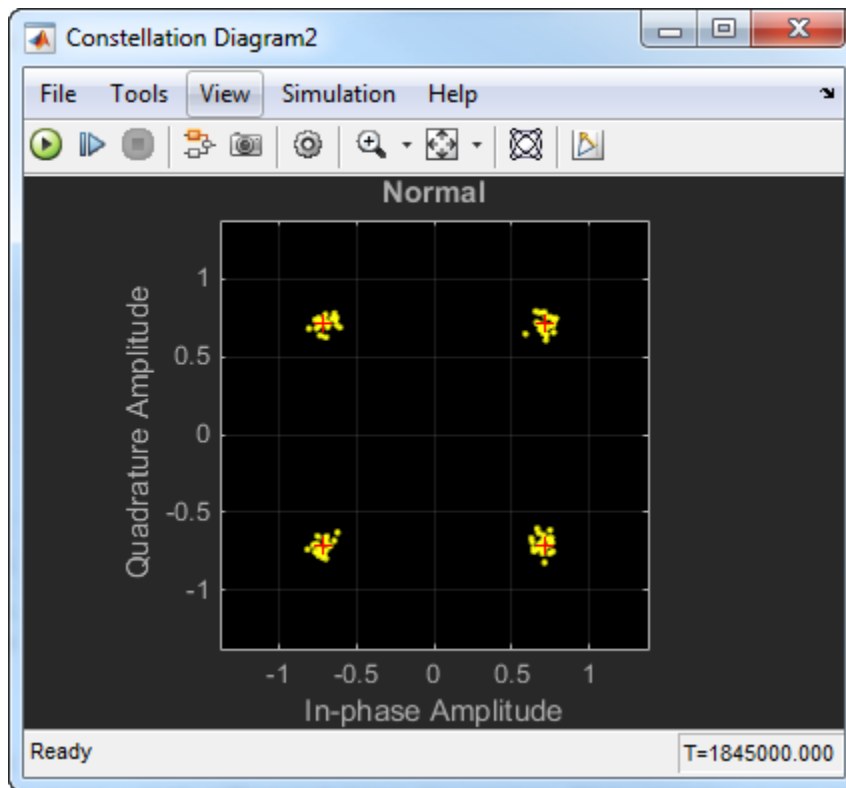
The filters share the same span and use the same number samples per symbol but the filters on the upper path have a square root shape while the filter on the lower path has the normal shape.

Run the model and observe the eye and constellation diagrams. The performance is nearly identical for the two methods. Note that the limited impulse response of practical square root raised cosine filters causes a slight difference between the response of two cascaded square root raised cosine filters and the response of one raised cosine filter.



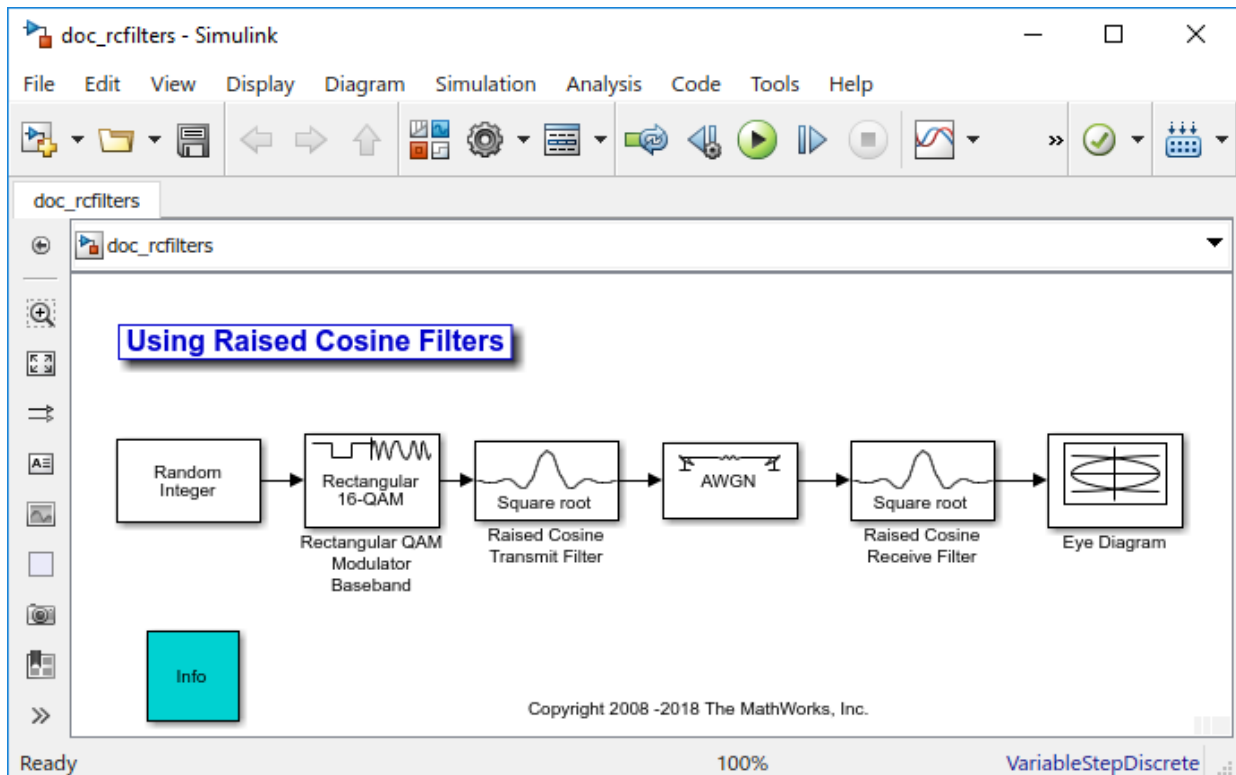






Design Raised Cosine Filters in Simulink

This example illustrates a typical setup in which a transmitter uses a square root raised cosine filter to perform pulse shaping and the corresponding receiver uses a square root raised cosine filter as a matched filter. The example plots an eye diagram from the filtered received signal.

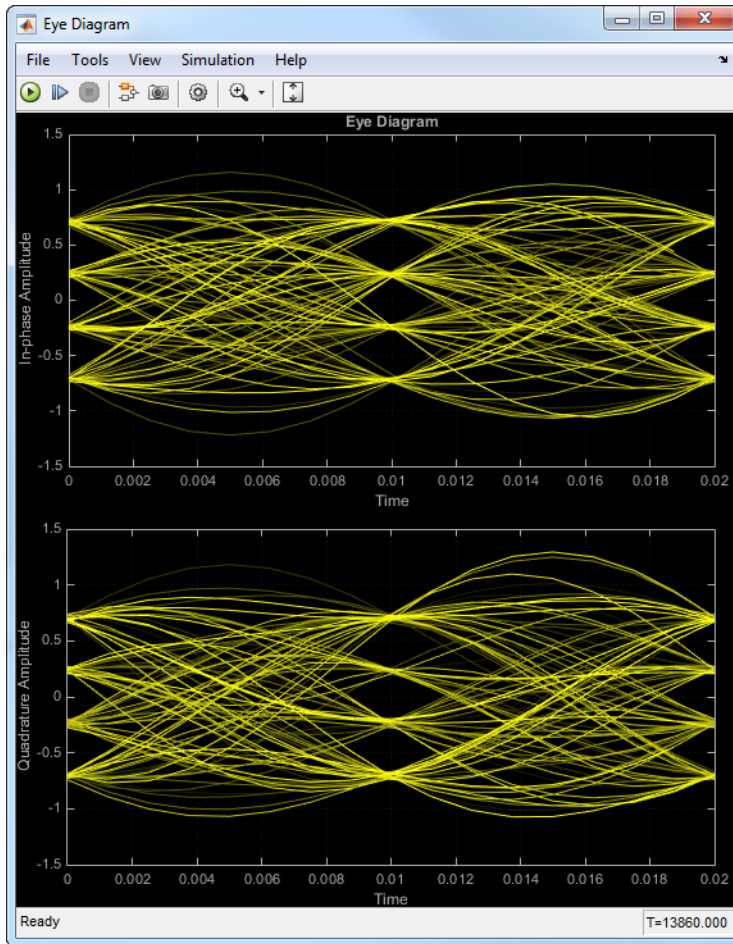


To open the model, enter `doc_rcfilters` at the MATLAB command line. The following is a summary of the block parameters used in the model:

- Random Integer Generator, in the Random Data Sources sublibrary of the Comm Sources library:
 - **M-ary number** is set to 16.

- **Sample time** is set to 1/100.
- **Frame-based outputs** is selected.
- **Samples per frame** is set to 100.
- Rectangular QAM Modulator Baseband, in the AM sublibrary of the Digital Baseband sublibrary of Modulation:
 - **Normalization method** is set to Peak Power.
 - **Peak power** is set to 1.
- Raised Cosine Transmit Filter, in the Comm Filters library:
 - **Filter span in symbols** is set to 8.
 - **Rolloff factor** is set to 0.2
- AWGN Channel, in the Channels library:
 - **Mode** is set to Signal to noise ratio (SNR).
 - **SNR** is set to 40.
 - **Input signal power** is set to 0.0694. The power gain of a square-root raised cosine transmit filter is $\frac{1}{N}$, where N represents the upsampling factor of the filter. The input signal power of filter is 0.5556. Because the **Peak power** of the 16-QAM Rectangular modulator is set to 1 watt, it translates to an average power of 0.5556. Therefore, the output signal power of filter is $\frac{0.5556}{8} = 0.0694$.
- Raised Cosine Receive Filter, in the Comm Filters library:
 - **Filter span in symbols** is set to 8.
 - **Rolloff factor** is set to 0.2.
- Eye Diagram, in the Comm Sinks library:
 - **Symbols per trace** is set to 2.
 - **Traces to display** is set to 100.

Running the simulation produces the following eye diagram. The eye diagram has two widely opened “eyes” that indicate appropriate instants at which to sample the filtered signal before demodulating. This illustrates the absence of intersymbol interference at the sampling instants of the received waveform.



The large signal-to-noise ratio in this example produces an eye diagram with large eye openings. If you decrease the **SNR** parameter in the AWGN Channel block, the eyes in the diagram will close more.

Reduce ISI Using Raised Cosine Filtering

Employ raised cosine filtering to reduce inter-symbol interference (ISI) that results from a nonlinear amplifier.

Initialize a simulation variable for modulation order.

```
M = 16; % Modulation order
```

Create square root raised cosine filter objects.

```
txfilter = comm.RaisedCosineTransmitFilter;  
rxfilter = comm.RaisedCosineReceiveFilter;
```

Create a memoryless nonlinearity System object to introduce nonlinear behavior to the modulated signal. Using name-value pairs, set the Method property to Saleh model to emulate a high power amplifier.

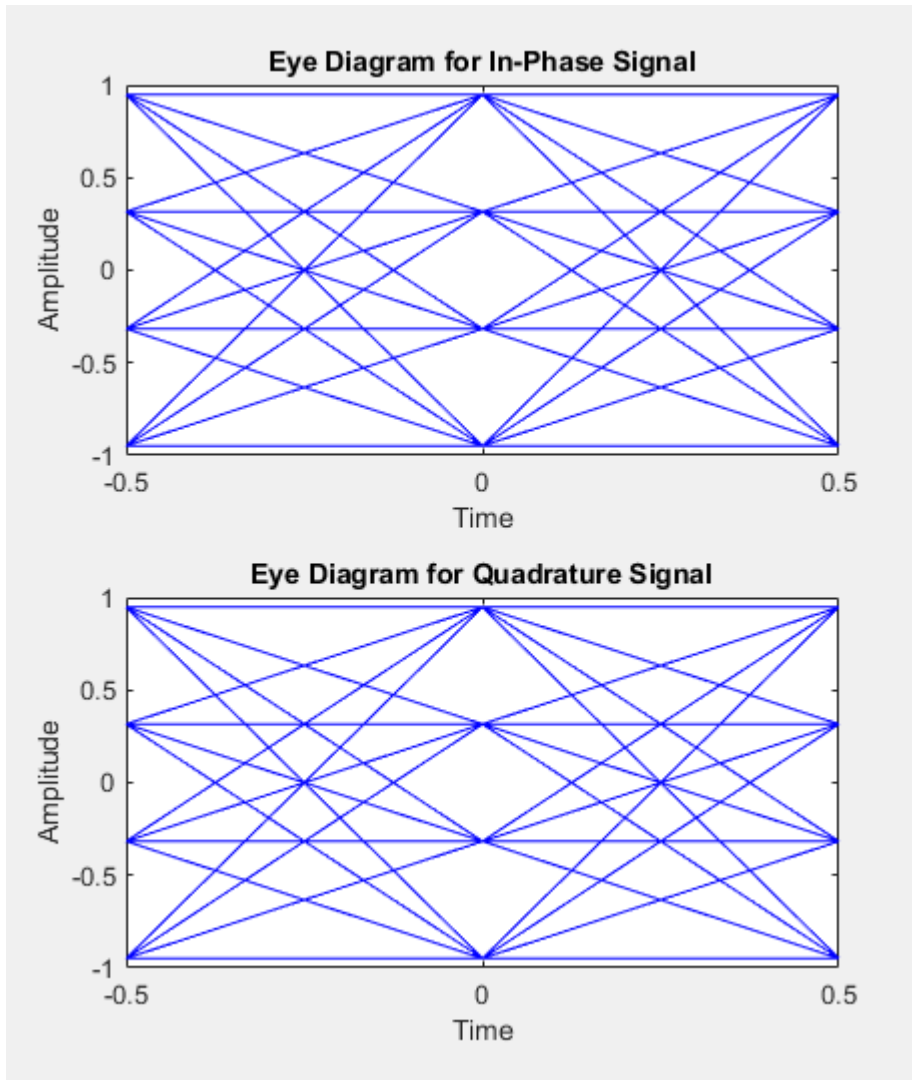
```
hpa = comm.MemorylessNonlinearity('Method','Saleh model', ...  
    'InputScaling',-10,'OutputScaling',0);
```

Generate random integers and apply 16-QAM modulation.

```
x = randi([0 M-1],1000,1);  
modSig = qammod(x,M,'UnitAveragePower',true);
```

Plot the eye diagram of the modulated signal. At time 0, there are three distinct "eyes" for 16-QAM modulation.

```
eyediagram(modSig,2)
```

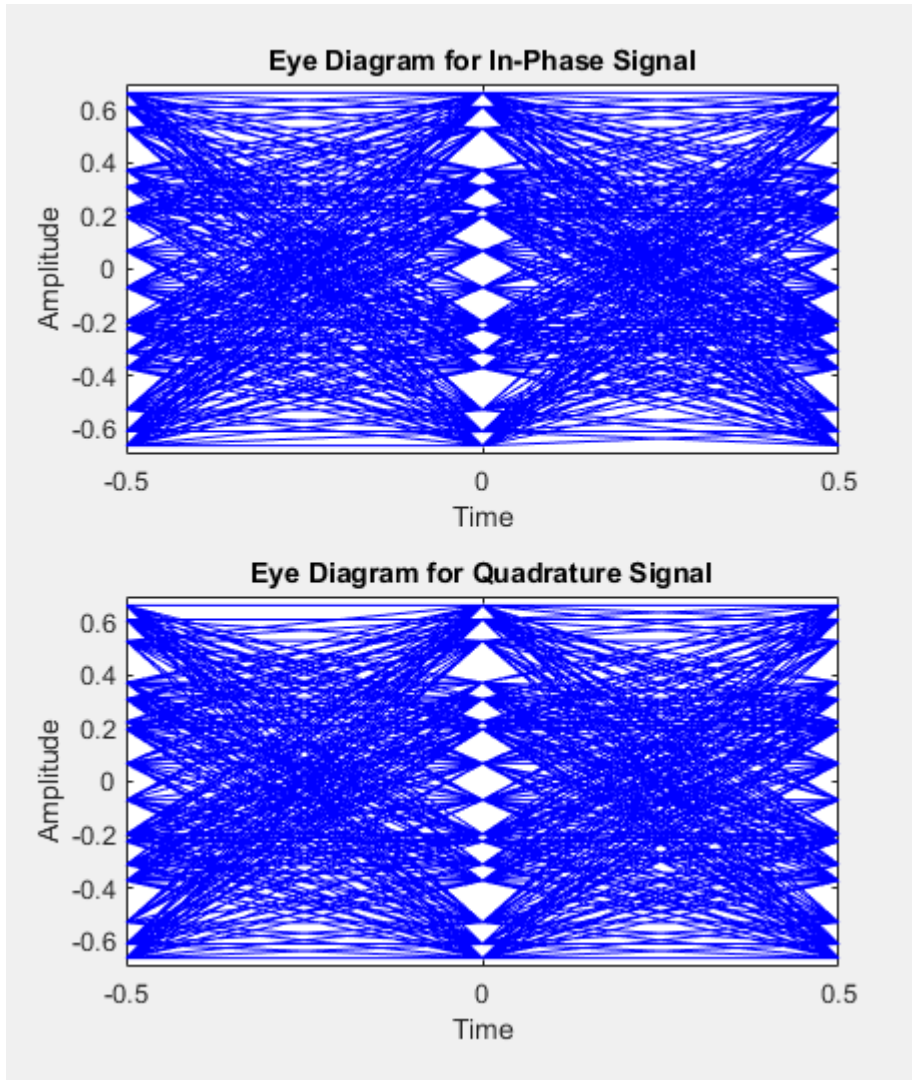



Amplify the modulated signal using `hpa`.

```
txSigNoFilt = hpa(modSig);
```

Plot the eye diagram of the amplified signal without RRC filtering. At time 0, there are multiple eyes. This is a result of inter-symbol interference from the nonlinear amplifier.

```
eyediagram(txSigNoFilt,2)
```



Filter the modulated signal using the RRC transmit filter.

```
filteredSig = txfilter(modSig);
```

Release `hpa` and amplify the filtered signal. The `release` function is needed because the input signal dimensions change due to filter interpolation.

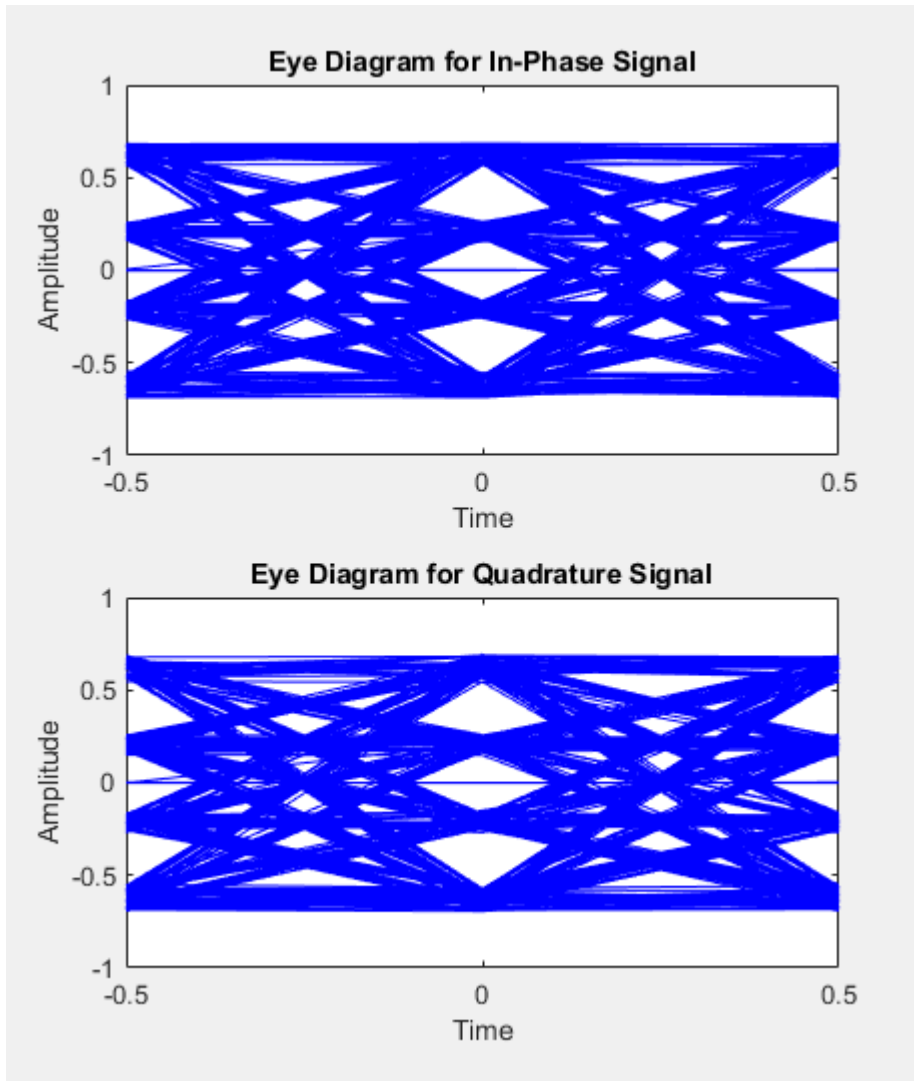
```
release(hpa)
txSig = hpa(filteredSig);
```

Filter `txSig` using the RRC matched receive filter.

```
rxSig = rxfilter(txSig);
```

Plot the eye diagram of the signal after the application of the receive filter. There are once again three distinct eyes as the matched RRC filters mitigate ISI.

```
eyediagram(rxSig,2)
```



Find Delay for Encoded and Filtered Signal

Determine the delay for a convolutionally encoded and filtered link. Use the delay to accurately determine the number of bit errors.

Create a QPSK modulator and demodulator pair. Specify the objects to operate on bits.

```
qpskmod = comm.QPSKModulator('BitInput',true);  
qpskdemod = comm.QPSKDemodulator('BitOutput',true);
```

Create a raised cosine transmit and receive filter pair.

```
txfilt = comm.RaisedCosineTransmitFilter;  
rxfilt = comm.RaisedCosineReceiveFilter;
```

Create a convolutional encoder and Viterbi decoder pair.

```
convEnc = comm.ConvolutionalEncoder;  
vitDec = comm.ViterbiDecoder('InputFormat','Hard');
```

Generate random binary data. Convolutionally encode the data.

```
txData = randi([0 1],1000,1);  
encData = convEnc(txData);
```

Modulate the encoded data. Pass the modulated data through the raised cosine transmit filter.

```
modSig = qpskmod(encData);  
txSig = txfilt(modSig);
```

Pass the filtered signal through an AWGN channel.

```
rxSig = awgn(txSig,20,'measured');
```

Filter and then demodulate the received signal.

```
filtSig = rxfilt(rxSig);  
demodSig = qpskdemod(filtSig);
```

Decode the demodulated data.

```
rxData = vitDec(demodSig);
```

Find the delay between the transmitted and received binary data by using the `finddelay` function.

```
td = finddelay(txData,rxData)
```

```
td = 44
```

Confirm that the computed delay matches the expected delay, which is equal to the sum of the group delay of the matched filters and the traceback depth of the Viterbi decoder.

```
tdexpected = (txfilt.FilterSpanInSymbols + rxfilt.FilterSpanInSymbols)/2 + ...  
             vitDec.TracebackDepth;  
isequal(td,tdexpected)
```

```
ans = logical  
     1
```

Calculate the number of bit errors by discarding the last `td` bits from the transmitted sequence and discarding the first `td` bits from the received sequence.

```
numErrors = biterr(txData(1:end-td),rxData(td+1:end))
```

```
numErrors = 0
```

Visual Analysis

- “Constellation Visualization” on page 17-2
- “Plot Signal Constellations” on page 17-9
- “Eye Diagram Analysis” on page 17-15
- “Scatter Plots and Constellation Diagrams” on page 17-21
- “Channel Visualization” on page 17-29
- “Visualize RF Impairments” on page 17-38

Constellation Visualization

Some linear modulator blocks provide the capability to visualize a signal constellation right from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. The following blocks support constellation visualization:

- BPSK Modulator Baseband
- QPSK Modulator Baseband
- M-PSK Modulator Baseband
- M-PAM Modulator Baseband
- Rectangular QAM Modulator Baseband
- General QAM Modulator Baseband

Note To display Fixed-Point settings, you need a Fixed-Point Designer™ user license.

Clicking **View Constellation** on a linear modulator block mask, plots the signal constellation using the block's mask parameters. If you set a modulator block to output single or fixed-point data types, clicking **View Constellation** generates two signal constellations plots overlaid on each other.

- One plot provides a reference constellation using double precision data type
- The other plot provides data whose data type selection is defined in the block mask

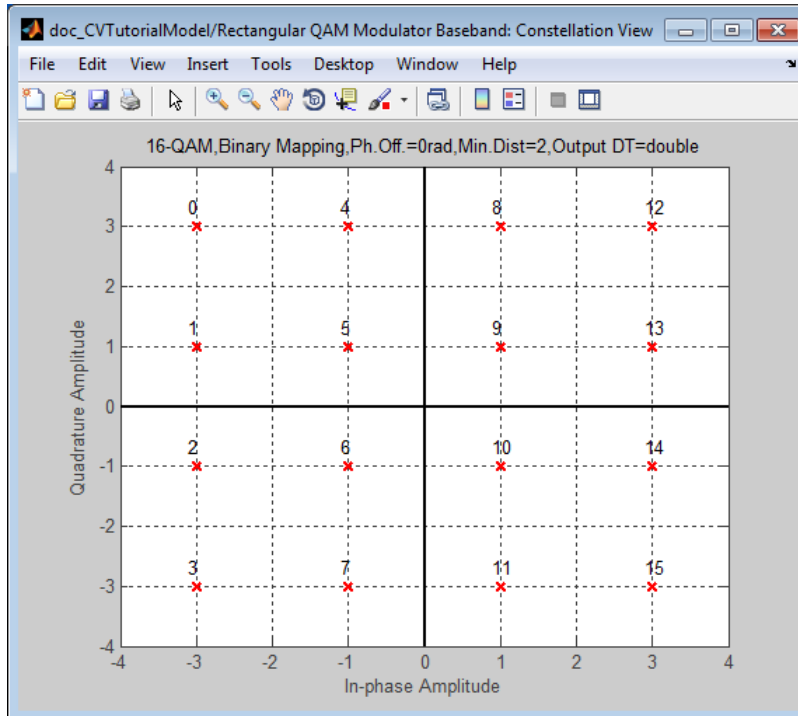
The title of the plot indicates the values of significant parameters. You can use the full array of MATLAB plot tools to manipulate plot figures. Selecting **Inherit via back propagation** for the **Output Data Type** generates a constellation plot with double as the **Output data type**.

Observe Modulator Design Affect Signal Constellation

In this tutorial, you will make changes to the modulator block. Without actually applying the changes to the model, you will observe how these changes effect the signal constellation.

- 1 Open the `constellation_visualization_tutorial` model by typing `doc_CVTutorialModel` at the MATLAB command line.

- 2 Double-click the Rectangular QAM Modulator Baseband block.
- 3 Next, click **View Constellation**

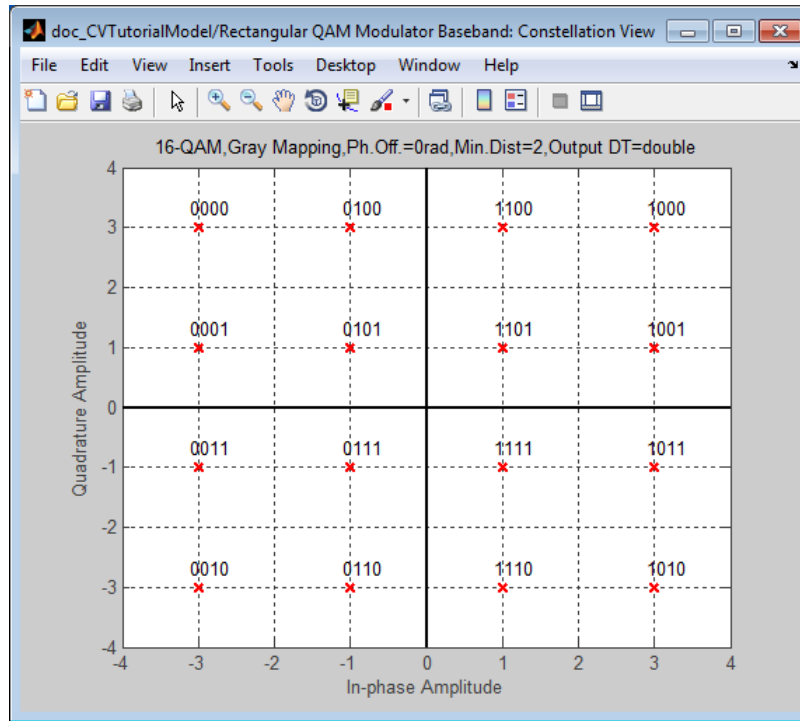


The constellation plot shows that the constellation:

- Uses a 16-QAM modulation scheme
- Uses Binary constellation mapping
- Has 0 degree phase offset
- Has a minimum distance between two constellation points of 2

The constellation plot also shows that the signal has a double precision data type. Because the **Input type** is integer, the constellation has integer symbol mapping.

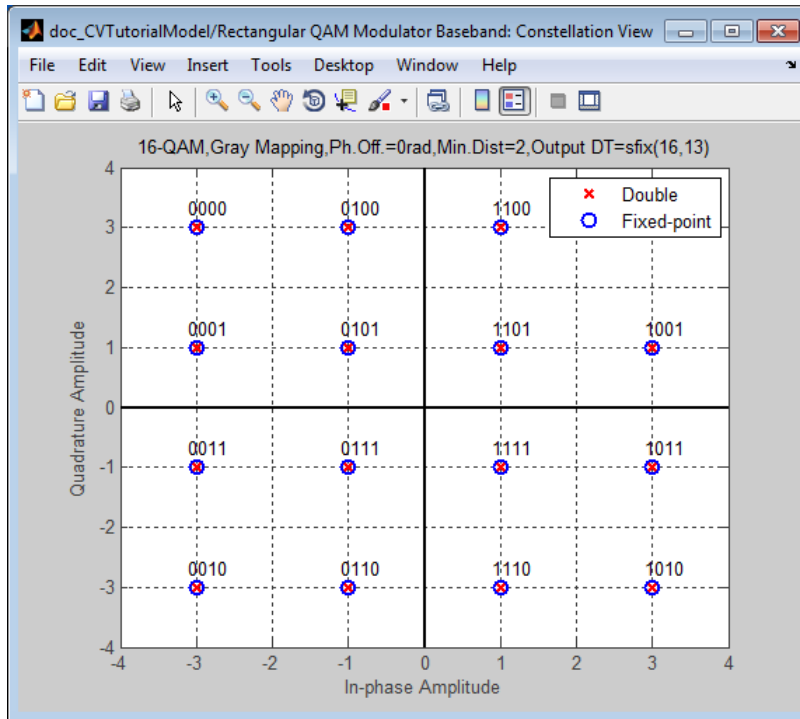
- 4 From the block mask, select **Bit** for the **Input type** parameter.
- 5 Select **Gray** for the **Constellation ordering** parameter.
- 6 Click **View Constellation**, and observe the results. Even though you did not click **Apply**, making these changes part of the model, the constellation plot still updates. The plot indicates gray constellation ordering using a bit representation of symbols.



- 7 You can overlay and compare the effect that two different data type selections have on a signal constellation. For example, you can compare the effect of changing **Output data type** from double to Fixed-point on the signal constellation.

To compare settings, perform the following tasks:

- Click the **Data Types** tab.
 - Set the **Output data type** parameter to Fixed-point.
 - Set the **Output word length** parameter to 16.
 - Set the **Set Output fraction length to** parameter to Best precision.
- 8 Click **Main** tab, and then click **View Constellation**.

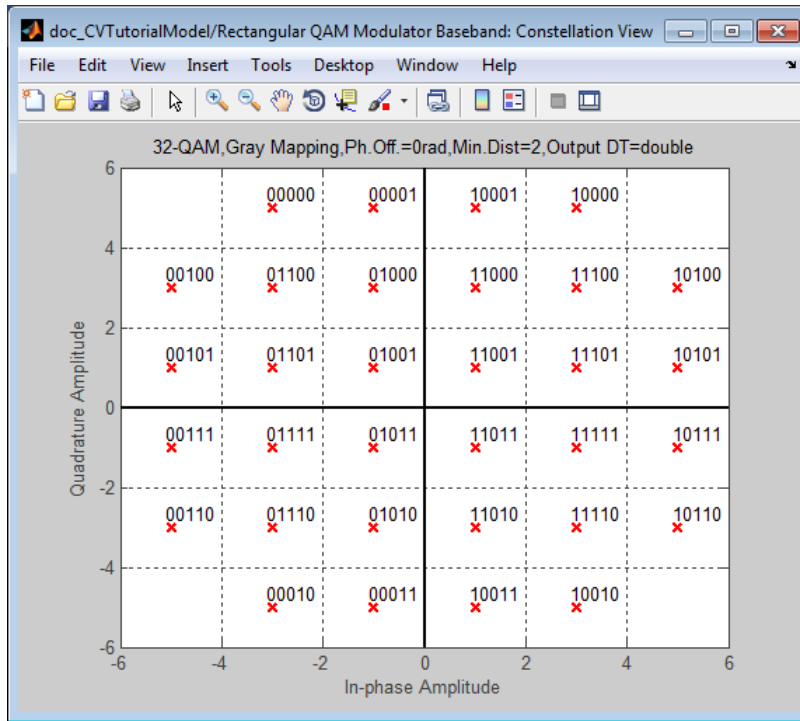


The plot overlays the fixed-point constellation on top of the double-precision constellation.

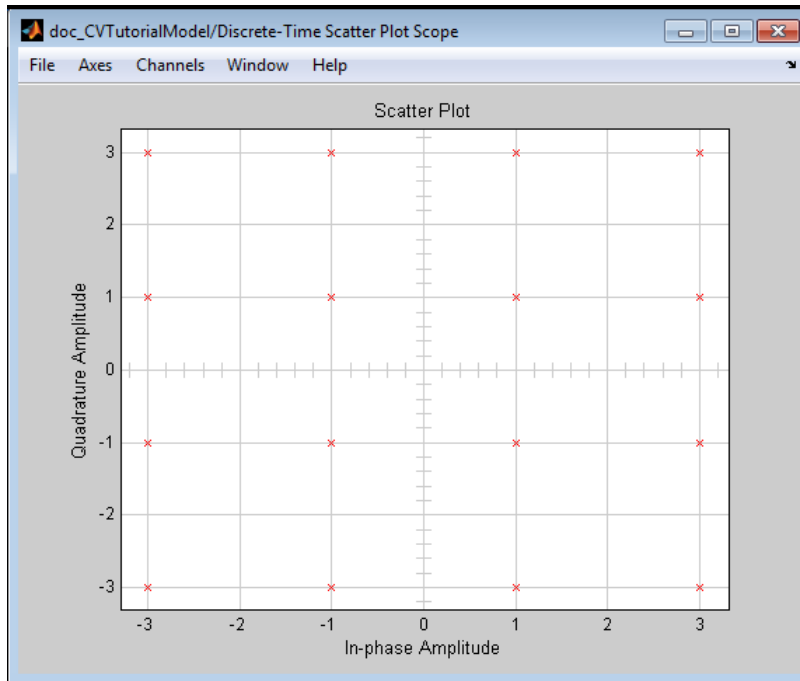
- 9 You can specify a block parameter value using variables defined in the MATLAB workspace. To define a variable, type `M=32` in the MATLAB workspace.

Note The model workspace in Simulink has priority over the base workspace in MATLAB.

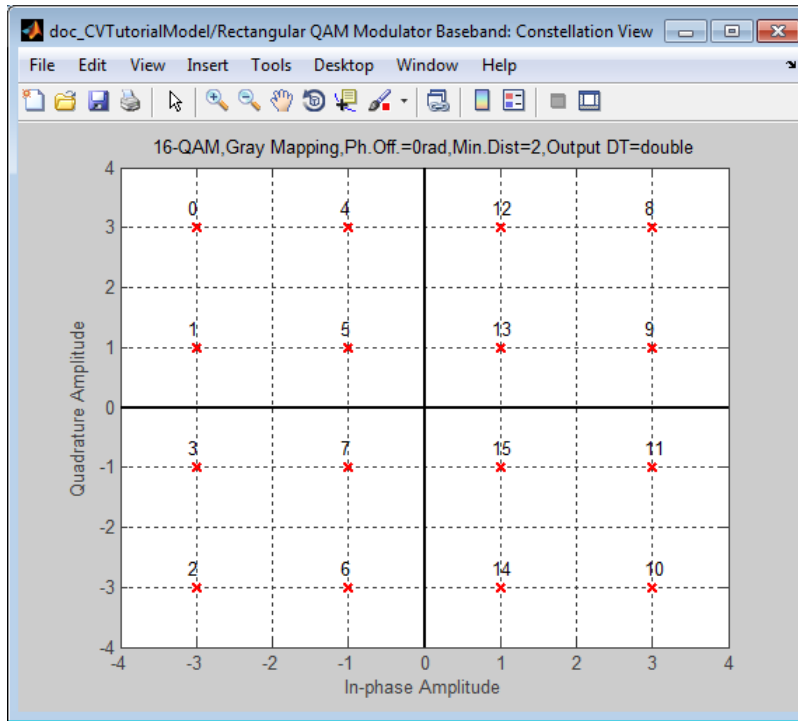
- 10 In the block mask, click the **Main** tab, and then type `M` for the **M-ary number** parameter. This parameter allows the block to use the variable value you defined in MATLAB workspace.
- 11 Click the **Data Types** tab and then select `double` for the **Output data type** parameter.
- 12 Click the **Main** tab. Then, click the **View Constellation** button and observe the results.



- 13 You can also use the Constellation Visualization feature while a simulation is running. Type $M=16$ in the MATLAB workspace, select Integer for the **Input type** and click **Apply**.
- 14 Simulate the model by clicking Run in the Simulink model window.



- 15** While the simulation is running, click **View Constellation**. Compare the signal constellation to the scatter plot generated in the previous step.



- 16** End the simulation by clicking the Stop button in the Simulink model window.

The Constellation Visualization feature provides full access to the MATLAB plotting capabilities, including: capturing a figure, saving a figure in multiple file formats, changing display settings, or saving files for archiving purposes. To capture a figure, select **Edit > Copy Figure**.

Using this tutorial, you have generated numerous constellation plots. If you close the Simulink model or delete the modulator block from the model, all the plots will close.

Tip If you capture a figure you want to archive for future use, save the figure before closing the model.

- 17** Close the Simulink model, and observe that all of the constellation figures also close.

Plot Signal Constellations

In this section...

“Create 16-PSK Constellation Diagram” on page 17-9

“Create 32-QAM Constellation Diagram” on page 17-10

“Create 8-QAM Gray Coded Constellation Diagram” on page 17-11

“Plot a Triangular Constellation for QAM” on page 17-13

Create 16-PSK Constellation Diagram

This example shows how to plot a PSK constellation having 16 points.

Set the parameters for 16-PSK modulation with no phase offset and binary symbol mapping.

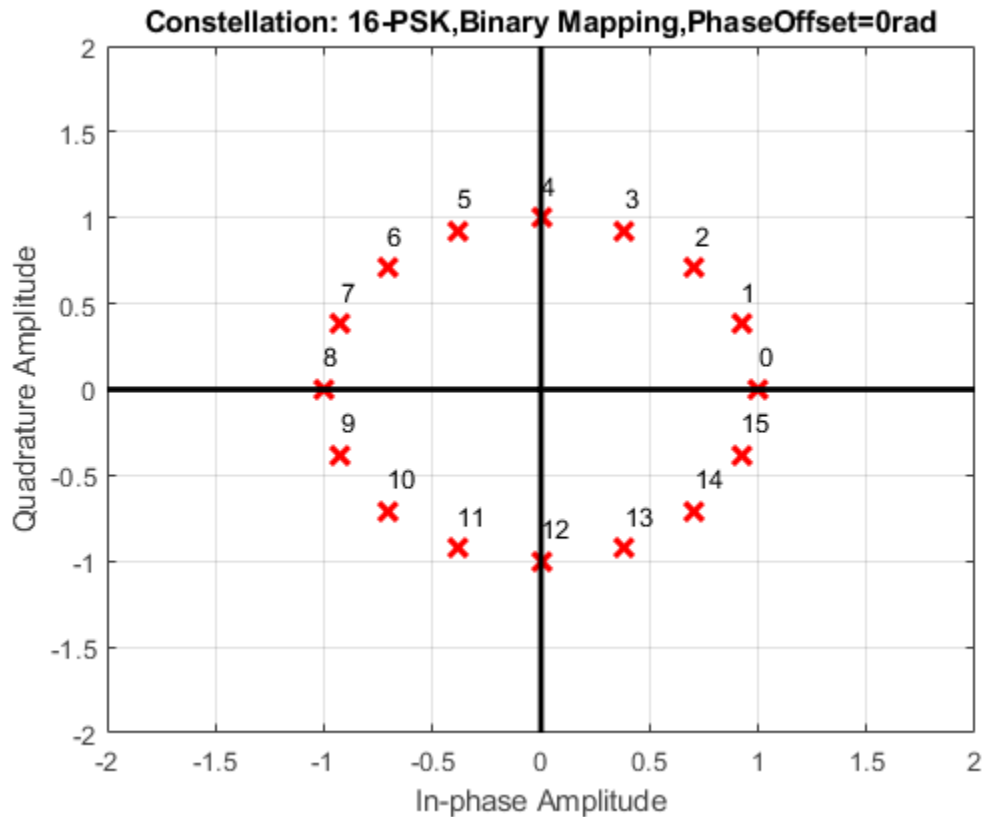
```
M = 16;           % Modulation alphabet size
phOffset = 0;    % Phase offset
symMap = 'binary'; % Symbol mapping (either 'binary' or 'gray')
```

Construct the modulator object.

```
pskModulator = comm.PSKModulator(M,phOffset,'SymbolMapping',symMap);
```

Plot the constellation.

```
constellation(pskModulator)
```



Create 32-QAM Constellation Diagram

This example shows how to plot a QAM constellation having 32 points.

Use the `qammod` function to generate the 32-QAM symbols with binary symbol ordering.

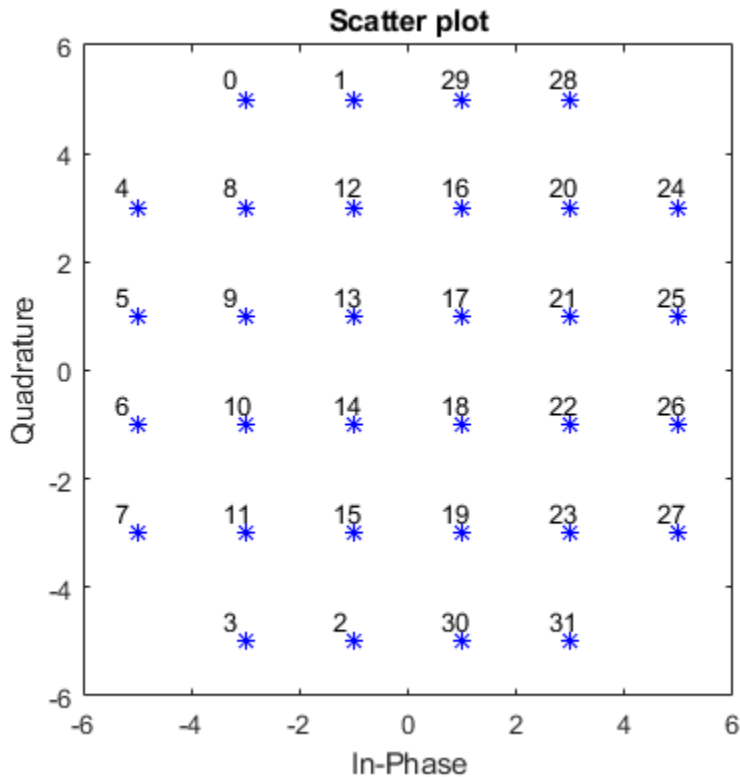
```
M = 32;  
data = 0:M-1;  
sym = qammod(data,M,'bin');
```

Plot the constellation. Label the order of the constellation symbols.


```

scatterplot(sym,1,0,'b*');
for k = 1:M
    text(real(sym(k))-0.4,imag(sym(k))+0.4,num2str(data(k)));
end
axis([-6 6 -6 6])

```



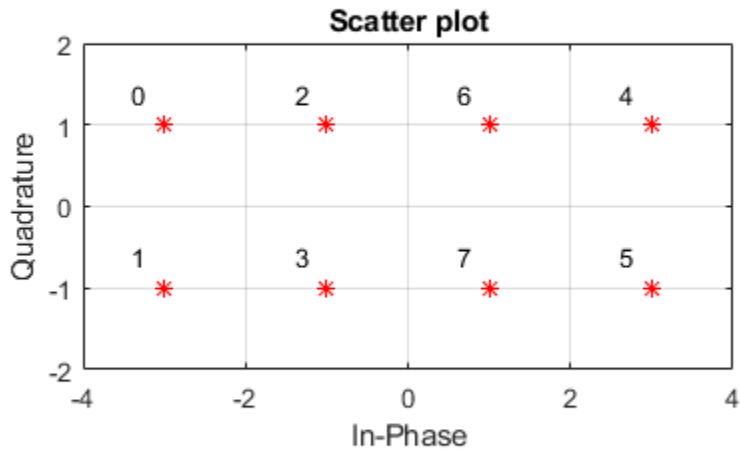
Create 8-QAM Gray Coded Constellation Diagram

Use the `qammod` function to generate the 8-QAM symbols with Gray symbol ordering. Note that Gray coding is the default symbol mapping for the `qammod` function.

```
M = 8;  
data = 0:M-1;  
sym = qammod(data,M);
```

Plot the constellation. Label the order of the constellation symbols.

```
scatterplot(sym,1,0,'r*');  
grid on  
for k = 1:M  
    text(real(sym(k))-0.4,imag(sym(k))+0.4,num2str(data(k)));  
end  
axis([-4 4 -2 2])
```



Plot a Triangular Constellation for QAM

This example shows how to plot a customized QAM reference constellation.

Describe the constellation.

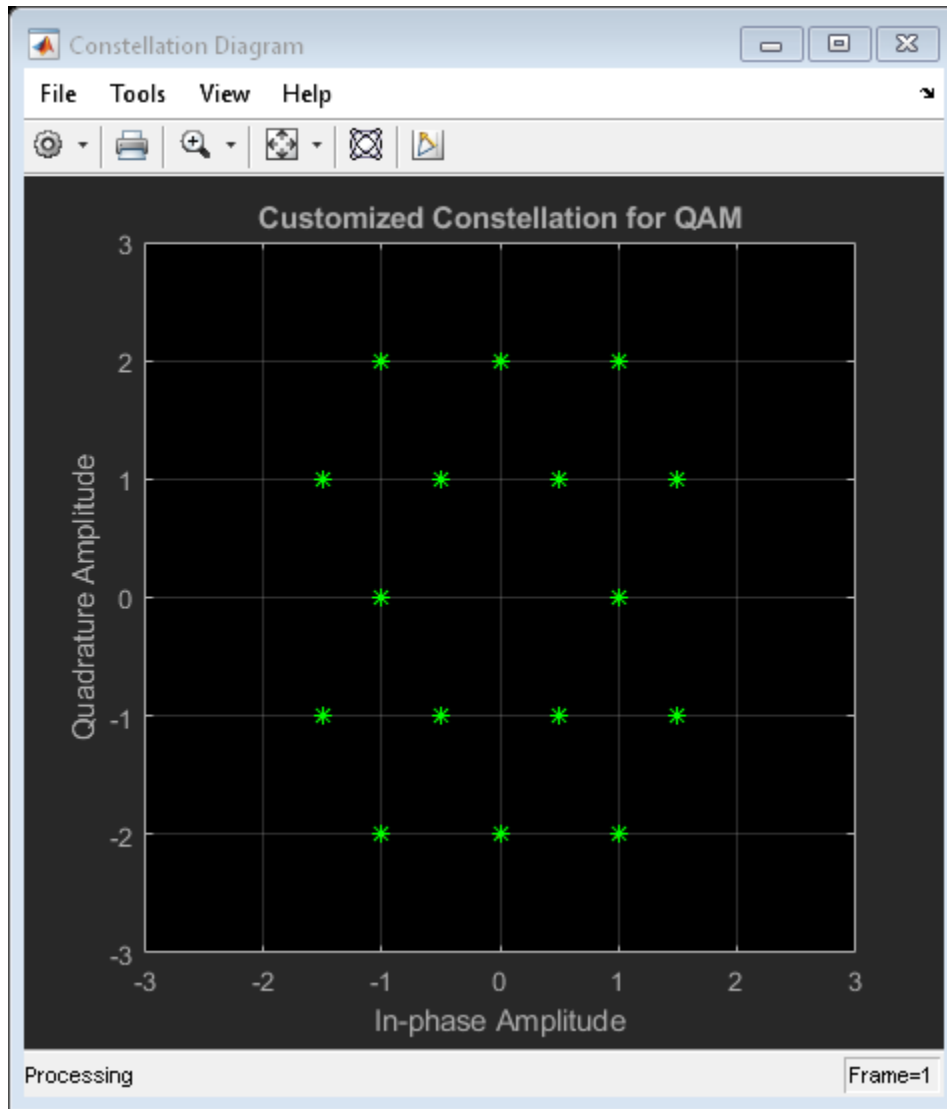
```
inphase = [1/2 -1/2 1 0 3/2 -3/2 1 -1];  
quadr = [1 1 0 2 1 1 2 2];  
inphase = [inphase; -inphase];  
inphase = inphase(:);  
quadr = [quadr; -quadr];  
quadr = quadr(:);  
refConst = inphase + 1i*quadr;
```

Construct a constellation diagram System object using name-value pairs to specify the title, the axes limits, the reference marker type, and the reference marker color.

```
constDiagram = comm.ConstellationDiagram('Title','Customized Constellation for QAM', .  
    'XLimits',[-3 3],'YLimits',[-3 3], ...  
    'ReferenceConstellation',refConst, ...  
    'ReferenceMarker','*','ReferenceColor',[0 1 0]);
```

Plot the customized constellation.

```
constDiagram(refConst)
```



See Also

"Constellation Visualization" on page 17-2

Eye Diagram Analysis

In digital communications, an eye diagram provides a visual indication of how noise might impact system performance.

Use the EyeScope tool to examine the data that an eye diagram object contains. EyeScope shows both the eye diagram plot and measurement results in a unified, graphical environment. You can import, and compare measurement results for, multiple eye diagram objects.

For information about constructing an eye diagram object, running a simulation, collecting data, and analyzing the simulated data, refer to the 'Eye Diagram Measurements' example. The Eye Diagram and Scatter Plot example covers eye diagram analysis applied to a communications system.

For a complete list of EyeScope measurements definitions, refer to 'Measurements' in the *Communications Toolbox User's Guide*.

For instructions on how to perform basic EyeScope tasks, see the EyeScope reference page.

Note `commscope.eyediagram` will be removed in a future release. Use `comm.EyeDiagram` instead.

Import Eye Diagrams and Compare Measurement Results

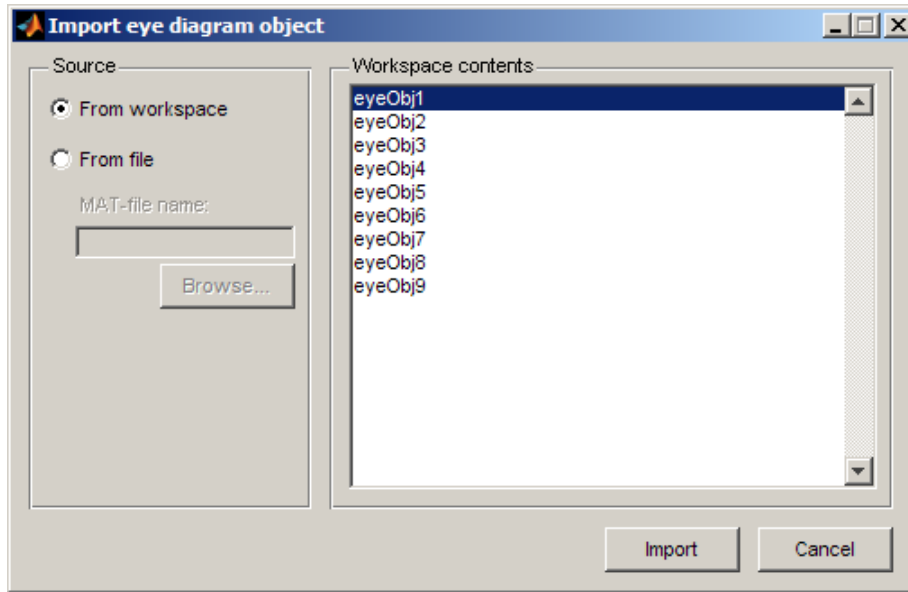
This section provides a step-by-step introduction for using EyeScope to import eye diagram objects, select and change which eye diagram measurements EyeScope displays, compare measurement results, and print a plot object.

MATLAB software includes a set of data containing nine eye diagram objects, which you can import into EyeScope. While EyeScope can import eye diagram objects from either the workspace or a MAT-file, this introduction covers importing from the workspace. EyeScope reconstructs the variable names it imports to reflect the origin of the eye diagram object.

- 1 Type `load commeye_EyeMeasureDemoData` at the MATLAB command line to load nine eye diagram objects into the MATLAB workspace.
- 2 Type `eyescope` at the MATLAB command line to start the EyeScope tool.

- 3 In the EyeScope window, select **File > Import Eye Diagram Object**.

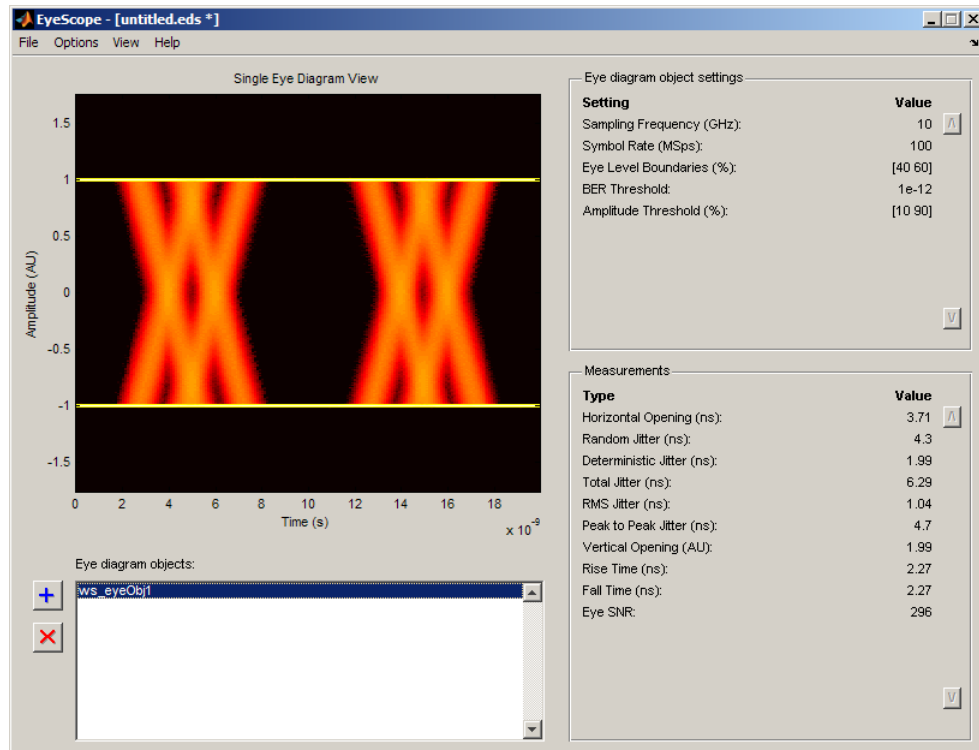
The **Import eye diagram object** dialog box opens.




In this window, the **Workspace contents** panel displays all eye diagram objects available in the source location.

- 4 Select **eyeObj1** and click **Import**. EyeScope imports the object, displaying an image in the object plot and listing the file name in the **Eye diagram objects** list.

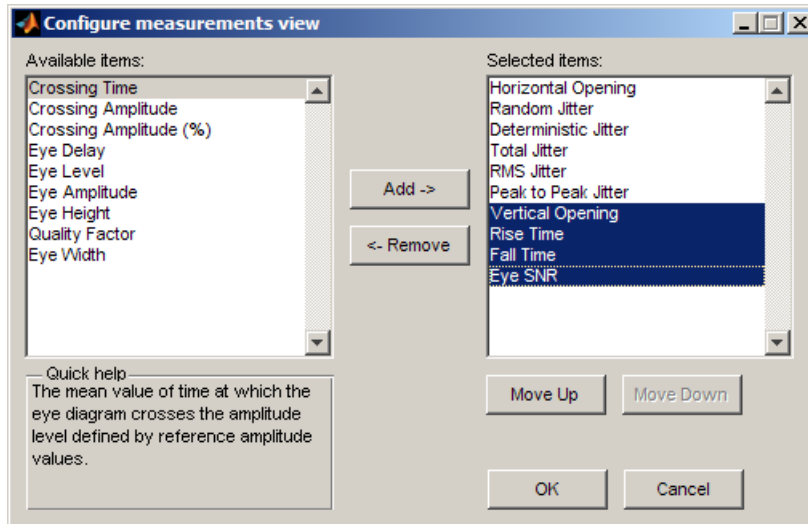
Note Object names associated with eye diagram objects that you import from the work space begin with the prefix **ws**.




Review the image and note the default **Eye diagram object settings** and **Measurements** selections. For more information, refer to the EyeScope reference page.

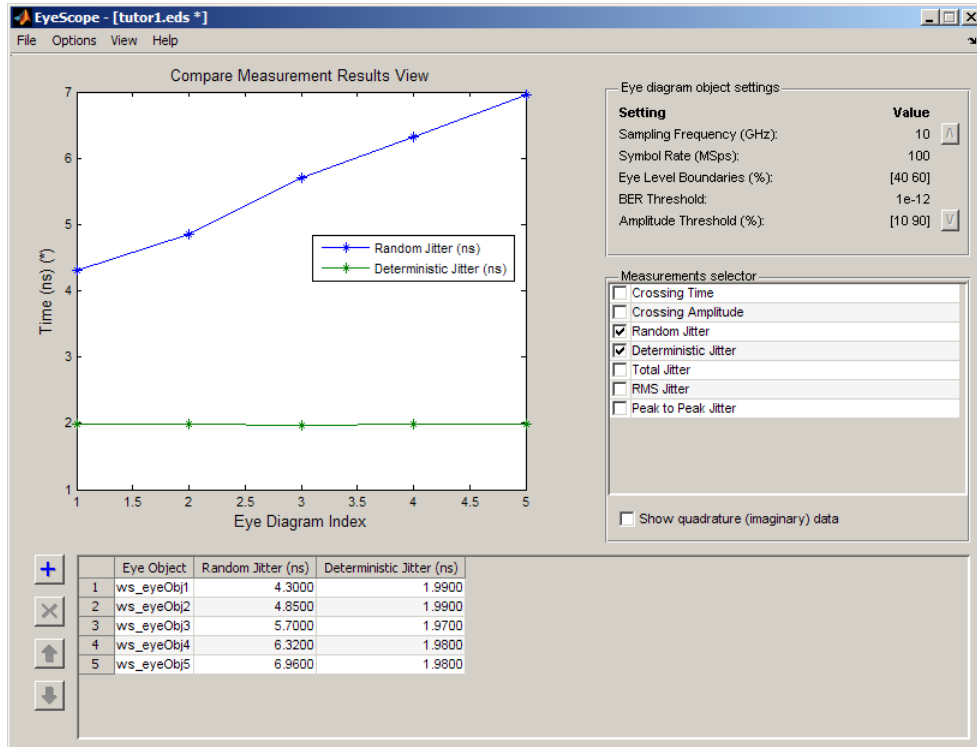
- 5 In the EyeScope window, click the Import  button.
- 6 From the Import eye diagram object window, click to select eyeObj5 then click the **Import** button.
 - The EyeScope window changes, displaying a new plot and adding ws_eyeObj5 to the **Eye diagram objects** list. EyeScope displays the same settings and measurements for both eye diagram objects.
 - You can switch between the eyediagram plots EyeScope displays by clicking on an object name in the Eye diagram object list.
 - Next, click **ws_eyeObj1** and note the EyeScope plot and measurement values changes.

- 7 To change or remove measurements from the EyeScope display:
- Select **Options > Measurements View**. The **Configure measurement view** shuttle control opens.



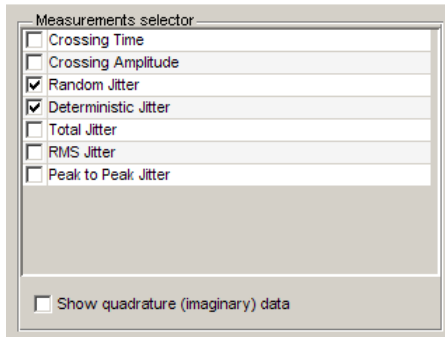
- Hold down the <Ctrl> key and click to select Vertical Opening, Rise Time, Fall Time, Eye SNR. Then click **Remove**.
- 8 From the left side of the shuttle control, select **Crossing Time** and **Crossing Amplitude** and then click **Add**. To display EyeScope with these new settings, click **OK**. EyeScope's **Measurement** region displays Crossing Time and Crossing Amplitude at the bottom of the Measurements section.
- 9 Change the list order so that **Crossing Time** and **Crossing Amplitude** appear at the top of the list.
- Select **Options > Measurements View**.
 - When the **Configure measurement view** shuttle control opens, hold down the <Ctrl> key and click to select **Crossing Time** and **Crossing Amplitude**.
 - Click the **Move Up** button until these selections appear at the top of the list. Then, click **OK**.
- 10 Select **File > Save session as** and then type a file name in the pop-up window.
- 11 Import **ws_eyeObj2**, **ws_eyeObj3**, and **ws_eyeObj4**. EyeScope now contains eye diagram objects 1, 5, 2, 3, and 4 in the list.
- 12 Select **ws_eyeObj5**, and click the delete  button.

- 13** Click **File > Import Eye Diagram Object**, and select **ws_eyeObj5**.
- 14** To compare measurement results for multiple eye diagram objects, click **View > Compare Measurement Results View**.



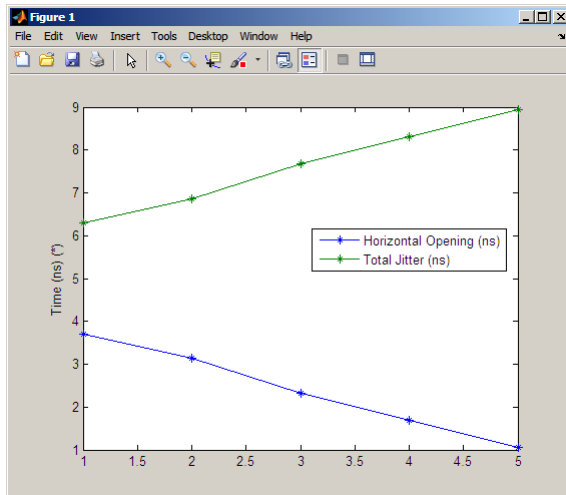
In the data set, random jitter increases from experiment 1 to experiment 5, as you can see in both the table and plot figure.

- 15** To include any data from the Measurements selection you chose earlier in this procedure, use the **Measurement selector**. Go to the **Measurement selector** and select **Total Jitter**. The object plot updates to display the additional measurements.



You can also remove measurements from the plot display. In the **Measurements selector**, select **Random Jitter** and **Deterministic Jitter**. The object plot updates, removing these two measurements.

- 16** To print the plot display, select **File > Print to Figure**. From **Figure** window, click the print button.



Scatter Plots and Constellation Diagrams

A scatter plot or constellation diagram is used to visualize the constellation of a digitally modulated signal.

To produce a scatter plot from a signal, use the `scatterplot` function or use the `comm.ConstellationDiagram` System object. A scatter plot or constellation diagram can be useful when comparing system performance to a published standard, such as 3GPP or DVB.

You create the `comm.ConstellationDiagram` object with a default object or by defining name-value pairs.

View Signals Using Constellation Diagrams

This example shows how to use constellation diagrams to view QPSK transmitted and received signals which are pulse shaped with a raised cosine filter.

Create a QPSK modulator.

```
qpsk = comm.QPSKModulator;
```

Create a raised cosine transmit filter with samples per symbol, `sps`, equal to 16.

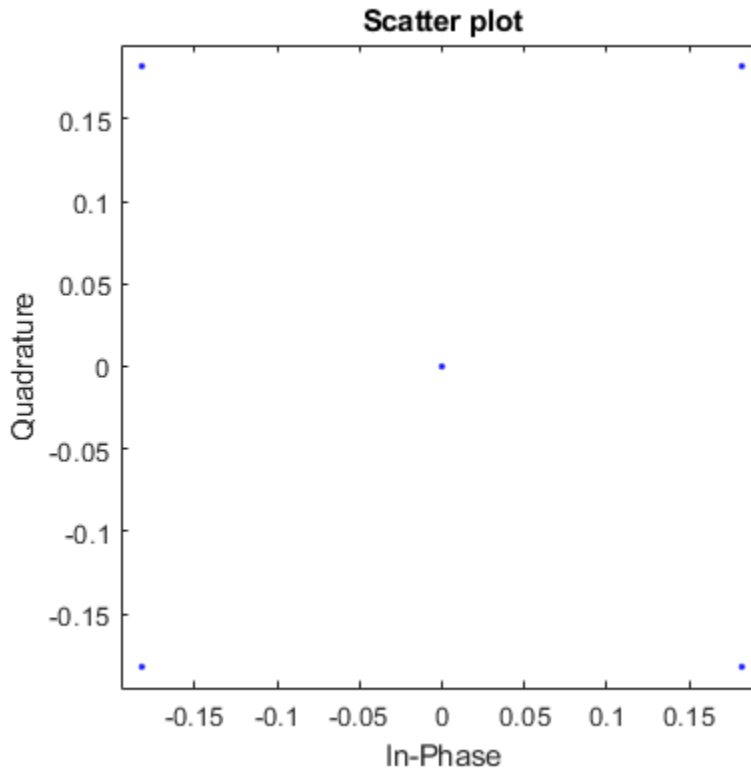
```
sps = 16;  
txfilter = comm.RaisedCosineTransmitFilter('Shape','Normal', ...  
    'RolloffFactor',0.22, ...  
    'FilterSpanInSymbols',20, ...  
    'OutputSamplesPerSymbol',sps);
```

Generate data symbols, apply QPSK modulation, and pass the modulated data through the raised cosine transmit filter.

```
data = randi([0 3],200,1);  
modData = qpsk(data);  
txSig = txfilter(modData);
```

You can display the constellation diagram of the transmitted signal using `scatterplot`. Since the signal is oversampled at the filter output, you need to decimate by the number of samples per symbol so that the scatter plot does not show the transition path between constellation points. If the signal had a timing offset, you could provide that as an input parameter to display the signal constellation with the timing offset corrected.

```
scatterplot(txSig,sps)
```



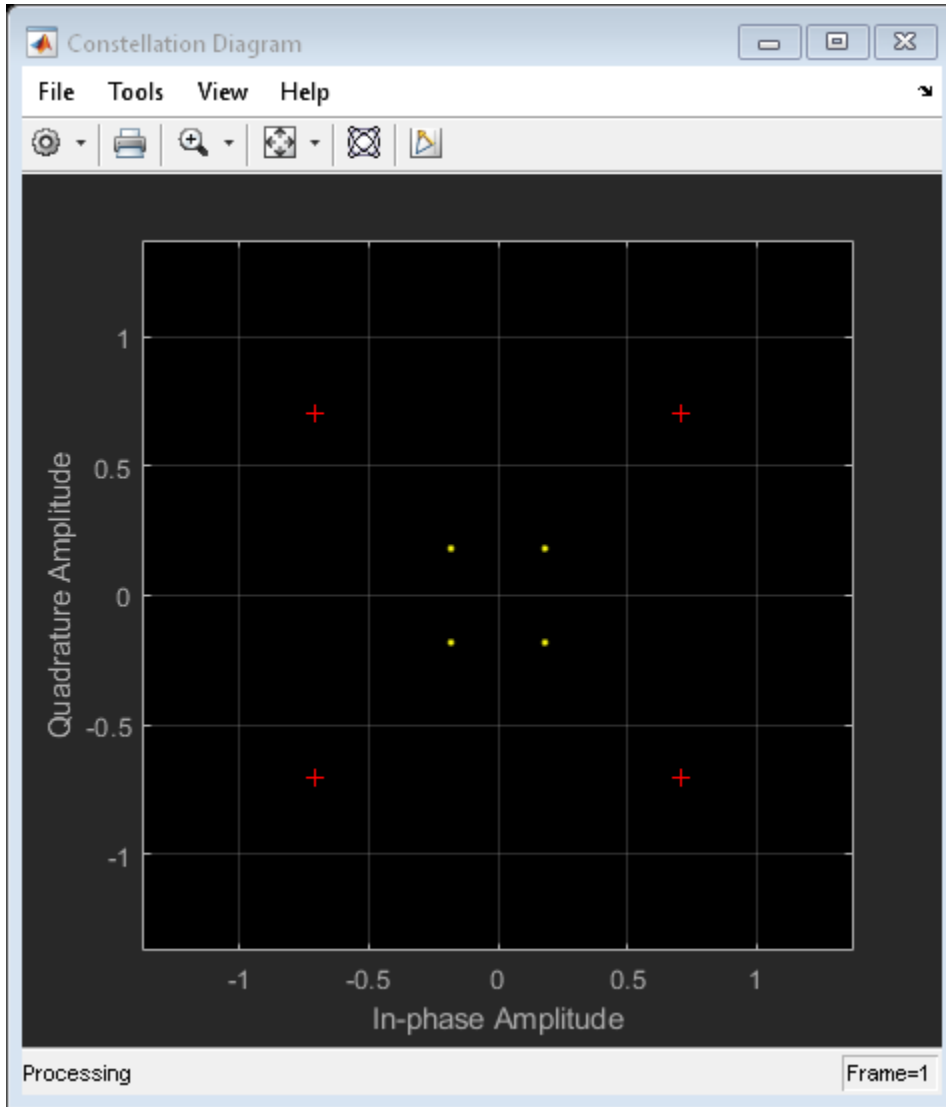
Alternately, you can use `comm.ConstellationDiagram`, specifying the number of samples per symbol, and if needed the timing offset. Also, using `comm.ConstellationDiagram` the reference constellation can be shown.

Create a constellation diagram and set the `SamplesPerSymbol` property to the oversampling factor of the signal. Specify the constellation diagram so that it only displays the last 100 samples. This hides the zero values output by the RRC filter for the first `FilterSpanInSymbols` samples.

```
constDiagram = comm.ConstellationDiagram('SamplesPerSymbol',sps, ...  
    'SymbolsToDisplaySource','Property','SymbolsToDisplay',100);
```

Display the constellation diagram of the transmitted signal.

```
constDiagram(txSig)
```



To match the signal to its reference constellation, normalize the filter by setting its gain to the square root of the `OutputSamplesPerSymbol` property. This was previously specified as `sps`. The filter gain is nontunable so the object must be released prior to changing this value.

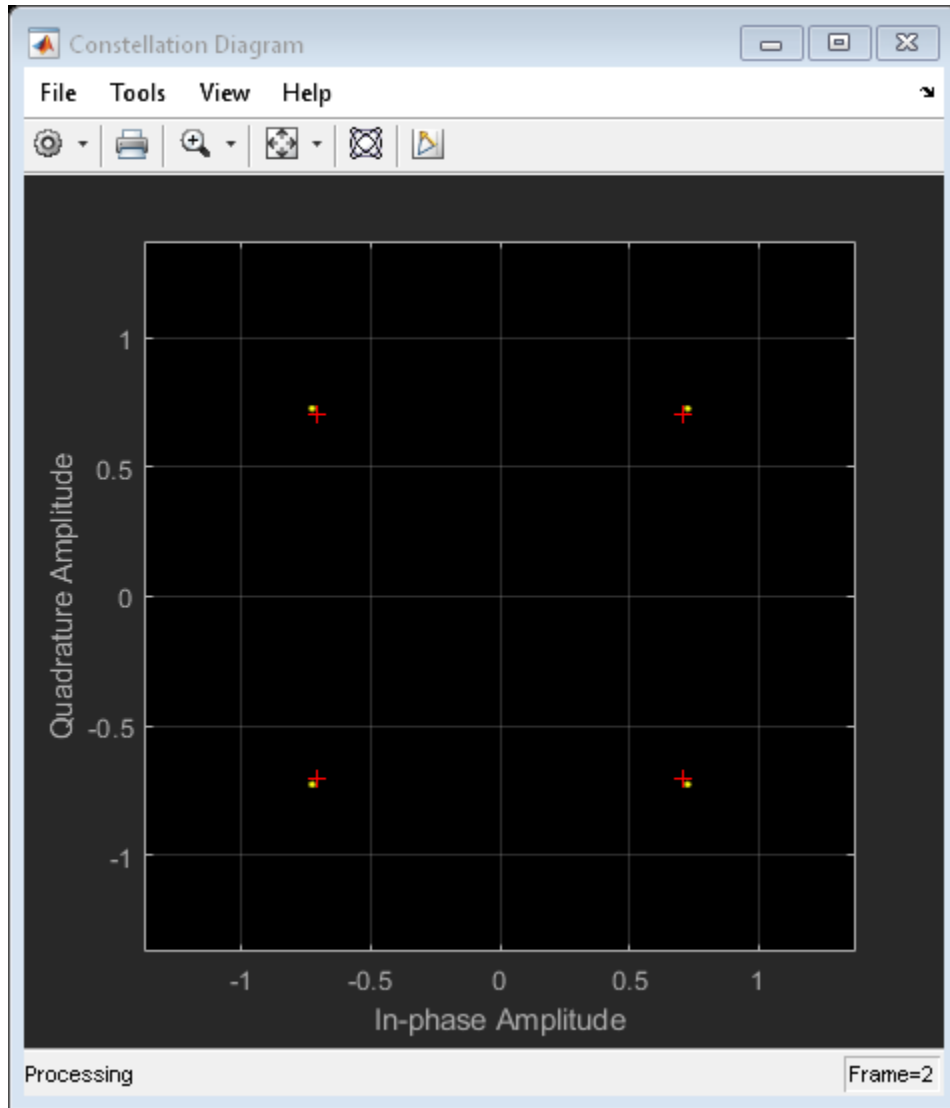
```
release(txfilter)
txfilter.Gain = sqrt(sps);
```

Pass the modulated signal through the normalized filter.

```
txSig = txfilter(modData);
```

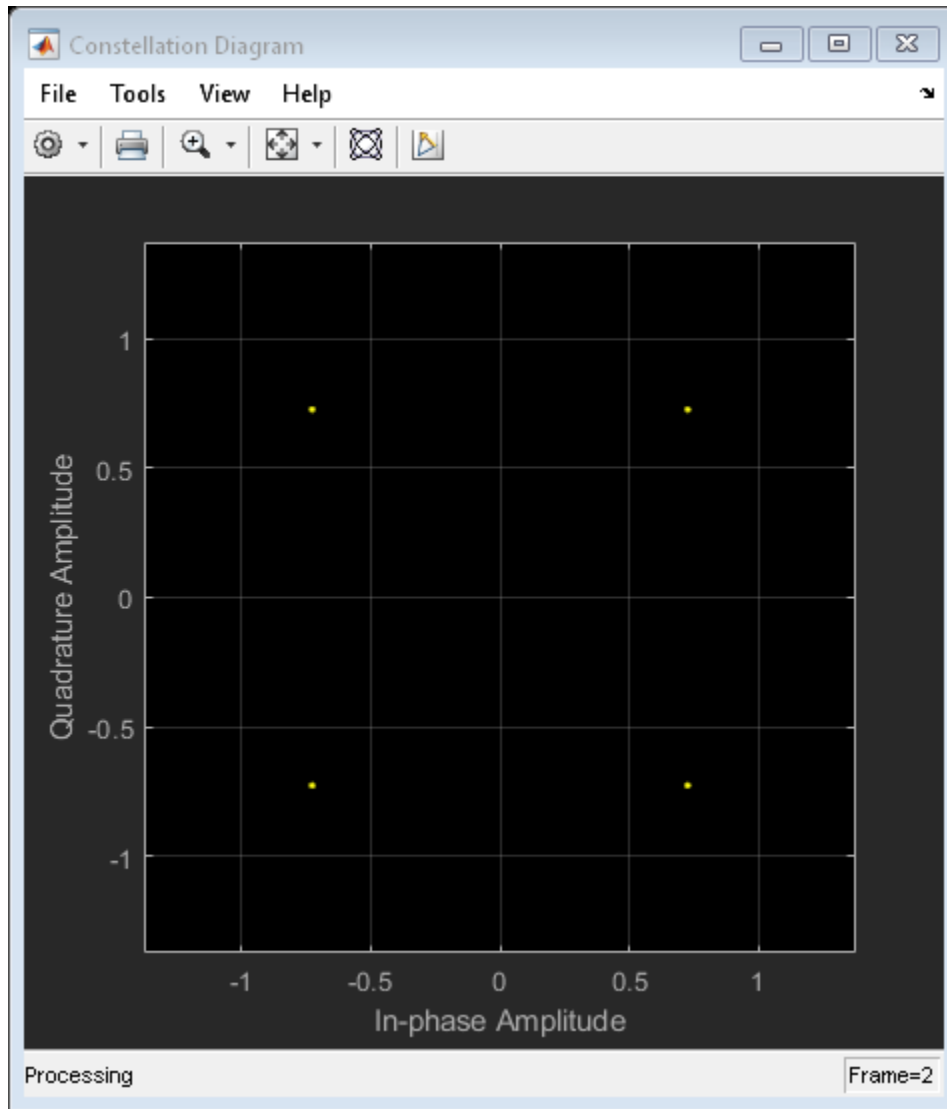
Display the constellation diagram of the normalized signal. The data points and reference constellation nearly overlap.

```
constDiagram(txSig)
```



To view the transmitted signal more clearly, hide the reference constellation by setting the `ShowReferenceConstellation` property to `false`.

```
constDiagram.ShowReferenceConstellation = false;
```

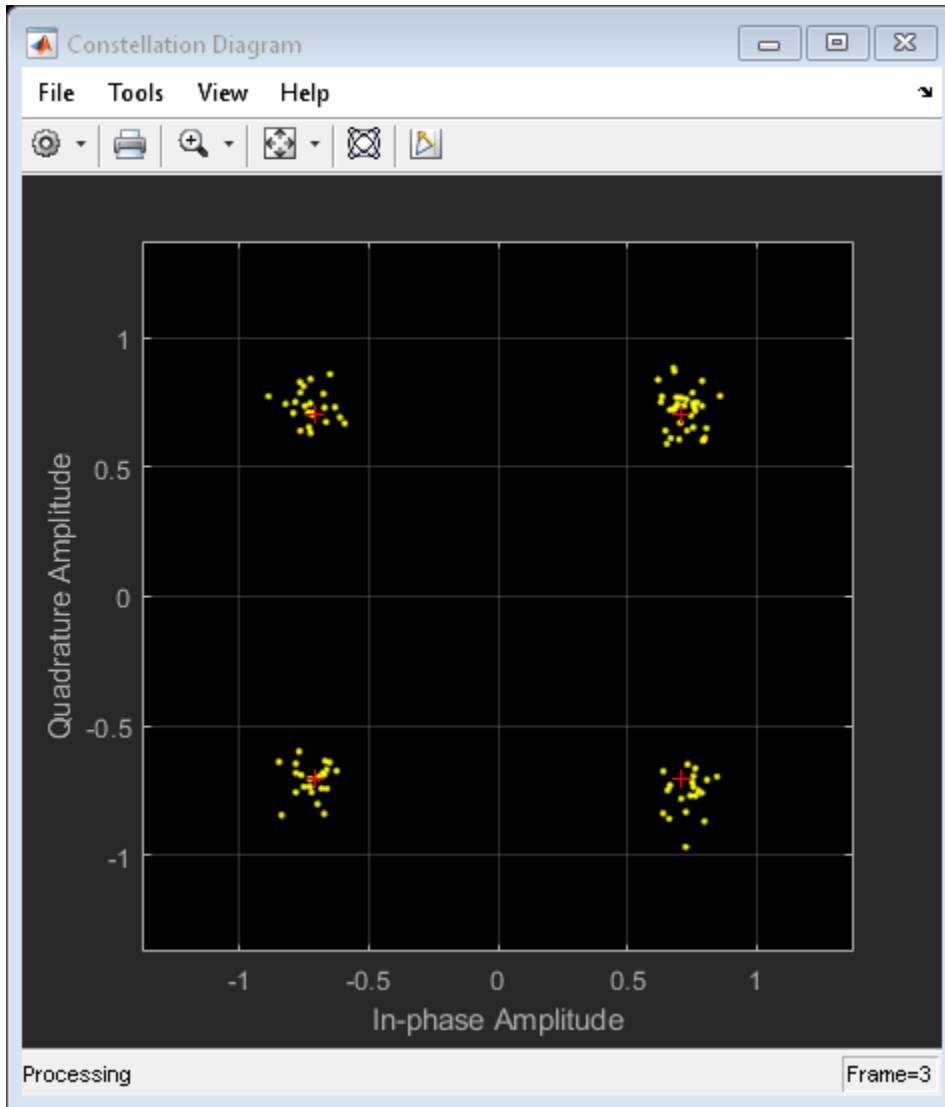


Create a noisy signal by passing `txSig` through an AWGN channel.

```
rxSig = awgn(txSig,20,'measured');
```

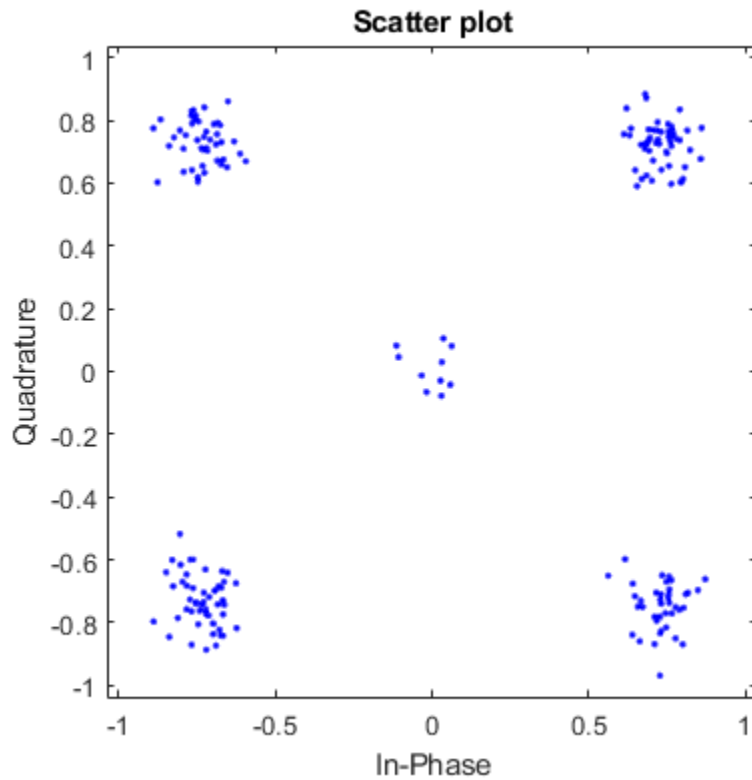
Show the reference constellation and plot the received signal constellation.


```
constDiagram.ShowReferenceConstellation = true;  
constDiagram(rxSig)
```



You can also use `scatterplot` to view this noisy signal but there is no built in option to add the reference constellation using `scatterplot`.

```
scatterplot(rxSig, sps)
```



See Also

“Constellation Visualization” on page 17-2

Channel Visualization

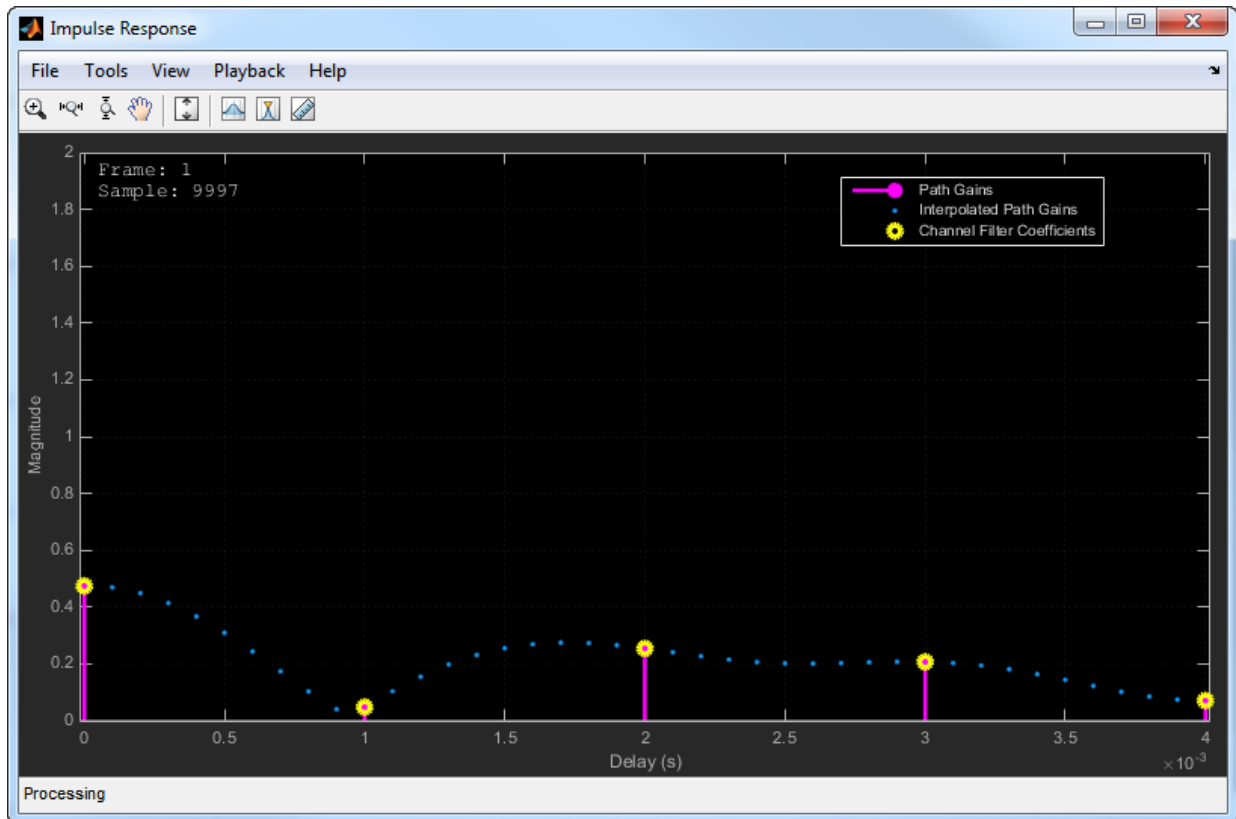
Communications Toolbox software provides a plotting function that helps you visualize the characteristics of a fading channel using a GUI. See Fading Channels on page 14-10 for a description of fading channels, objects, and blocks.

Select the desired visualization setting to plot the Impulse Response on page 17-29, Frequency Response on page 17-33, or Doppler Spectrum on page 17-35 of the channel.

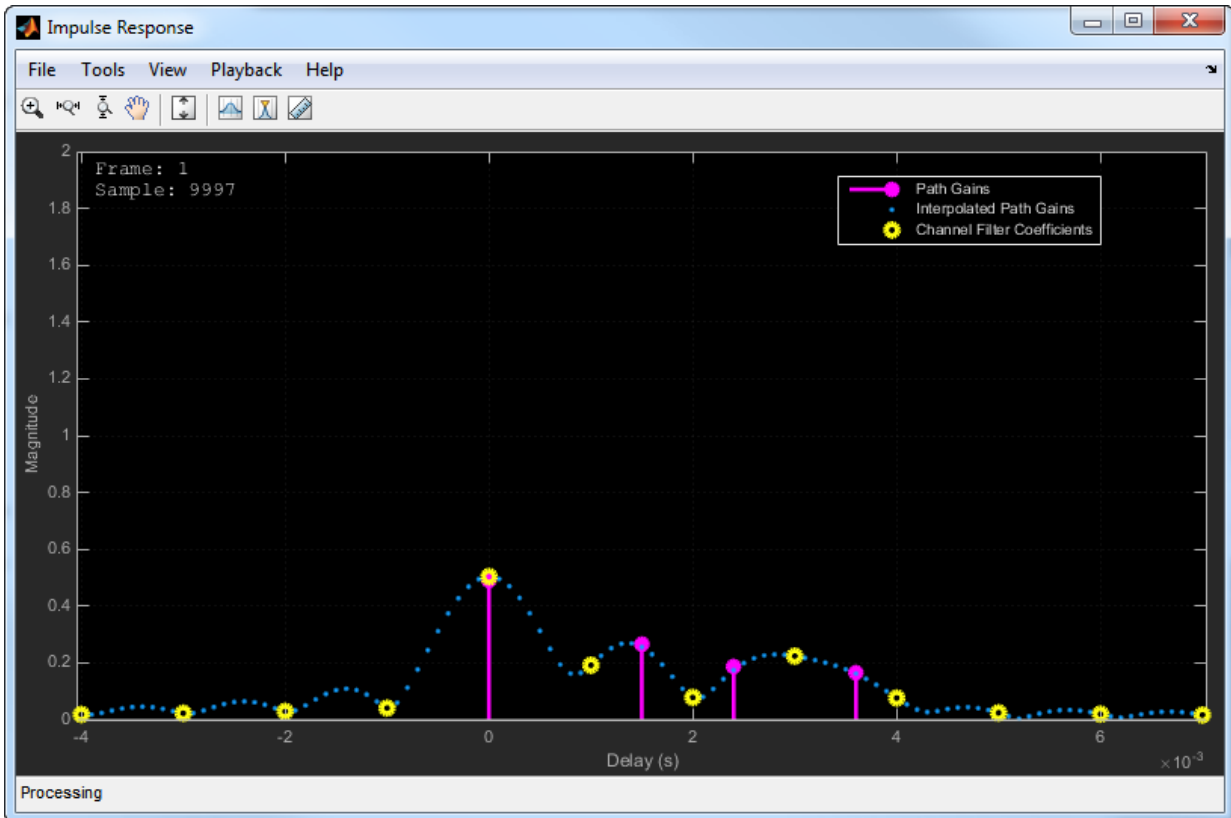
Impulse Response

The Impulse Response plot displays the path gains, the channel filter coefficients, and the interpolated path gains of the channel. The path gains shown in magenta occur at time instances that correspond to the specified path delays. These might not be aligned with the input sampling time. The channel filter coefficients shown in yellow are used to model the channel. They are interpolated from the actual path gains and are aligned with the input sampling time. When the path gains align with the sampling time, they overlap the filter coefficients. Sinc interpolation is used to generate the blue points that appear between the channel filter coefficients. These points are used solely for display purposes and not used in subsequent channel filtering. For a flat fading channel (one path), the sinc interpolation points are not displayed. For all impulse response plots, the frame and sample numbers appear in the upper left corner of the display. The Impulse Response plot shares the same toolbar and menus as the System object it was based on, `dsp.ArrayPlot`.

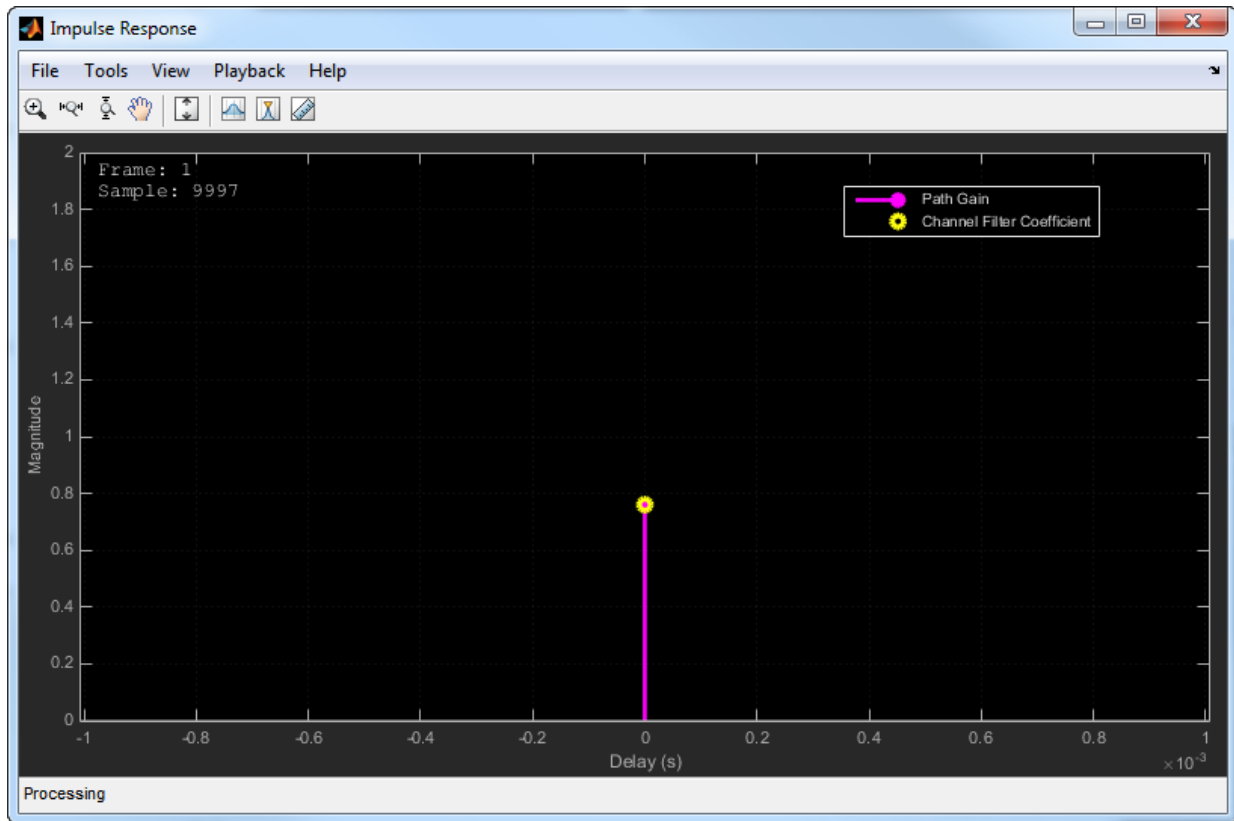
The figure shows the impulse response of a channel in which the path gains align with the sample time. The path gains and channel filter coefficients overlap.



The next plot shows when the specified path gains are not aligned with the sample rate. Observe that the path gains and the channel filter coefficients do not overlap and that the filter coefficients are equally distributed.



The impulse response for a frequency-flat channel is shown next. Because the channel is represented by a single coefficient, no interpolation is done, and the interpolated path gains do not appear.



Note

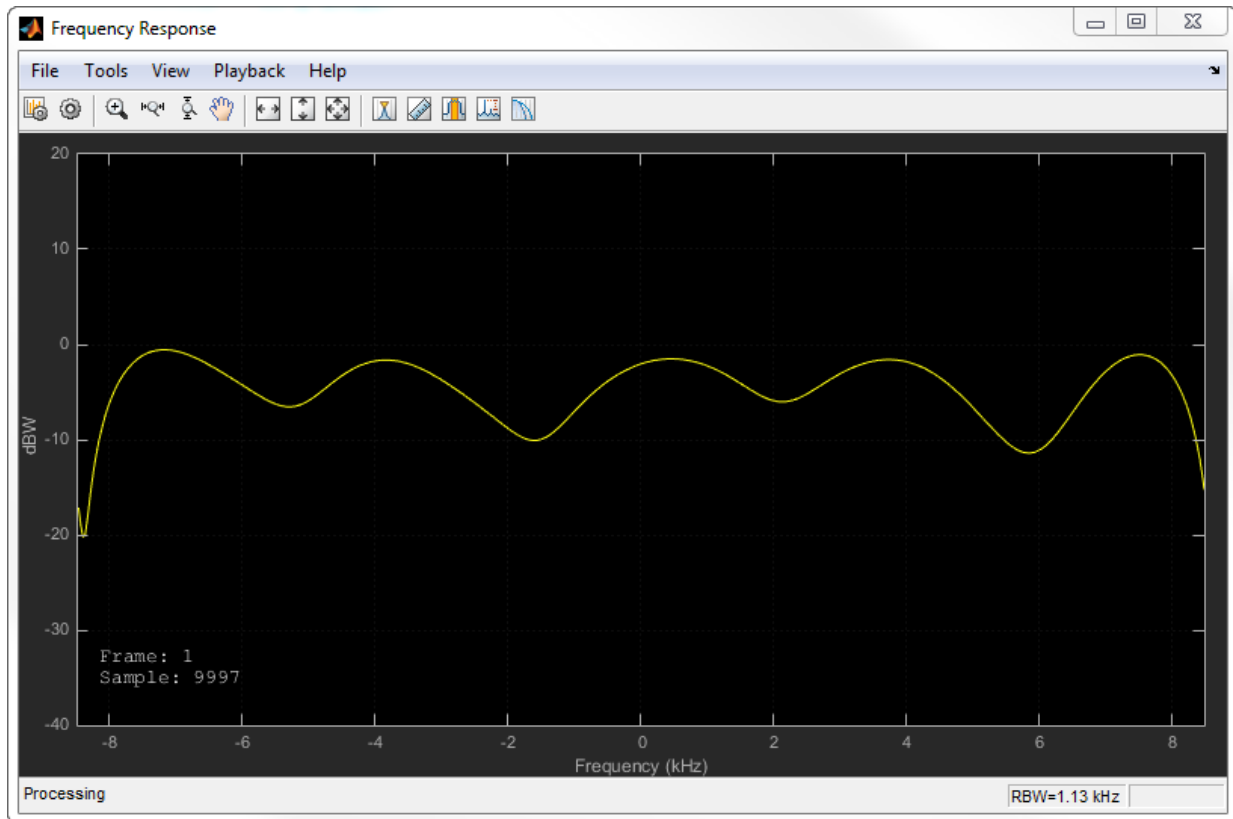
- The displayed and specified path gain locations can differ by as much as 5% of the input sample time.
- For MIMO, when the antenna selection property is set to any value other than `Off` and the specified transmit-receive pair is not selected for the current frame transmission, nothing will be displayed. Antenna selection is not applicable for SISO.
- The visualization display speed is controlled by the combination of the samples to display property and the **Playback > Reduce Updates to Improve Performance** menu item. Reducing the percentage of samples to display and enabling reduced updates speeds up the rendering of the impulse response.

- After the Impulse Response plots are manually closed, the channel model executes at its normal speed.
 - Code generation is available only when the visualization property is set to Off.
-

Frequency Response

The Frequency Response plot displays the channel spectrum by taking a discrete Fourier transform of the channel filter coefficients. For the MIMO case, this transform is performed for the specified transmit-receive antenna pair. The Frequency Response plot shares the same toolbar and menus as the System object it was based on, `dsp.SpectrumAnalyzer`. The default settings use a rectangular window. The window length is set based on the channel model configuration. Use the **View > Spectrum Settings** menu to change property values from their default settings.

The frequency response plot for a frequency-selective channel is shown.

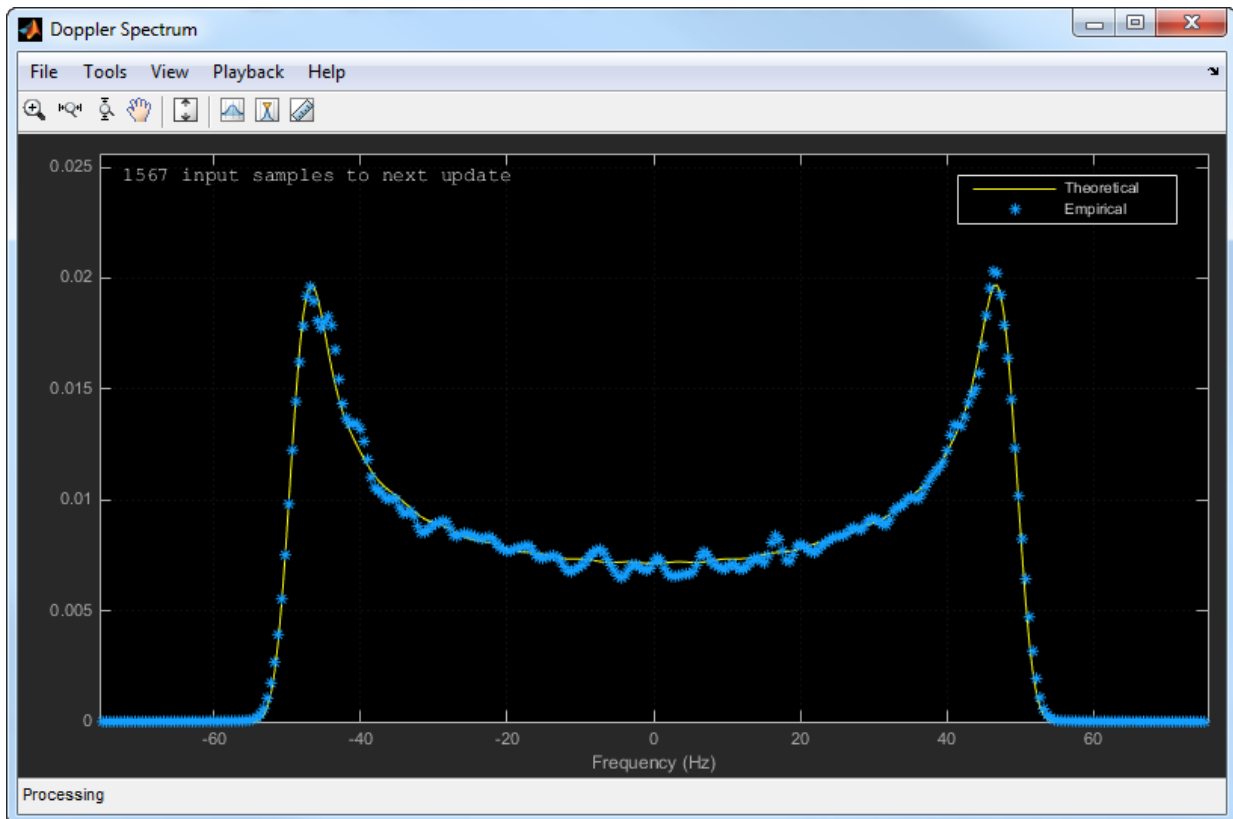


Note

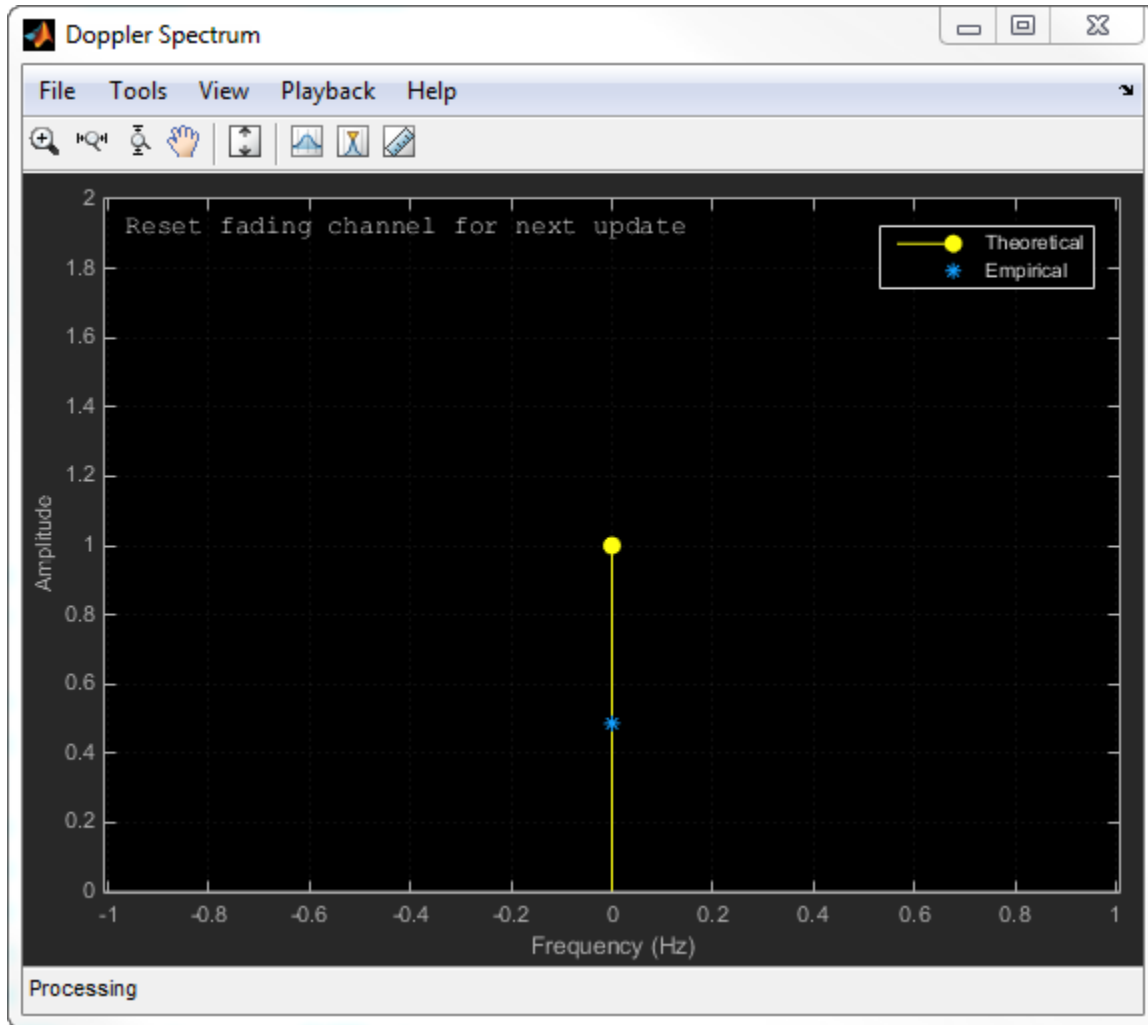
- The visualization display speed is controlled by the combination of the samples to display property and the **Reduce Plot Rate to Improve Performance** menu item. Reducing the percentage of samples to display and enabling reduced updates speeds up the rendering of the frequency response.
- After the Frequency Response plots are manually closed, the channel model executes at its normal speed.
- Code generation is available only when the visualization property is set to Off.

Doppler Spectrum

The Doppler Spectrum plot displays both the theoretical Doppler spectrum and the empirically determined data points. The theoretical data is displayed as a yellow line for nonstatic channels and as a yellow point for static channels. The empirical data is shown in blue. When the internal buffer is completely filled with filtered Gaussian samples, the empirical plot is updated. The empirical plot is the running mean of the spectrum calculated from each full buffer. For nonstatic channels, the number of input samples needed before the next update is displayed in the upper-left corner. The samples needed is a function of the sample rate and the maximum Doppler shift. The Doppler Spectrum plot shares the same toolbar and menus as the System object it was based on, `dsp.ArrayPlot`.



For static channels, the text `Reset fading channel for next update` is displayed.



Note

- After the Doppler Spectrum plots are manually closed, the channel model executes at its normal speed.

- Code generation is available only when the visualization property is Off.
-

See Also

Blocks

MIMO Fading Channel | SISO Fading Channel

Objects

`comm.MIMOChannel`

Visualize RF Impairments

Apply various RF impairments to a QAM signal. Observe the effects by using constellation diagrams, time-varying error vector magnitude (EVM) plots, and spectrum plots. Estimate the equivalent signal-to-noise ratio (SNR).

Initialization

Set the sample rate, modulation order, and SNR. Calculate the reference constellation points.

```
fs = 1000;  
M = 16;  
snrdB = 30;  
refConst = qammod(0:M-1,M, 'UnitAveragePower', true);
```

Create constellation diagram and time scope objects to visualize the impairment effects.

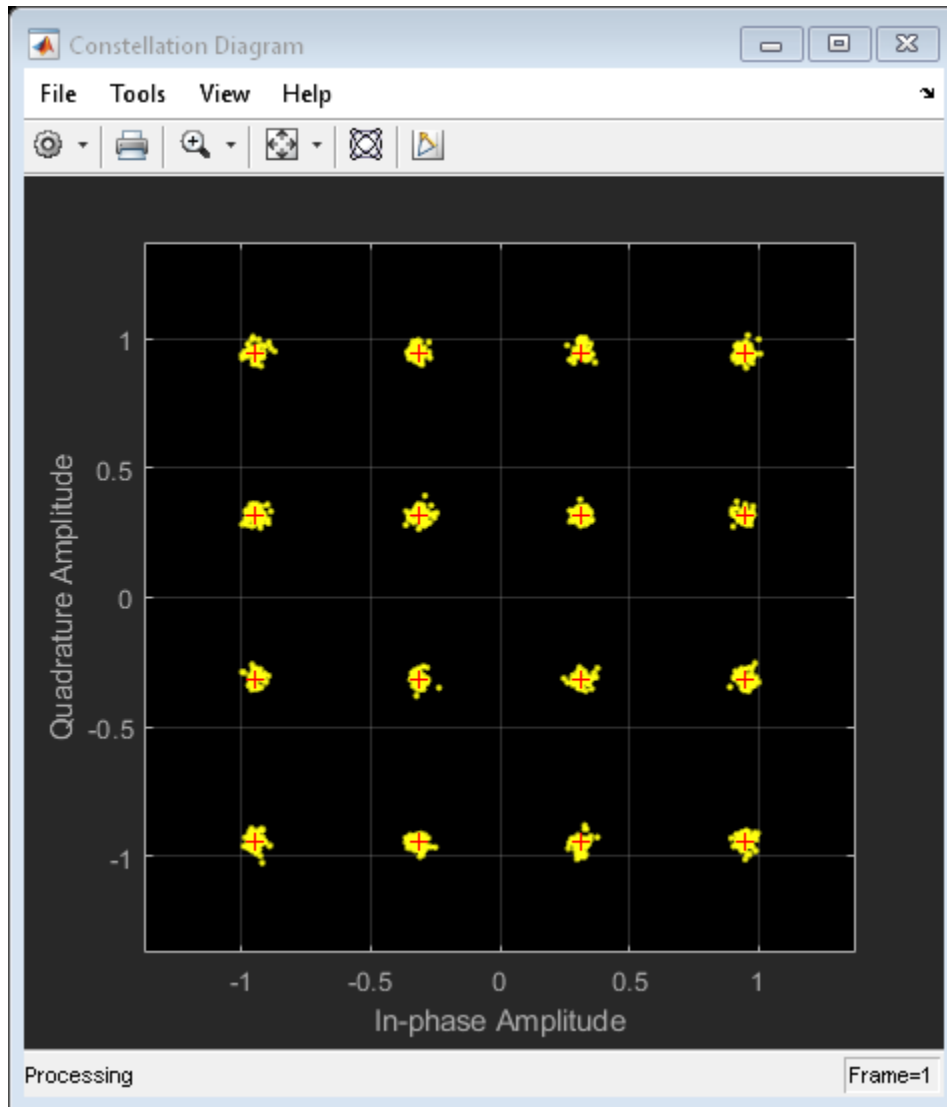
```
constDiagram = comm.ConstellationDiagram('ReferenceConstellation', refConst);  
timeScope = dsp.TimeScope('YLimits', [0 40], 'SampleRate', fs, 'TimeSpan', 1, ...  
    'ShowGrid', true, 'YLabel', 'EVM (%)');
```

White Noise

Generate a 16-QAM signal, and pass it through an AWGN channel. Plot its constellation.

```
data = randi([0 M-1], 1000, 1);  
modSig = qammod(data, M, 'UnitAveragePower', true);  
noisySig = awgn(modSig, snrdB);
```

```
constDiagram(noisySig)
```



Estimate the EVM of the noisy signal from the reference constellation points.

```
evm = comm.EVM('ReferenceSignalSource','Estimated from reference constellation', ...
    'ReferenceConstellation',refConst, ...
    'Normalization','Average constellation power');
```

```
rmsEVM = evm(noisySig)
```

```
rmsEVM =  
    3.1768
```

The modulation error rate (MER) closely corresponds to the SNR. Create an MER object, and estimate the SNR.

```
mer = comm.MER('ReferenceSignalSource','Estimated from reference constellation', ...  
              'ReferenceConstellation',refConst);  
snrEst = mer(noisySig)
```

```
snrEst =  
    30.1071
```

The estimate is quite close to the specified SNR of 30 dB.

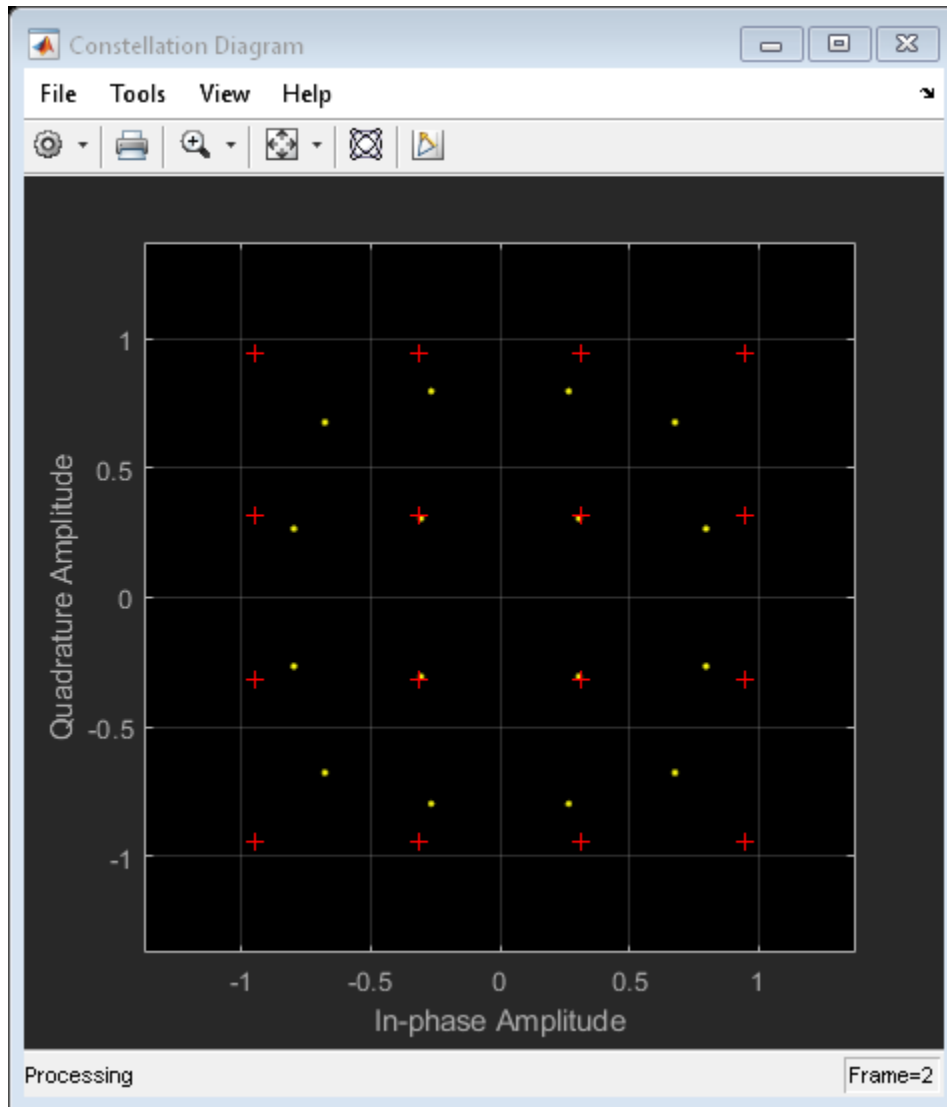
Amplifier Distortion

Create an amplifier using the memoryless nonlinearity object.

```
amp = comm.MemorylessNonlinearity('IIP3',38,'AMPMConversion',0);
```

Pass the modulated signal through the nonlinear amplifier, and plot its constellation diagram.

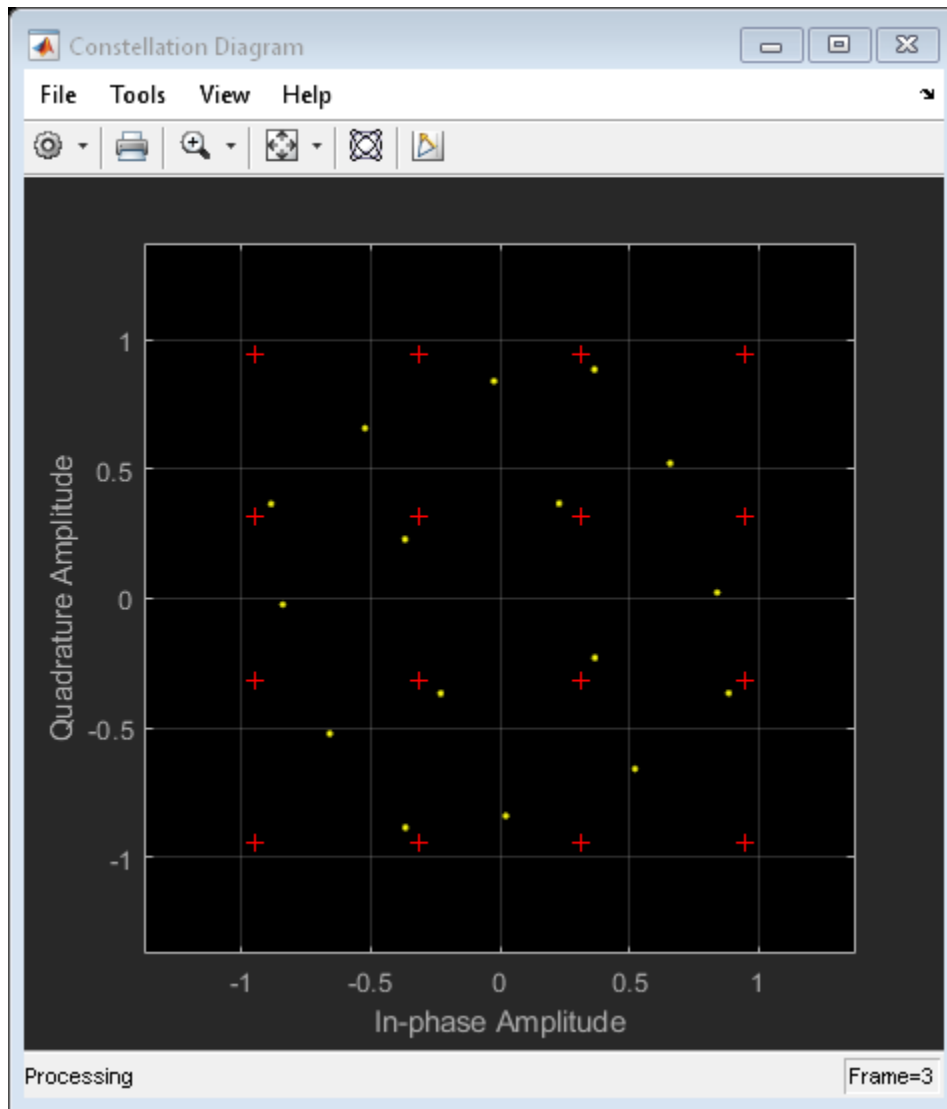
```
txSig = amp(modSig);  
constDiagram(txSig)
```



The corner points of the constellation have moved toward the origin due to amplifier gain compression.

Introduce a small AM/PM conversion, and display the received signal constellation.

```
amp.AMPMConversion = 1;  
txSig = amp(modSig);  
constDiagram(txSig)
```

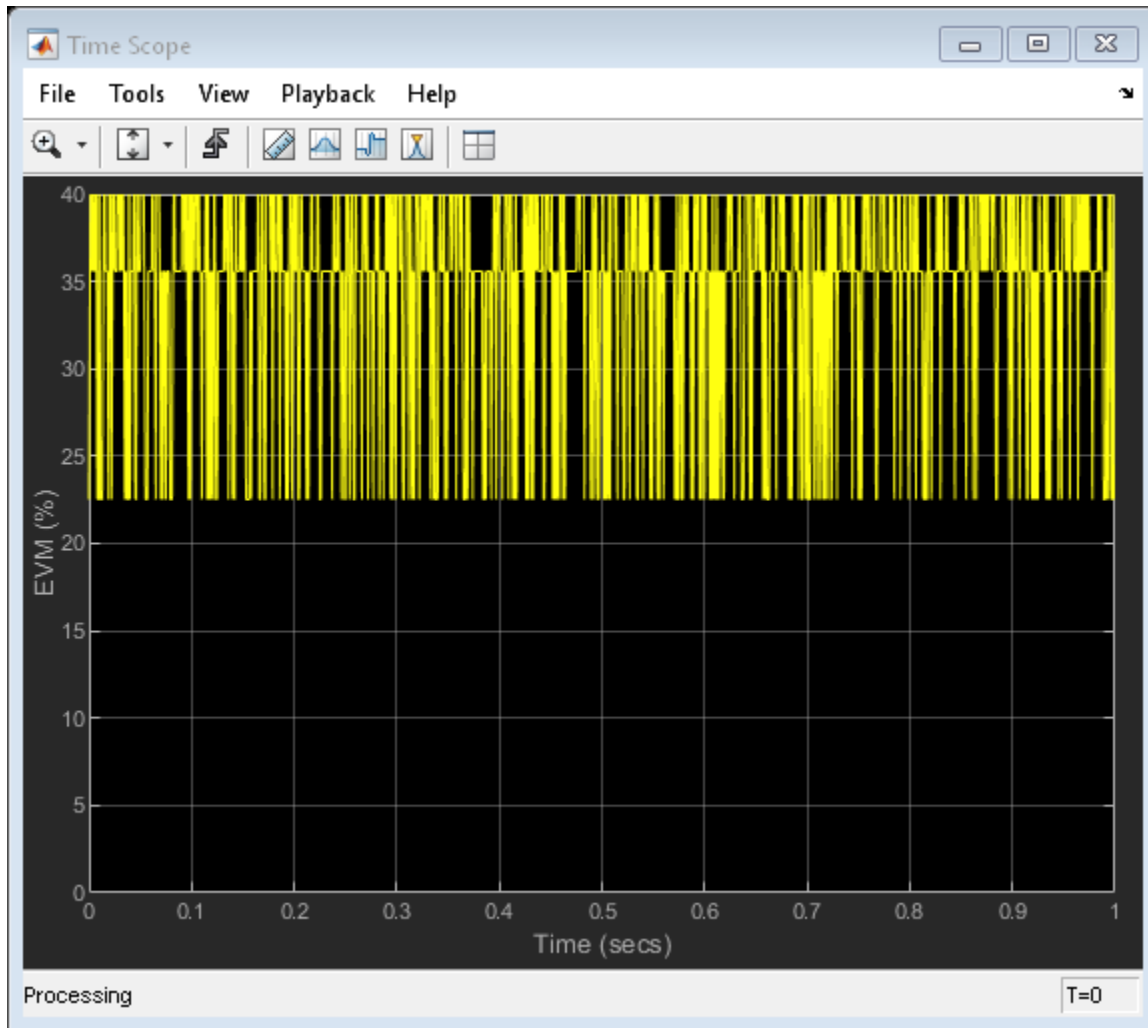


The constellation has rotated due to the AM/PM conversion. To compute the time-varying EVM, release the EVM object and set the `AveragingDimensions` property to 2. To estimate the EVM against an input signal, omit the `ReferenceSignalSource` property definition. This method produces more accurate results.

```
evm = comm.EVM('AveragingDimensions',2);  
evmTime = evm(modSig,txSig);
```

Plot the time-varying EVM of the distorted signal.

```
timeScope(evmTime)
```



Compute the RMS EVM.

```
evmRMS = sqrt(mean(evmTime.^2))
```

```
evmRMS =
```

35.5919

Compute the MER.

```
mer = comm.MER;
snrEst = mer(modSig,txSig)
```

```
snrEst =
```

8.1392

The SNR (≈ 8 dB) is reduced from its initial value (∞) due to amplifier distortion.

Specify input power levels ranging from 0 to 40 dBm. Convert those levels to their linear equivalent in W. Initialize the output power vector.

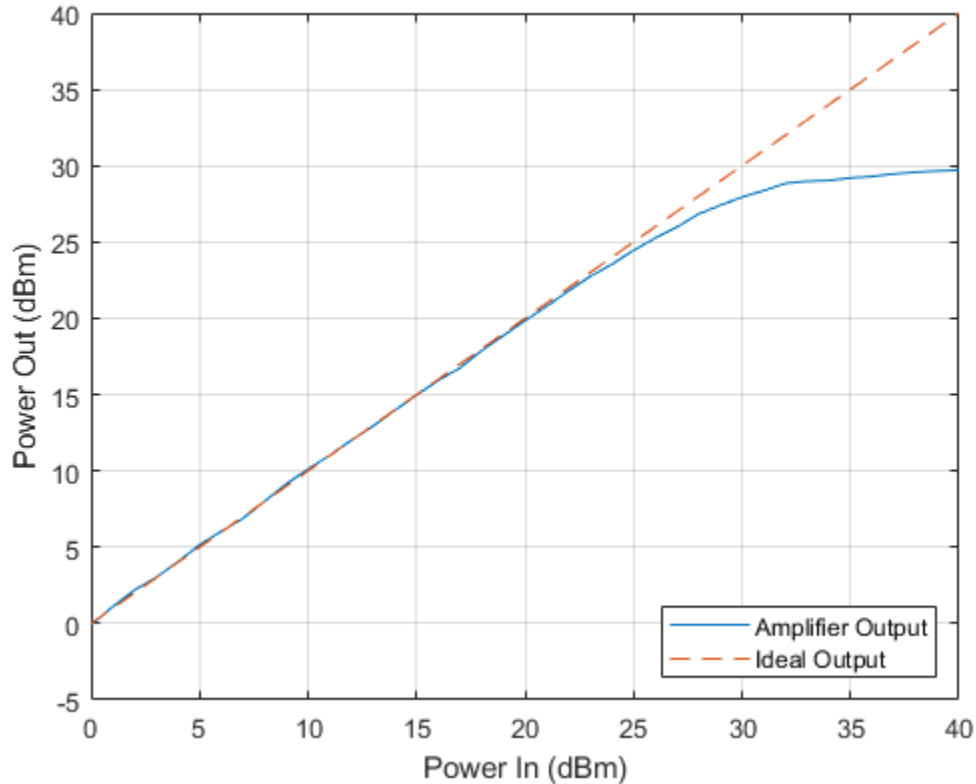
```
powerIn = 0:40;
pin = 10.^((powerIn-30)/10);
powerOut = zeros(length(powerIn),1);
```

Measure the amplifier output power for the range of input power levels.

```
for k = 1:length(powerIn)
    data = randi([0 15],1000,1);
    txSig = qammod(data,16,'UnitAveragePower',true)*sqrt(pin(k));
    ampSig = amp(txSig);
    powerOut(k) = 10*log10(var(ampSig))+30;
end
```

Plot the power output versus power input curve.

```
figure
plot(powerIn,powerOut,powerIn,powerIn,'--')
legend('Amplifier Output','Ideal Output','location','se')
xlabel('Power In (dBm)')
ylabel('Power Out (dBm)')
grid
```



The output power levels off at 30 dBm. The amplifier exhibits nonlinear behavior for input power levels greater than 25 dBm.

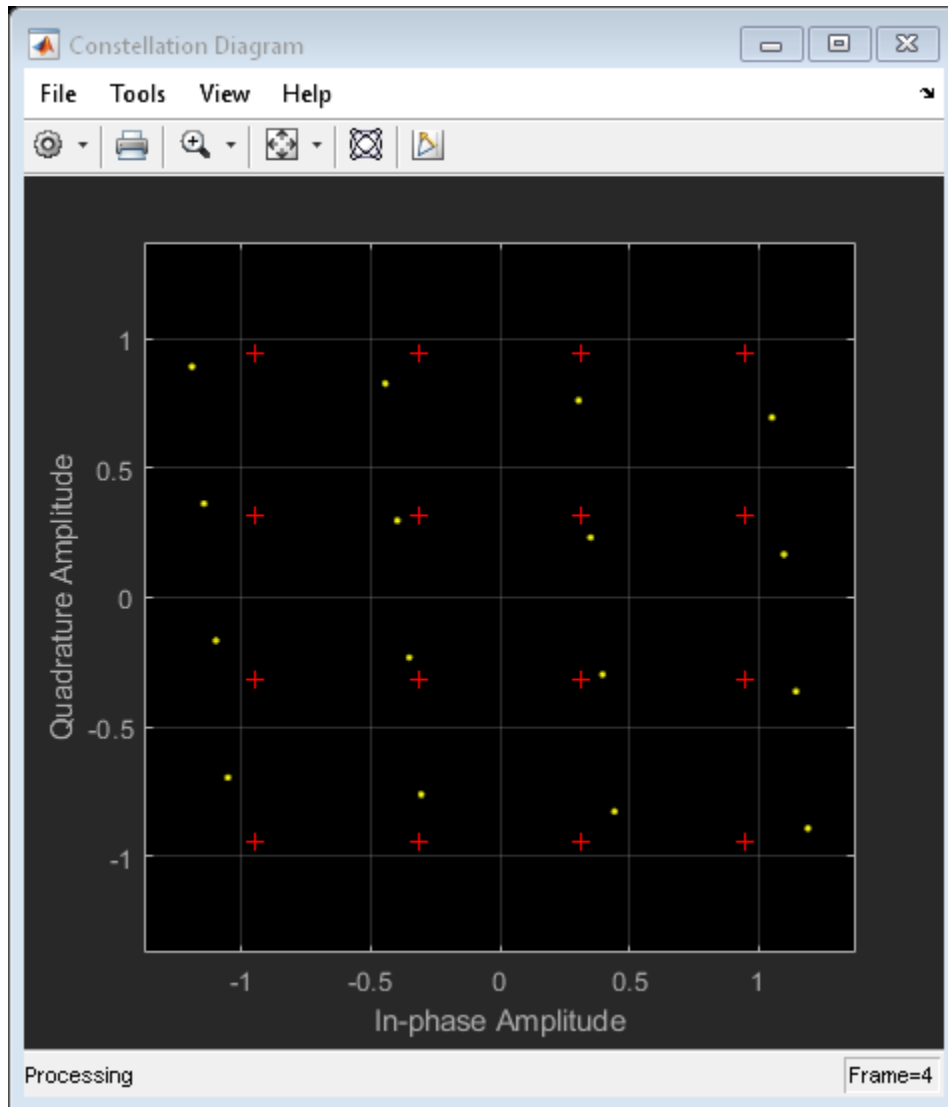
I/Q Imbalance

Apply an amplitude and phase imbalance to the modulated signal.

```
ampImb = 3;
phImb = 10;
gainI = 10.^(0.5*ampImb/20);
gainQ = 10.^(-0.5*ampImb/20);
imbI = real(modSig)*gainI*exp(-0.5i*phImb*pi/180);
imbQ = imag(modSig)*gainQ*exp(1i*(pi/2 + 0.5*phImb*pi/180));
rxSig = imbI + imbQ;
```

Plot the received constellation.

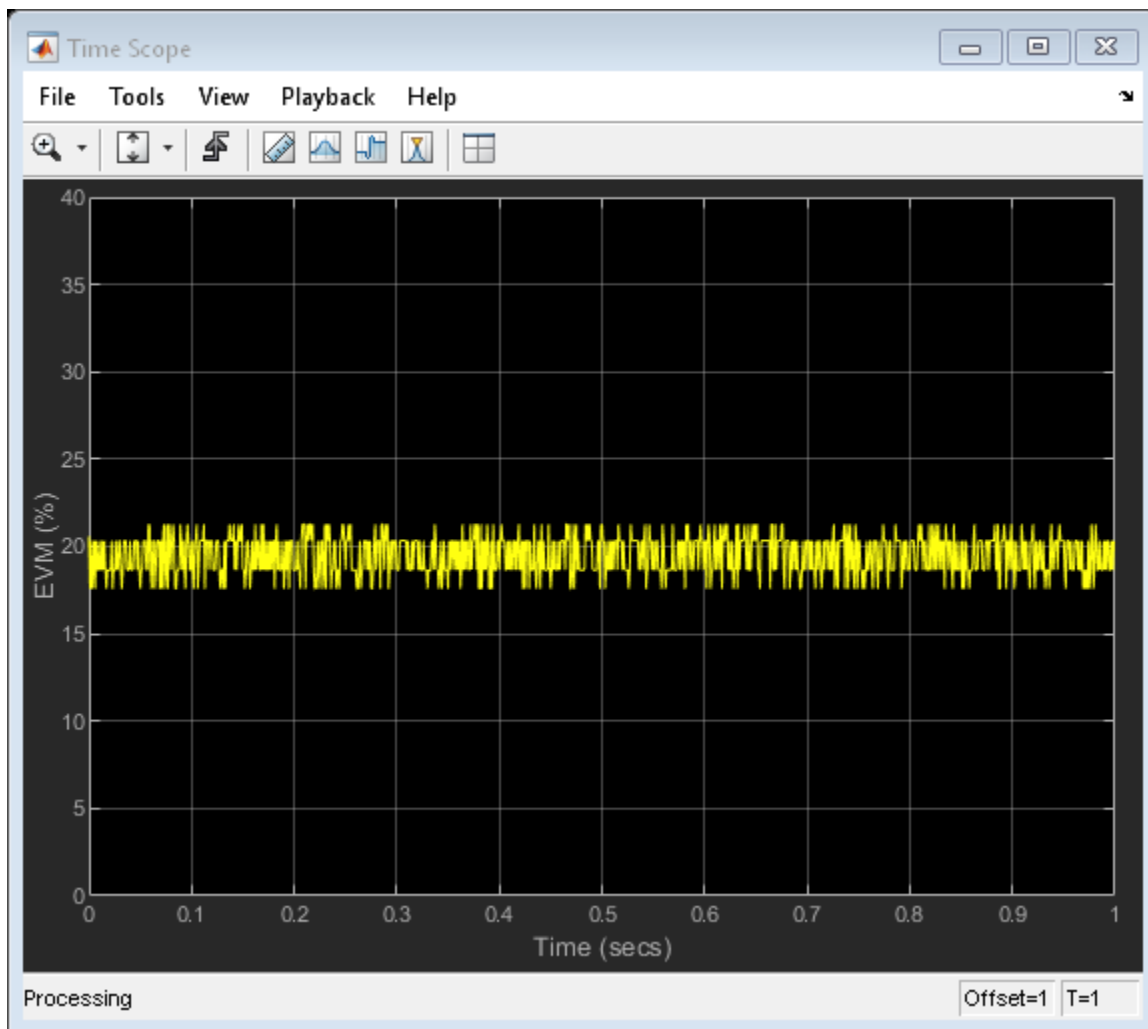
```
constDiagram(rxSig)
```



The magnitude and phase of the constellation has changed as a result of the I/Q imbalance.

Calculate and plot the time-varying EVM.

```
evmTime = evm(modSig, rxSig);  
timeScope(evmTime)
```



The EVM exhibits a behavior that is similar to that experienced with a nonlinear amplifier though the variance is smaller.

Create a 100 Hz sine wave having a 1000 Hz sample rate.

```
sinewave = dsp.SinWave('Frequency',100,'SampleRate',1000, ...  
    'SamplesPerFrame',1e4,'ComplexOutput',true);
```

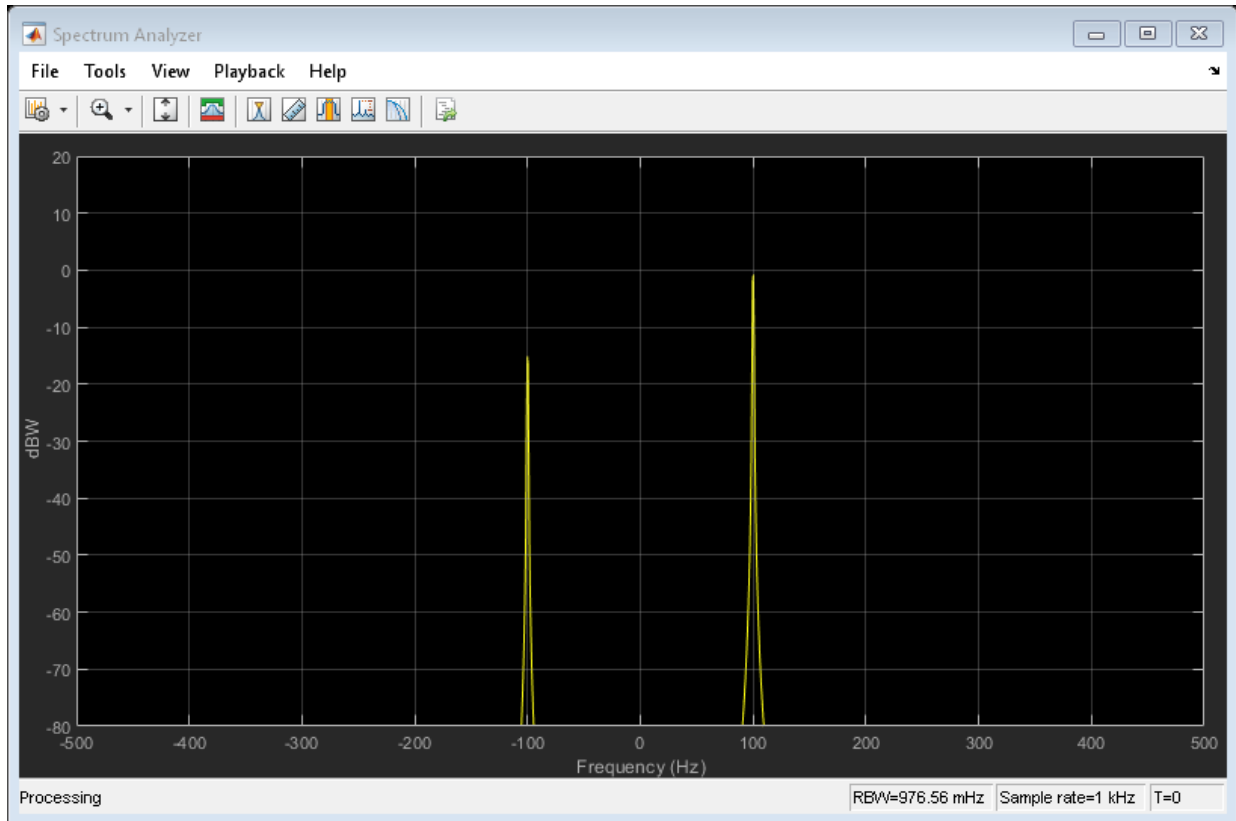
```
x = sinewave();
```

Apply the same 3 dB and 10 degree I/Q imbalance.

```
ampImb = 3;  
phImb = 10;  
gainI = 10.^(0.5*ampImb/20);  
gainQ = 10.^(-0.5*ampImb/20);  
imbI = real(x)*gainI*exp(-0.5i*phImb*pi/180);  
imbQ = imag(x)*gainQ*exp(1i*(pi/2 + 0.5*phImb*pi/180));  
y = imbI + imbQ;
```

Plot the spectrum of the imbalanced signal.

```
spectrum = dsp.SpectrumAnalyzer('SampleRate',1000,'PowerUnits','dBW');  
spectrum(y)
```

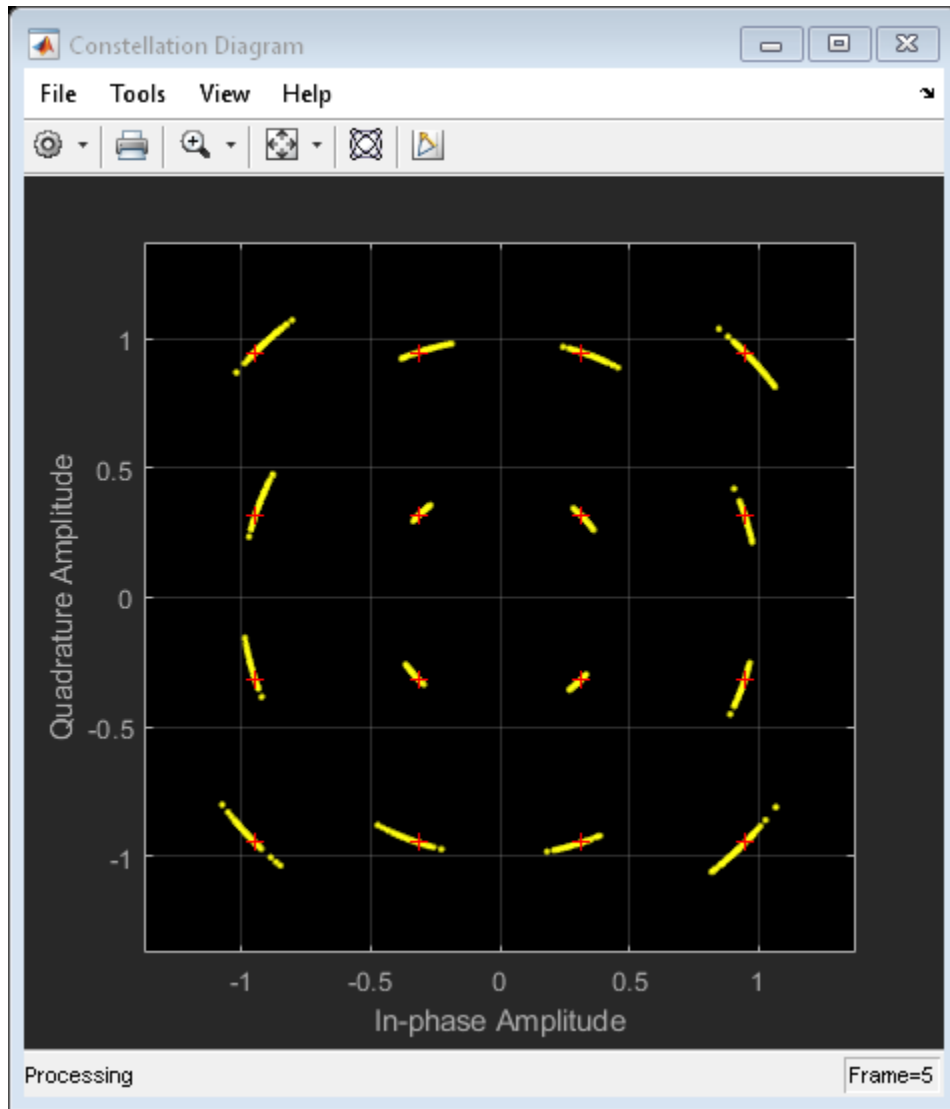


The I/Q imbalance introduces a second tone at -100 Hz, which is the inverse of the input tone.

Phase Noise

Apply phase noise to the transmitted signal. Plot the resulting constellation diagram.

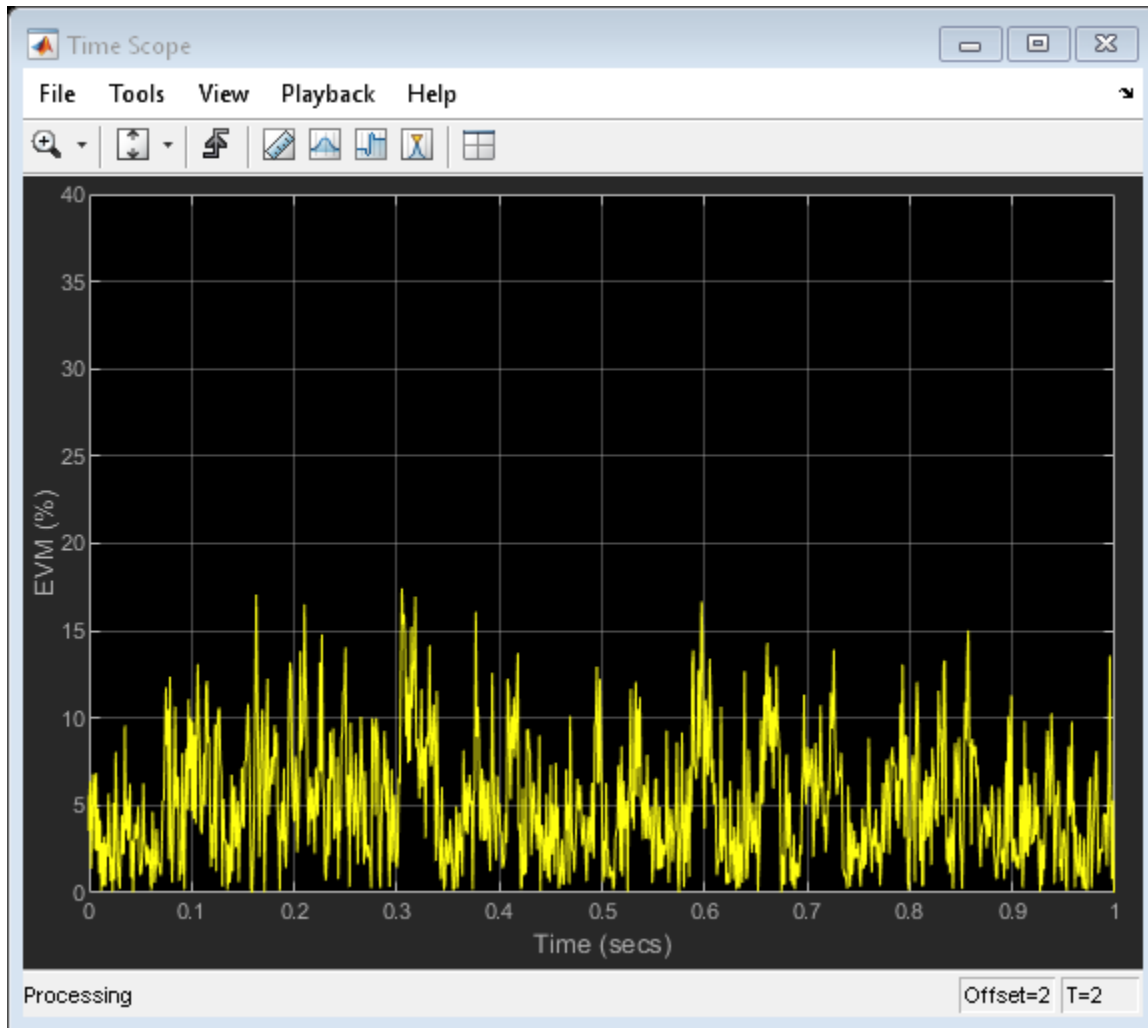
```
pnoise = comm.PhaseNoise('Level',-50,'FrequencyOffset',20,'SampleRate',fs);
pnoiseSig = pnoise(modSig);
constDiagram(pnoiseSig)
```

The phase noise introduces a rotational jitter.

Compute and plot the EVM of the received signal.

```
evmTime = evm(modSig,pnoiseSig);  
timeScope(evmTime)
```



Determine the RMS EVM.

```
evmRMS = sqrt(mean(evmTime.^2))
```

```
evmRMS =
    6.1989
```

Filter Effects

Specify the samples per symbol parameter. Create a pair of raised cosine matched filters.

```
sps = 4;
txfilter = comm.RaisedCosineTransmitFilter('RolloffFactor',0.2,'FilterSpanInSymbols',8,
    'OutputSamplesPerSymbol',sps,'Gain',sqrt(sps));

rxfilter = comm.RaisedCosineReceiveFilter('RolloffFactor',0.2,'FilterSpanInSymbols',8,
    'InputSamplesPerSymbol',sps,'Gain',1/sqrt(sps), ...
    'DecimationFactor',sps);
```

Determine the delay through the matched filters.

```
fltDelay = 0.5*(txfilter.FilterSpanInSymbols + rxfilter.FilterSpanInSymbols);
```

Pass the modulated signal through the matched filters.

```
filtSig = txfilter(modSig);
rxSig = rxfilter(filtSig);
```

To account for the delay through the filters, discard the first fltDelay samples.

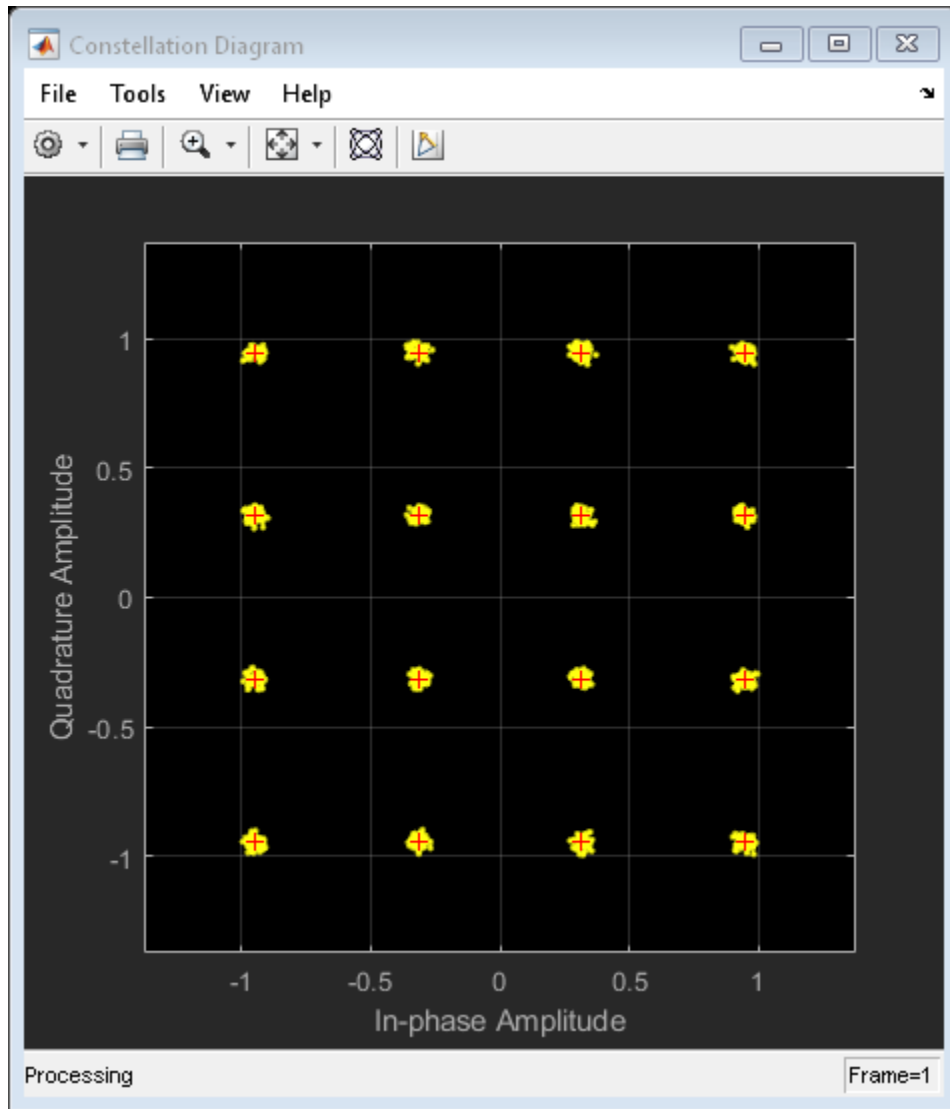
```
rxSig = rxSig(fltDelay+1:end);
```

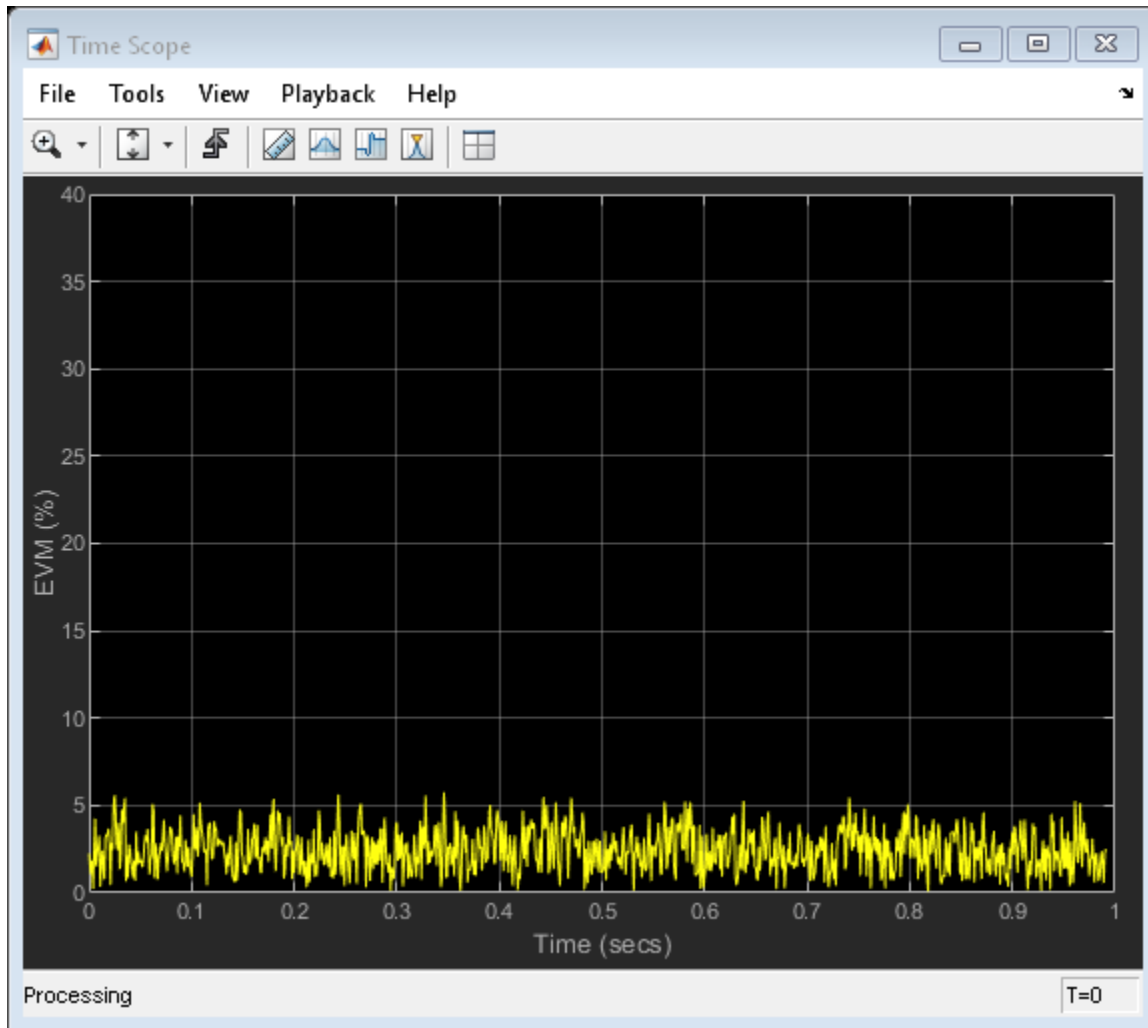
To accommodate the change in the number of received signal samples, create new constellation diagram and time scope objects.

```
constDiagram = comm.ConstellationDiagram('ReferenceConstellation',refConst);
timeScope = dsp.TimeScope('YLimits',[0 40],'SampleRate',fs,'TimeSpan',1, ...
    'ShowGrid',true,'YLabel','EVM (%)');
```

Estimate EVM. Plot the received signal constellation diagram and the time-varying EVM.

```
evm = comm.EVM('ReferenceSignalSource','Estimated from reference constellation', ...
    'ReferenceConstellation',refConst, ...
    'Normalization','Average constellation power','AveragingDimensions',2);
evmTime = evm(rxSig);
constDiagram(rxSig)
timeScope(evmTime)
```





Determine the RMS EVM.

```
evmRMS = sqrt(mean(evmTime.^2))
```

```
evmRMS =
```

2.7199

Determine the equivalent SNR.

```
mer = comm.MER;  
snrEst = mer(modSig(1:end-fltDelay),rxSig)
```

```
snrEst =  
  
31.4603
```

Combined Effects

Combine the effects of the filters, nonlinear amplifier, AWGN, and phase noise. Display the constellation and EVM diagrams.

Create EVM, time scope and constellation diagram objects.

```
evm = comm.EVM('ReferenceSignalSource','Estimated from reference constellation', ...  
              'ReferenceConstellation',refConst, ...  
              'Normalization','Average constellation power','AveragingDimensions',2);  
timeScope = dsp.TimeScope('YLimits',[0 40],'SampleRate',fs,'TimeSpan',1, ...  
                          'ShowGrid',true,'YLabel','EVM (%)');  
constDiagram = comm.ConstellationDiagram('ReferenceConstellation',refConst);
```

Specify the nonlinear amplifier and phase noise objects.

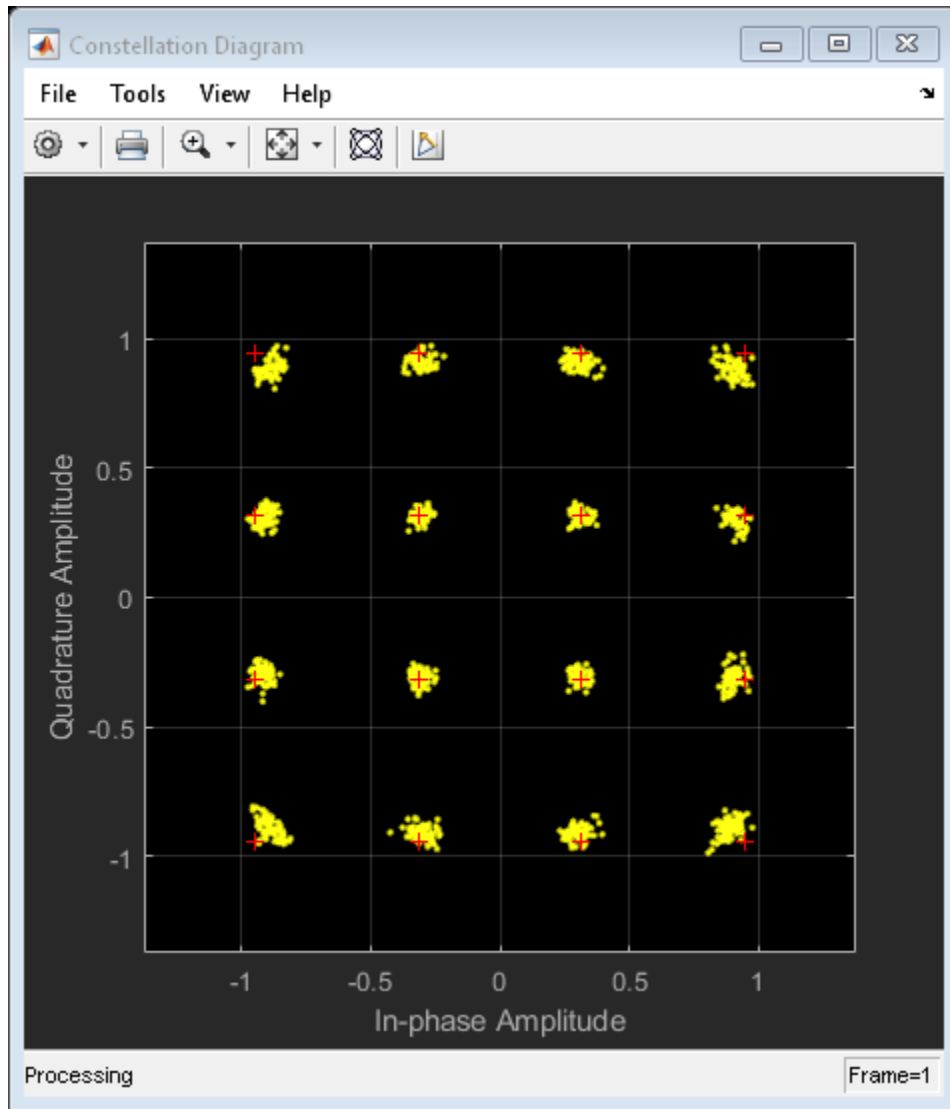
```
amp = comm.MemorylessNonlinearity('IIP3',45,'AMPMConversion',0);  
pnoise = comm.PhaseNoise('Level',-55,'FrequencyOffset',20,'SampleRate',fs);
```

Filter and then amplify the modulated signal.

```
txfiltOut = txfilter(modSig);  
txSig = amp(txfiltOut);
```

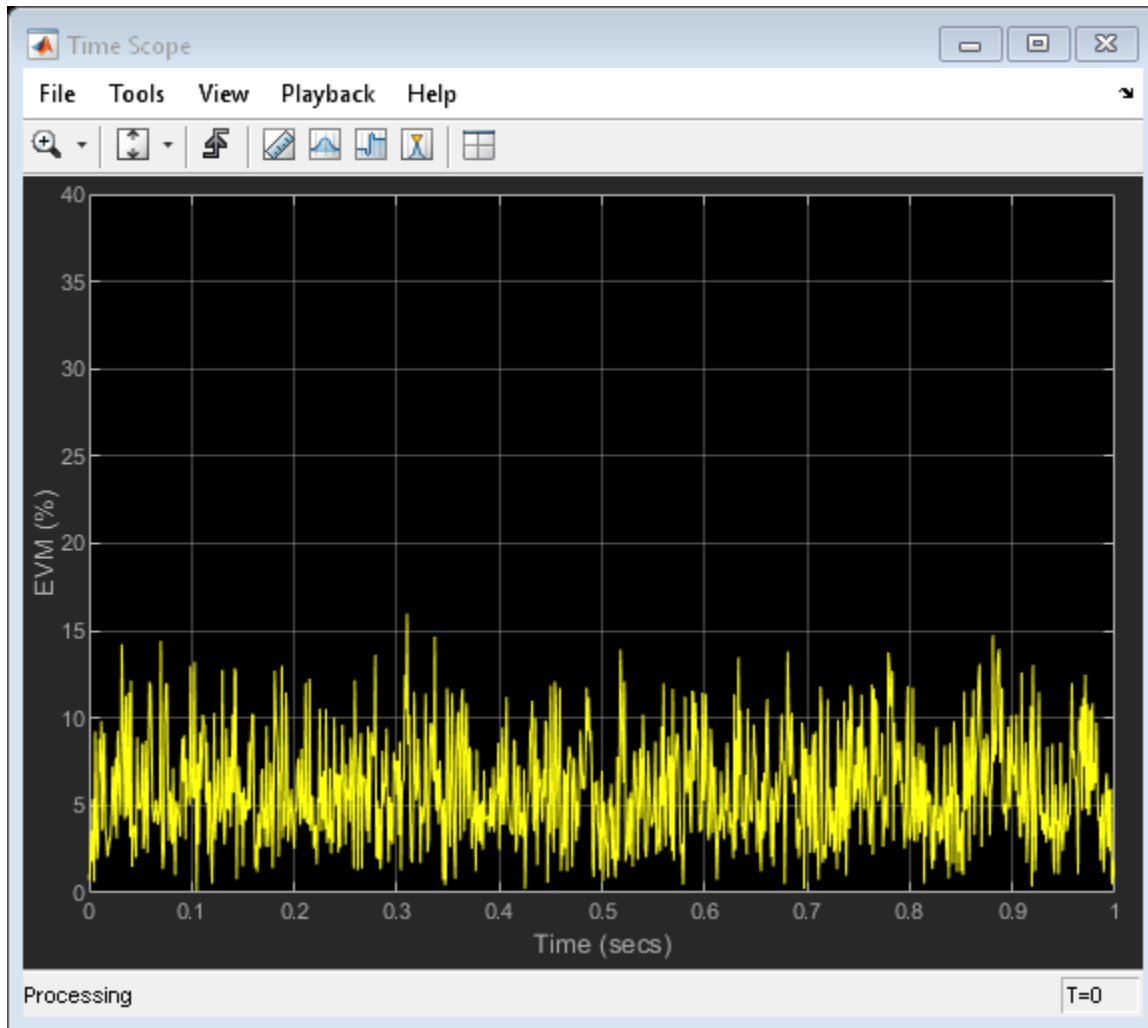
Add phase noise. Pass the impaired signal through the AWGN channel. Plot the constellation diagram.

```
rxSig = awgn(txSig,snrdB);  
pnoiseSig = pnoise(rxSig);  
rxfiltOut = rxfilter(pnoiseSig);  
constDiagram(rxfiltOut)
```



Calculate the time-varying EVM. Plot the result.

```
evmTime = evm(rxfiltOut);  
timeScope(evmTime)
```



Determine the RMS EVM.

```
evmRMS = sqrt(mean(evmTime.^2))
```

```
evmRMS =
```


6.6444

Estimate the SNR.

```
mer = comm.MER('ReferenceSignalSource', 'Estimated from reference constellation', ...  
             'ReferenceConstellation', refConst);  
snrEst = mer(rxfiltOut)
```

```
snrEst =
```

```
23.6978
```

This value is approximately 6 dB worse than the specified value of 30 dB, which means that the effects of the other impairments are significant and will degrade the bit error rate performance.

See Also

Fading Channels on page 14-10 | RF Impairments on page 14-53

C Code Generation

- “Generate C Code from MATLAB Code” on page 18-2
- “Generate C Code from Simulink Model” on page 18-12
- “C Code Generation Support List for System Objects and Functions” on page 18-19

Generate C Code from MATLAB Code

MATLAB Coder generates highly optimized ANSI C and C++ code from functions and System objects in Communications Toolbox. You can deploy this code in a wide variety of applications. The workflow described in this topic uses DSP System Toolbox features but the same workflow applies for Communications Toolbox.

This example generates C code from the “Construct a Sinusoidal Signal Using High Energy FFT Coefficients” (DSP System Toolbox) example and builds an executable from the generated code.

Here is the MATLAB code for this example:

```
L = 1020;
Sineobject = dsp.SineWave('SamplesPerFrame',L,...
'PhaseOffset',10,'SampleRate',44100,'Frequency',1000);
ft = dsp.FFT('FFTImplementation','FFTW');
ift = dsp.IFFT('FFTImplementation','FFTW','ConjugateSymmetricInput',true);
rng(1);
numIter = 1000;
for Iter = 1:numIter
    Sinewave1 = Sineobject();
    Input = Sinewave1 + 0.01*randn(size(Sinewave1));
    FFTCoeff = ft(Input);
    FFTCoeffMagSq = abs(FFTCoeff).^2;

    EnergyFreqDomain = (1/L)*sum(FFTCoeffMagSq);
    [FFTCoeffSorted, ind] = sort((1/L)*FFTCoeffMagSq,1,'descend');

    CumFFTCoeffs = cumsum(FFTCoeffSorted);
    EnergyPercent = (CumFFTCoeffs/EnergyFreqDomain)*100;
    Vec = find(EnergyPercent > 99.99);
    FFTCoeffsModified = zeros(L,1);
    FFTCoeffsModified(ind(1:Vec(1))) = FFTCoeff(ind(1:Vec(1)));
    ReconstrSignal = ift(FFTCoeffsModified);
end
max(abs(Input-ReconstrSignal))
plot(Input,'*');
hold on;
plot(ReconstrSignal,'o');
hold off;
```

You can run the generated executable inside the MATLAB environment. In addition, you can package and relocate the code to another development environment that does not have MATLAB installed. You can generate code using the MATLAB Coder app or the codegen function. This example shows you the workflow using the codegen function. For more information on the app workflow, see “Generate C Code by Using the MATLAB Coder App” (MATLAB Coder).

Set Up the Compiler

The first step is to set up a supported C compiler. MATLAB Coder automatically locates and uses a supported installed compiler. You can change the default compiler using `mex -setup`. For more details, see “Change Default Compiler” (MATLAB). For a current list of supported compilers, see Supported and Compatible Compilers.

Break Out the Computational Part of the Algorithm into a MATLAB Function

To generate C code, the entry point must be a function. You do not have to generate code for the entire MATLAB application. If you have specific portions that are computationally intensive, generate code from these portions in order to speed up your algorithm. The harness or the driver that calls this MATLAB function does not need to generate code. The harness runs in MATLAB and can contain visualization and other verification tools that are not actually part of the system under test. For example, in the “Construct a Sinusoidal Signal Using High Energy FFT Coefficients” (DSP System Toolbox) example, the `plot` functions plot the input signal and the reconstructed signal. `plot` is not supported for code generation and must stay in the harness. To generate code from the harness that contains the visualization tools, rewrite the harness as a function and declare the visualization functions as extrinsic functions using `coder.extrinsic`. To run the generated code that contains the extrinsic functions, you must have MATLAB installed on your machine.

The MATLAB code in the `for` loop that reconstructs the original signal using high-energy FFT coefficients is the computationally intensive portion of this algorithm. Speed up the `for` loop by moving this computational part into a function of its own, `GenerateSignalWithHighEnergyFFTCoeffs.m`.

```
L = 1020;
Sineobject = dsp.SineWave('SamplesPerFrame',L,...
    'SampleRate',44100, 'Frequency',1000);
rng(1);
numIter = 1000;
for Iter = 1:numIter
    Sinewave1 = Sineobject();
    Input = Sinewave1 + 0.01*randn(size(Sinewave1));
    [ReconstrSignal,numCoeff] = GenerateSignalWithHighEnergyFFTCoeffs(Input);
end
max(abs(Input-ReconstrSignal))
figure(1);
plot(Input)
hold on;
plot(ReconstrSignal, '*')
hold off
```

```

function [ReconstrSignal,numCoeff] = GenerateSignalWithHighEnergyFFTCoeffs(Input)

ft = dsp.FFT('FFTImplementation','FTW');
ift = dsp.IFFT('FFTImplementation','FTW','ConjugateSymmetricInput',true);

FFTCoeff = ft(Input);
FFTCoeffMagSq = abs(FFTCoeff).^2;
L = size(Input,1);
EnergyF = (1/L)*sum(FFTCoeffMagSq);
[FFTCoeffSorted, ind] = sort(((1/L)*FFTCoeffMagSq),1,'descend');

CumFFTCoeffs = cumsum(FFTCoeffSorted);
EnergyPercent = (CumFFTCoeffs/EnergyF)*100;
Vec = find(EnergyPercent > 99.99);
FFTCoeffsModified = zeros(L,1);
FFTCoeffsModified(ind(1:Vec(1))) = FFTCoeff(ind(1:Vec(1)));
numCoeff = Vec(1);
ReconstrSignal = ift(FFTCoeffsModified);
end

```

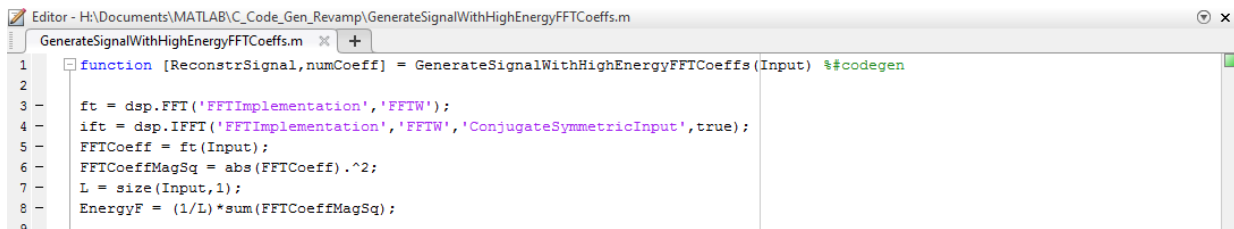
Make Code Suitable for Code Generation

Before you generate code, you must prepare your MATLAB code for code generation.

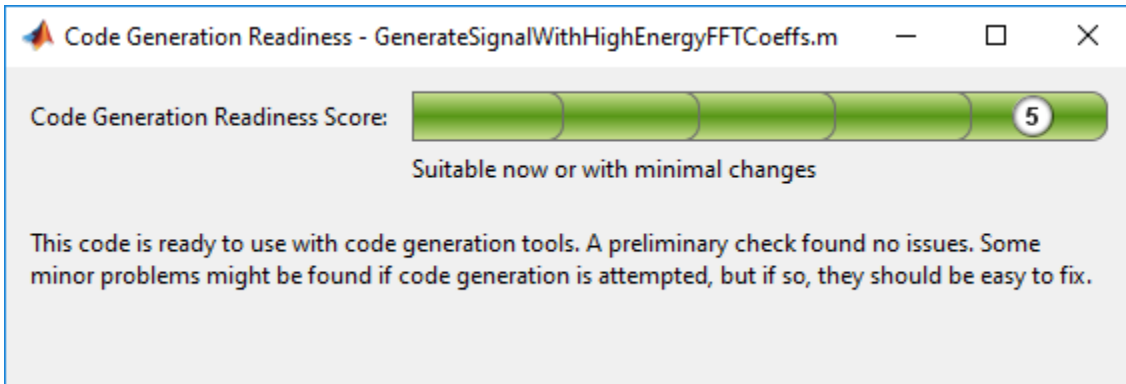
Check Issues at Design Time

The first step is to eliminate unsupported constructs and check for any code generation issues. For a list of DSP System Toolbox features supported by MATLAB Coder, see “Functions and System Objects Supported for C Code Generation” (DSP System Toolbox). For a list of supported language constructs, see “MATLAB Language Features Supported for C/C++ Code Generation” (MATLAB Coder).

The code analyzer detects coding issues at design time as you enter the code. To enable the code analyzer, you must add the `%#codegen` pragma to your MATLAB file.



The code generation readiness tool screens MATLAB code for features that are not supported for code generation. One of the ways to access this tool is by right-clicking on the MATLAB file in its current folder. Running the code generation tool on `GenerateSignalWithHighEnergyFFTCoeffs.m` finds no issues.



Check Issues at Code Generation Time

Before you generate C code, ensure that the MATLAB code successfully generates a MEX function. The `codegen` command used to generate the MEX function detects any errors that prevent the code from being suitable for code generation.

Run `codegen` on `GenerateSignalWithHighEnergyFFTCoeffs.m` function.

```
codegen -args {Input} GenerateSignalWithHighEnergyFFTCoeffs
```

The following message appears in the MATLAB command prompt:

```
??? The left-hand side has been constrained to be non-complex, but the right-hand side is complex. To correct this problem, make the right-hand side real using the function REAL, or change the initial assignment to the left-hand side variable to be a complex value using the COMPLEX function.
```

```
Error in ==> GenerateSignalWithHighEnergy Line: 24 Column: 1
Code generation failed: View Error Report
Error using codegen
```

This message is referring to the variable `FFTCoeffsModified`. The coder is expecting this variable to be initialized as a complex variable. To resolve this issue, initialize the `FFTCoeffsModified` variable as complex.

```
FFTCoeffsModified = zeros(L,1)+0i;
```

Rerun the `codegen` function and you can see that a MEX file is generated successfully in the current folder with a `.mex` extension.

```
codegen -args {Input} GenerateSignalWithHighEnergyFFTCoeffs
```

Check Issues at Run Time

Run the generated MEX function to see if there are any run-time issues reported. To do so, replace

```
[ReconstrSignal,numCoeff] = GenerateSignalWithHighEnergyFFTCoeffs(Input);
```

with

```
[ReconstrSignalMex,numCoeffMex] = GenerateSignalWithHighEnergyFFTCoeffs_mex(Input);
```

inside the harness.

The harness now looks like:

```
L = 1020;
Sineobject = dsp.SineWave('SamplesPerFrame',L,...
    'SampleRate',44100,'Frequency',1000);
rng(1);
numIter = 1000;
for Iter = 1:numIter
    Sinewave1 = Sineobject();
    Input = Sinewave1 + 0.01*randn(size(Sinewave1));
    [ReconstrSignalMex,numCoeffMex] = GenerateSignalWithHighEnergyFFTCoeffs_mex(Input,L);
end
max(abs(Input-ReconstrSignalMex))
figure(1);
plot(Input)
hold on;
plot(ReconstrSignalMex, '*')
hold off
```

The code runs successfully, indicating that there are no run-time errors.

Compare the MEX Function with the Simulation

Notice that the harness runs much faster with the MEX function compared to the regular function. The reason for generating the MEX function is not only to detect code generation and run-time issues, but also to speed up specific parts of your algorithm. For an example, see “Signal Processing Algorithm Acceleration in MATLAB” (DSP System Toolbox).

You must also check that the numeric output results from the MEX and the regular function match. Compare the reconstructed signal generated by the `GenerateSignalWithHighEnergyFFTCoeffs.m` function and its MEX counterpart `GenerateSignalWithHighEnergyFFTCoeffs_mex`.

```
max(abs(ReconstrSignal-ReconstrSignalMex))
```

```
ans =
```


2.2204e-16

The results match very closely, confirming that the code generation is successful.

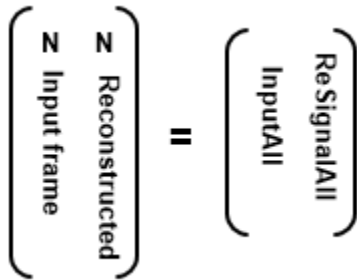
Generate a Standalone Executable

If your goal is to run the generated code inside the MATLAB environment, your build target can just be a MEX function. If deployment of code to another application is the goal, then generate a standalone executable from the entire application. To do so, the harness must be a function that calls the subfunction

`GenerateSignalWithHighEnergyFFTCoeffs`. Rewrite the harness as a function.

```
function reconstructSignalTestbench()
L = 1020;
Sineobject = dsp.SineWave('SamplesPerFrame',L,...
    'SampleRate',44100,'Frequency',1000);
rng(1);
numIter = 1000;
for Iter = 1:numIter
    Sinewave1 = Sineobject();
    Input = Sinewave1 + 0.01*randn(size(Sinewave1));
    [ReconstrSignal,numCoeff] = GenerateSignalWithHighEnergyFFTCoeffs(Input,L);
end
```

Log all 1000 frames of the input and reconstructed signal and the number of FFT coefficients used to reconstruct each frame of the signal. Write all this data to a binary file named `data.bin` using the `dsp.BinaryFileWriter` System object. This example logs the number of coefficients, which are scalar values, as the first element of each frame of the input signal and the reconstructed signal. The data to be written has a frame size of $M = L + 1$ and has a format that looks like this figure.



N is the number of FFT coefficients that represent 99.99% of the signal energy of the current input frame. The meta data of the binary file specifies this information. Release the binary file writer and close the binary file at the end.

The updated harness function, `reconstructSignalTestbench`, is shown here:

```
function reconstructSignalTestbench()
L = 1020;
Sineobject = dsp.SineWave('SamplesPerFrame',L,...
    'SampleRate',44100,'Frequency',1000);
header = struct('FirstElemInBothCols','Number of Coefficients',...
    'FirstColumn','Input','SecondColumn','ReconstructedSignal');
bfw = dsp.BinaryFileWriter('data.bin','HeaderStructure',header);
numIter = 1000;

M = L+1;
ReSignalAll = zeros(M*numIter,1);
InputAll = zeros(M*numIter,1);
rng(1);

for Iter = 1 : numIter
    Sinewave1 = Sineobject();
    Input = Sinewave1 + 0.01*randn(size(Sinewave1));
    [ReconstrSignal,numCoeffs] = GenerateSignalWithHighEnergyFFTCoeffs(Input);
    InputAll(((Iter-1)*M)+1:Iter*M) = [numCoeffs;Input];
    ReSignalAll(((Iter-1)*M)+1:Iter*M) = [numCoeffs;ReconstrSignal];
end

bfw([InputAll ReSignalAll]);
release(bfw);
```

The next step in generating a C executable is to create a `coder.config` object for an executable and provide a `main.c` function to this object.

```
cfg = coder.config('exe');
cfg.CustomSource = 'reconstructSignalTestbench_Main.c';
```

Here is how the `reconstructSignalTestbench_Main.c` function looks for this example.

```
/*
** reconstructSignalTestbench_main.c
*
* Copyright 2017 The MathWorks, Inc.
*/
#include <stdio.h>
#include <stdlib.h>

#include "reconstructSignalTestbench_initialize.h"
#include "reconstructSignalTestbench.h"
#include "reconstructSignalTestbench_terminate.h"
```

```
int main()
{
    reconstructSignalTestbench_initialize();
    reconstructSignalTestbench();
    reconstructSignalTestbench_terminate();

    return 0;
}
```

For additional details on creating the main function, see “Generating Standalone C/C++ Executables from MATLAB Code” (MATLAB Coder).

Set the `CustomInclude` property of the configuration object to specify the location of the main file. In this example, the location is the current folder.

```
cfg.CustomInclude = ['"',pwd,''];
```

Generate the C executable by running the following command in the MATLAB command prompt:

```
codegen -config cfg -report reconstructSignalTestbench
```

MATLAB Coder compiles and links the main function with the C code that it generates from the `reconstructSignalTestbench.m`.

If you are using Windows, you can see that `reconstructSignalTestbench.exe` is generated in the current folder. If you are using Linux, the generated executable does not have the `.exe` extension.

Read and Verify the Binary File Data

Running the executable creates a binary file, `data.bin`, in the current directory and writes the input, reconstructed signal, and the number of FFT coefficients used to reconstruct the signal.

```
!reconstructSignalTestbench
```

You can read this data from the binary file using the `dsp.BinaryFileReader` object. To verify that the data is written correctly, read data from the binary file in MATLAB and compare the output with variables `InputAll` and `ReSignalAll`.

The header prototype must have a structure similar to the header structure written to the file. Read the data as two channels.

```
M = 1021;
numIter = 1000;
headerPro = struct('FirstElemInBothCols','Number of Coefficients',...
    'FirstColumn','Input','SecondColumn','ReconstructedSignal');
bfr = dsp.BinaryFileReader('data.bin','HeaderStructure',...
    headerPro,'SamplesPerFrame',M*numIter,'NumChannels',2);
Data = bfr();
```

Compare the first channel with `InputAll` and the second channel with `ReSignalAll`.

```
isequal(InputAll,Data(:,1))
```

```
ans =
```

```
logical
```

```
1
```

```
isequal(ReSignalAll,Data(:,2))
```

```
ans =
```

```
logical
```

```
1
```

The results match exactly, indicating a successful write operation.

Relocate Code to Another Development Environment

Once you generate code from your MATLAB algorithm, you can relocate the code to another development environment, such as a system or an integrated development environment (IDE) that does not include MATLAB. You can package the files into a compressed file using the `packNGo` function at the command line or the **Package** option in the MATLAB Coder app. For an example that illustrates both the workflows, see “Package Code for Other Development Environments” (MATLAB Coder). For more information on the `packNGo` option, see `packNGo` in “Build Information Methods” (MATLAB Coder). You can relocate and unpack the compressed zip file using a standard zip utility. For an example on how to package the executable generated in this example, see “Relocate Code Generated from MATLAB Code to Another Development Environment” (DSP System Toolbox).

See Also

Functions

codegen

More About

- “Relocate Code Generated from MATLAB Code to Another Development Environment” (DSP System Toolbox)
- “Generate C Code from Simulink Model” (DSP System Toolbox)
- “Generate C Code by Using the MATLAB Coder App” (MATLAB Coder)
- “Generate C Code at the Command Line” (MATLAB Coder)
- “Code Generation Workflow” (MATLAB Coder)

External Websites

- Supported and Compatible Compilers

Generate C Code from Simulink Model

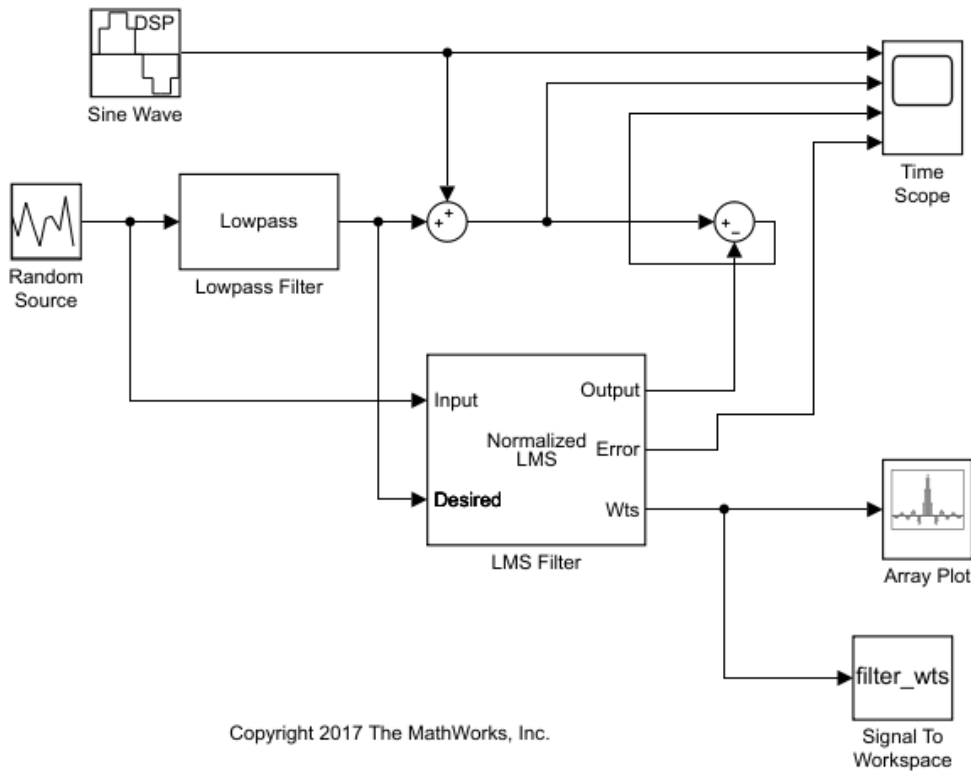
Simulink Coder generates standalone C and C++ code from Simulink models for deployment in a wide variety of applications. The workflow described in this topic uses DSP System Toolbox features but the same workflow applies for Communications Toolbox.

This example generates C code from the `ex_codegen_dsp` model and builds an executable from the generated code. You can run the executable inside the MATLAB environment. In addition, you can package and relocate the code to another development environment that does not have the MATLAB and Simulink products installed.

Open the Model

The `ex_codegen_dsp` model implements a simple adaptive filter to remove noise from a signal while simultaneously identifying a filter that characterizes the noise frequency content. To open this model, enter the following command in MATLAB command prompt:

```
open_system('ex_codegen_dsp')
```



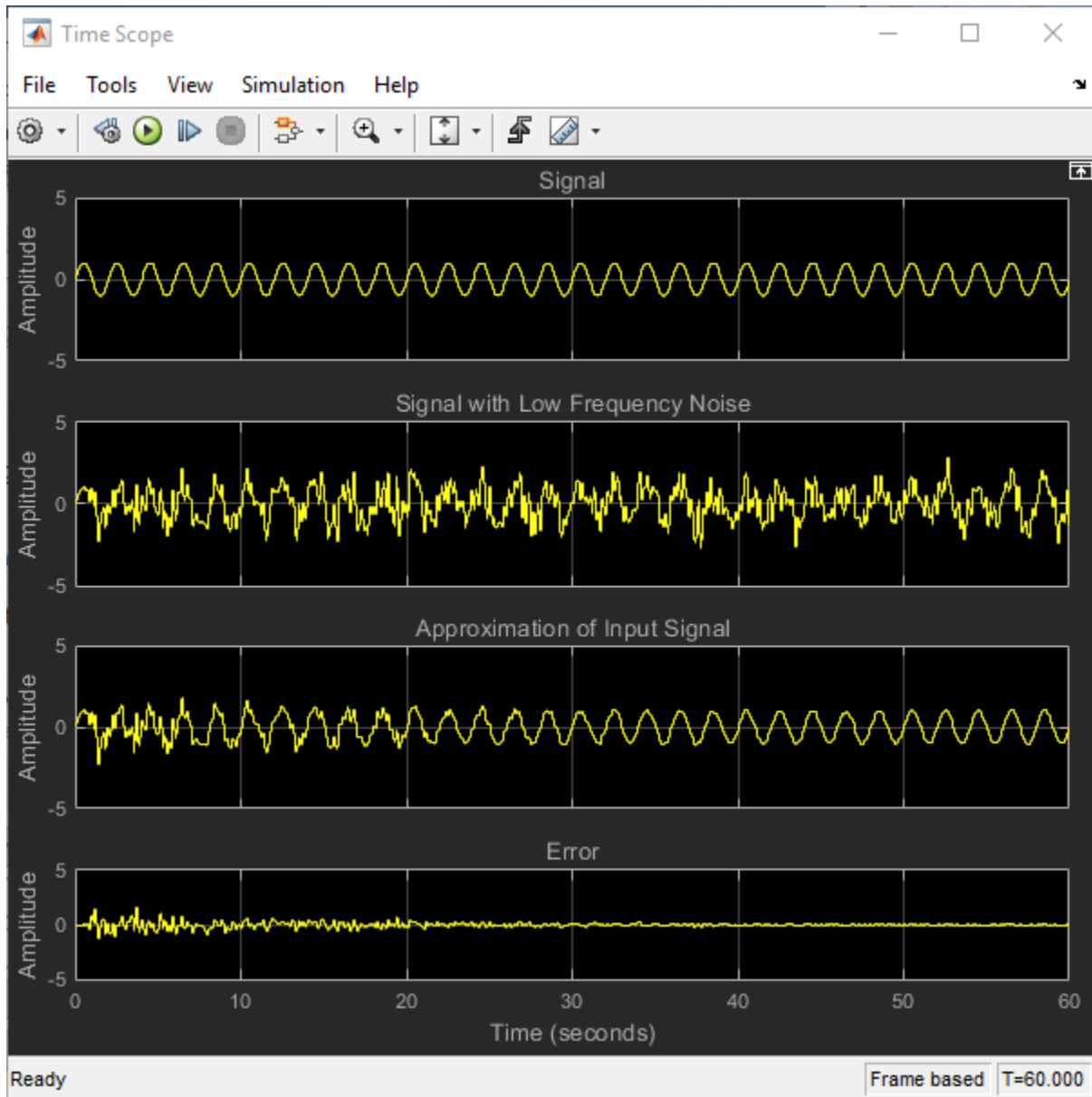
You can alternatively create the model using the **DSP System** template. For more information, see “Configure the Simulink Environment for Signal Processing Models” (DSP System Toolbox).

Configure Model for Code Generation

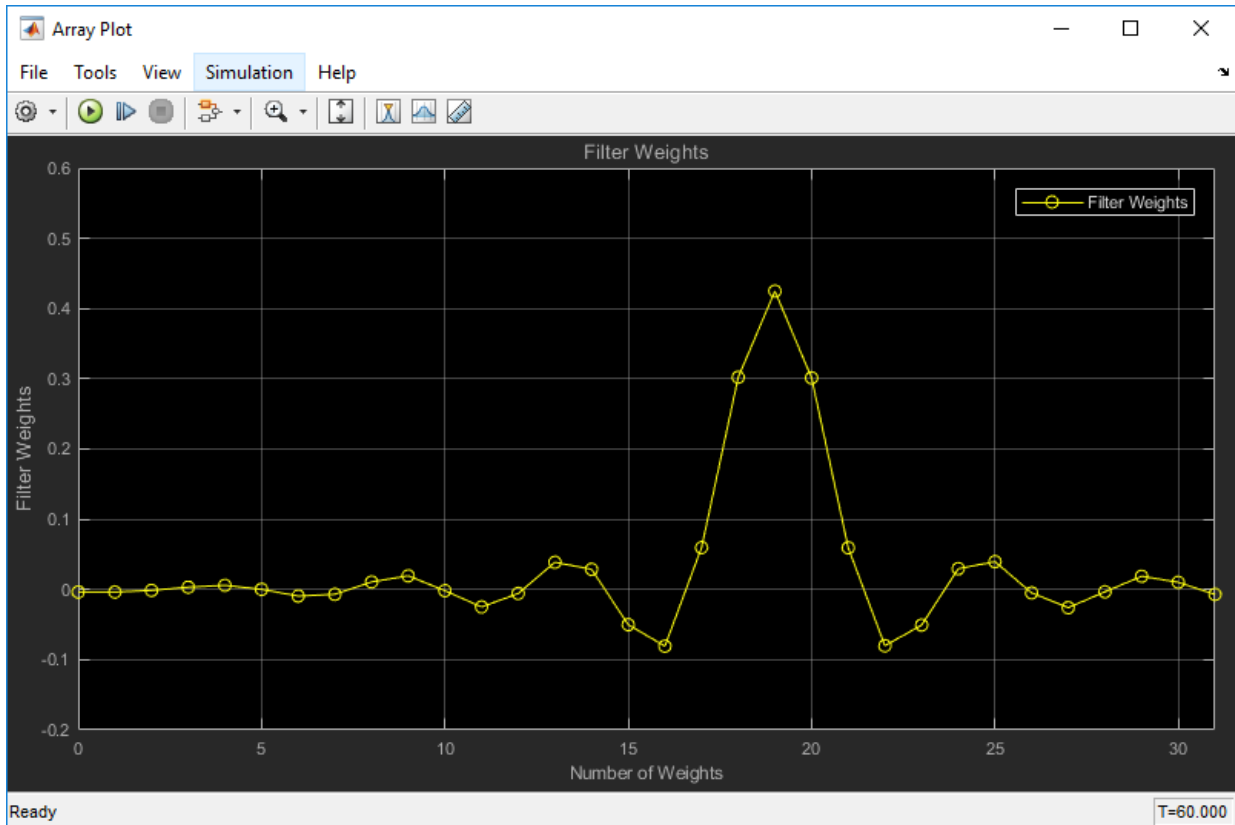
Prepare the model for code generation by specifying code generation settings in the **Configuration Parameters** dialog box. Choose the appropriate solver and code generation target, and check the model configuration for execution efficiency. For more details on each of these steps, see “Generate C Code for a Model” (Simulink Coder).

Simulate the Model

Simulate the model. The Time Scope shows the input and filtered signal characteristics.



The Array Plot shows the last 32 filter weights for which the LMS filter has effectively adapted and filtered out the noise from the signal.



These coefficients can also be accessed using the following command:

```
filter_wts(:,:,1201)
```

Generate Code from the Model

Before you generate code from the model, you must first ensure that you have write permission in your current folder.

To generate code, you must make the following changes:

- 1 In the **Modeling** tab of the model toolstrip, click **Model Settings**. The **Configuration Parameters** dialog opens. Navigate to the **Code Generation** tab, select the **Generate code only** parameter, and click **Apply**.
- 2 In the Apps gallery, click **Simulink Coder**. The **C Code** tab appears. Click the

Generate Code icon ().

After the model finishes generating code, the **Code Generation Report** appears, allowing you to inspect the generated code. Note that the build process creates a new subfolder called `ex_codegen_dsp_grt_rtw` in your current MATLAB working folder. This subfolder contains all the files created by the code generation process, including those that contain the generated C source code. For more information on viewing the generated code, see “Generate C Code for a Model” (Simulink Coder).

Build and Run the Generated Code

Set Up the C/C++ Compiler

To build an executable, you must set up a supported C compiler. For a list of compilers supported in the current release, see Supported and Compatible Compilers.

To set up your compiler, run the following command in the MATLAB command prompt:

```
mex -setup
```

Build the Generated Code

After your compiler is setup, you can build and run the compiled code. The `ex_codegen_dsp` model is currently configured to generate code only. To build the generated code, you must first make the following changes:

- 1 In the **Modeling** tab of the model toolstrip, click **Model Settings**. The **Configuration Parameters** dialog opens. Navigate to the **Code Generation** tab, clear the **Generate code only** parameter, and click **Apply**.

2

In the **C Code** tab of the model toolstrip, click the **Build** icon ().

The code generator builds the executable and generates the **Code Generation Report**. The code generator places the executable in the working folder. On Windows, the executable is `ex_codegen_dsp.exe`. On Linux, the executable is `ex_codegen_dsp`.

Run the Generated Code

To run the generated code, enter the following command in the MATLAB command prompt:

```
!ex_codegen_dsp
```

Running the generated code creates a MAT-file that contains the same variables as those generated by simulating the model. The variables in the MAT-file are named with a prefix of `rt_`. After you run the generated code, you can load the variables from the MAT-file by typing the following command at the MATLAB prompt:

```
load ex_codegen_dsp.mat
```

You can now compare the variables from the generated code with the variables from the model simulation. To access the last set of coefficients from the generated code, enter the following in the MATLAB prompt:

```
rt_filter_wts(:,:,1201)
```

Note that the coefficients in `filter_wts(:,:,1201)` and `rt_filter_wts(:,:,1201)` match.

For more details on building and running the executable, see “Generate C Code for a Model” (Simulink Coder).

Relocate Code to Another Development Environment

Once you generate code from your Simulink model, you can relocate the code to another development environment using the pack-and-go utility. Use this utility when the development environment does not have the MATLAB and Simulink products.

The pack-and-go utility uses the tools for customizing the build process after code generation and a `packNGo` function to find and package files for building an executable image. The files are packaged in a compressed file that you can relocate and unpack using a standard zip utility.

You can package the code by either using the user interface or by using the command-line interface. The command-line interface provides more control over the details of code packaging. For more information on each of these methods, see “Relocate Code to Another Development Environment” (Simulink Coder).

For an example on how to package the C code and executable generated from this example, see “Relocate Code Generated from a Simulink Model to Another Development Environment” (DSP System Toolbox).

See Also

More About

- “Generate C Code for a Model” (Simulink Coder)
- “Relocate Code Generated from a Simulink Model to Another Development Environment” (DSP System Toolbox)
- “Relocate Code to Another Development Environment” (Simulink Coder)
- “Generate C Code from MATLAB Code” (DSP System Toolbox)
- “How To Run a Generated Executable Outside MATLAB” (DSP System Toolbox)

External Websites

- Supported and Compatible Compilers

C Code Generation Support List for System Objects and Functions

The following System objects and functions support code generation in MATLAB via the MATLAB Coder product. To use the `codegen` function, you must have a MATLAB Coder license.

An asterisk (*) indicates that the reference page has usage notes and limitations for C/C++ code generation.

<code>algdeintrlv</code>	Restore ordering of symbols using algebraically derived permutation table
<code>algintrlv</code>	Reorder symbols using algebraically derived permutation table
<code>alignsignals</code>	Align two signals by delaying earliest signal
<code>apskdemod</code>	Amplitude phase shift keying (APSK) demodulation
<code>apskmod</code>	Amplitude phase shift keying (APSK) modulation
<code>awgn*</code>	Add white Gaussian noise to signal
<code>bchgenpoly*</code>	Generator polynomial of BCH code
<code>bi2de</code>	Convert binary vectors to decimal numbers
<code>bin2gray</code>	Convert positive integers into corresponding Gray-encoded integers
<code>bleATTPDU</code>	Generate BLE ATT PDU
<code>bleATTPDUConfig</code>	Create configuration object for BLE ATT PDU
<code>bleATTPDUDecode</code>	Decode BLE ATT PDU
<code>bleChannelSelection*</code>	Select BLE channel index
<code>bleGAPDataBlock</code>	Generate BLE GAP data block
<code>bleGAPDataBlockConfig</code>	Create configuration object for BLE GAP data block
<code>bleGAPDataBlockDecode</code>	Decode BLE GAP data block
<code>bleIdealReceiver</code>	Ideal receiver for BLE PHY waveform
<code>bleL2CAPFrame</code>	Generate BLE L2CAP frame

bleL2CAPFrameConfig	Create configuration object for BLE L2CAP frame
bleL2CAPFrameDecode	Decode BLE L2CAP frame
bleLLAdvertisingChannelPDU	Generate BLE LL advertising channel PDU
bleLLAdvertisingChannelPDUDecode	Decode BLE LL advertising channel PDU
bleLLControlPDUConfig	Create configuration object for BLE LL control PDU payload configuration
bleLLDataChannelPDU*	Generate BLE LL data channel PDU
bleLLDataChannelPDUConfig	Create configuration object for BLE LL data channel PDU
bleLLDataChannelPDUDecode*	Decode BLE LL data channel PDU
bleWaveformGenerator	Waveform generator for BLE PHY
bsc*	Binary symmetric channel
comm.ACPR*	Adjacent Channel Power Ratio measurements
comm.AGC*	Adaptively adjust gain for constant signal-level output
comm.AlgebraicDeinterleaver*	(To be removed) Deinterleave input symbols using algebraically derived permutation vector
comm.AlgebraicInterleaver*	(To be removed) Permute input symbols using algebraically derived permutation vector
comm.APPDecoder*	Decode convolutional code using the a posteriori probability method
comm.AWGNChannel*	Add white Gaussian noise to input signal
comm.BarkerCode*	Generate bipolar Barker code
comm.BasebandFileReader*	Read baseband signals from file
comm.BasebandFileWriter*	Write baseband signals to file

<code>comm.BCHDecoder*</code>	Decode data using BCH decoder
<code>comm.BCHEncoder*</code>	Encode data using BCH encoder
<code>comm.BinarySymmetricChannel*</code>	(To be removed) Introduce binary errors
<code>comm.BlockDeinterleaver*</code>	(To be removed) Deinterleave input symbols using permutation vector
<code>comm.BlockInterleaver*</code>	(To be removed) Permute input symbols using permutation vector
<code>comm.BPSKDemodulator*</code>	Demodulate using BPSK method
<code>comm.BPSKModulator*</code>	Modulate using BPSK method
<code>comm.CarrierSynchronizer*</code>	Compensate for carrier frequency offset
<code>comm.CCDF*</code>	Measure complementary cumulative distribution function
<code>comm.CoarseFrequencyCompensator*</code>	Compensate for frequency offset for PAM, PSK, or QAM
<code>comm.ConstellationDiagram*</code>	Display constellation diagram for input signals
<code>comm.ConvolutionalDeinterleaver*</code>	Restore ordering of symbols using shift registers
<code>comm.ConvolutionalEncoder*</code>	Convolutionally encode binary data
<code>comm.ConvolutionalInterleaver*</code>	Permute input symbols using shift registers with same property values
<code>comm.CPFSKDemodulator*</code>	Demodulate using CPFSK method and Viterbi algorithm
<code>comm.CPFSKModulator*</code>	Modulate using CPFSK method
<code>comm.CPMCarrierPhaseSynchronizer*</code>	(To be removed) Recover carrier phase of baseband CPM signal
<code>comm.CPMDemodulator*</code>	Demodulate using CPM method and Viterbi algorithm

<code>comm.CPModulator*</code>	Modulate using CPM method
<code>comm.CRCDetector*</code>	Detect errors in input data using CRC
<code>comm.CRCGenerator*</code>	Generate CRC code bits and append to input data
<code>comm.DBPSKDemodulator*</code>	Demodulate using DBPSK method
<code>comm.DBPSKModulator*</code>	Modulate using DBPSK method
<code>comm.DecisionFeedbackEqualizer</code>	Equalize modulated signals using decision feedback filtering
<code>comm.Descrambler*</code>	Descramble input signal
<code>comm.DifferentialDecoder*</code>	Decode binary signal using differential decoding
<code>comm.DifferentialEncoder*</code>	Encode binary signal using differential coding
<code>comm.DiscreteTimeVCO*</code>	Generate variable frequency sinusoid
<code>comm.DPD*</code>	Digital predistorter
<code>comm.DPDCoefficientEstimator*</code>	Estimate memory-polynomial coefficients for digital predistortion
<code>comm.DPSKDemodulator*</code>	Demodulate using M-ary DPSK method
<code>comm.DPSKModulator*</code>	Modulate using M-ary DPSK method
<code>comm.DQPSKDemodulator*</code>	Demodulate using DQPSK method
<code>comm.DQPSKModulator*</code>	Modulate using DQPSK method
<code>comm.ErrorRate*</code>	Compute bit or symbol error rate of input data
<code>comm.EVM*</code>	Measure error vector magnitude
<code>comm.EyeDiagram*</code>	Display eye diagram of time-domain signals

<code>comm.FMBroadcastDemodulator*</code>	Demodulate broadcast FM signal
<code>comm.FMBroadcastModulator*</code>	Modulate broadcast FM signal
<code>comm.FMDemodulator*</code>	Demodulate using FM method
<code>comm.FMModulator*</code>	Modulate using FM method
<code>comm.FSKDemodulator*</code>	Demodulate using M-ary FSK method
<code>comm.FSKModulator*</code>	Modulate using M-ary FSK method
<code>comm.GeneralQAMDemodulator*</code>	Demodulate using arbitrary QAM constellation
<code>comm.GeneralQAMModulator*</code>	Modulate using arbitrary QAM constellation
<code>comm.GeneralQAMTCMDemodulator*</code>	Demodulate convolutionally encoded data mapped to arbitrary QAM constellation
<code>comm.GeneralQAMTCMModulator*</code>	Convolutionally encode binary data and map using arbitrary QAM constellation
<code>comm.GMSKDemodulator*</code>	Demodulate using GMSK method and the Viterbi algorithm
<code>comm.GMSKModulator*</code>	Modulate using GMSK method
<code>comm.GMSKTimingSynchronizer*</code>	Recover symbol timing phase using fourth-order nonlinearity method
<code>comm.GoldSequence*</code>	Generate Gold sequence
<code>comm.HadamardCode*</code>	Generate Hadamard code
<code>comm.HDLCRCDetector*</code>	Detect errors in input data using CRC
<code>comm.HDLCRCGenerator*</code>	Generate CRC code bits and append to input data

<code>comm.HDLRSDecoder*</code>	Decode message using Reed-Solomon decoder
<code>comm.HDLRSEncoder*</code>	Encode message using Reed-Solomon encoder
<code>comm.HelicalDeinterleaver*</code>	Restore ordering of symbols using helical array
<code>comm.HelicalInterleaver*</code>	Permute input symbols using helical array
<code>comm.IntegrateAndDumpFilter*</code>	Integrate discrete-time signal with periodic resets
<code>comm.IQImbalanceCompensator*</code>	Compensate for I/Q imbalance
<code>comm.KasamiSequence*</code>	Generate Kasami sequence
<code>comm.LDPCDecoder*</code>	Decode binary low-density parity-check code
<code>comm.LDPCEncoder*</code>	Encode binary low-density parity-check code
<code>comm.LinearEqualizer</code>	Equalize modulated signals using linear filtering
<code>comm.LTEMIMOChannel*</code>	(To be removed) Filter input signal through LTE MIMO multipath fading channel
<code>comm.MatrixDeinterleaver*</code>	(To be removed) Deinterleave input symbols using permutation matrix
<code>comm.MatrixHelicalScanDeinterleaver*</code>	(To be removed) Deinterleave input symbols by filling a matrix along diagonals
<code>comm.MatrixHelicalScanInterleaver*</code>	(To be removed) Permute input symbols by selecting matrix elements along diagonals
<code>comm.MatrixInterleaver*</code>	(To be removed) Permute input symbols using permutation matrix
<code>comm.MemorylessNonlinearity*</code>	Apply memoryless nonlinearity to input signal
<code>comm.MER*</code>	Measure modulation error ratio
<code>comm.MIMOChannel*</code>	Filter input signal through MIMO multipath fading channel

<code>comm.MLSEEqualizer*</code>	Equalize using maximum likelihood sequence estimation
<code>comm.MSKDemodulator*</code>	Demodulate using MSK method and the Viterbi algorithm
<code>comm.MSKModulator*</code>	Modulate using MSK method
<code>comm.MSKTimingSynchronizer*</code>	Recover symbol timing phase using fourth-order nonlinearity method
<code>comm.MultiplexedDeinterleaver*</code>	Deinterleave input symbols using set of shift registers with specified delays
<code>comm.MultiplexedInterleaver*</code>	Permute input symbols using set of shift registers with specified delays
<code>comm.OFDMDemodulator*</code>	Demodulate using OFDM method
<code>comm.OFDMModulator*</code>	Modulate using OFDM method
<code>comm.OQPSKDemodulator*</code>	Demodulation using OQPSK method
<code>comm.OQPSKModulator*</code>	Modulation using OQPSK method
<code>comm.OSTBCCombiner*</code>	Combine inputs using orthogonal space-time block code
<code>comm.OSTBCEncoder*</code>	Encode input using orthogonal space-time block code
<code>comm.PAMDemodulator*</code>	(Not recommended) Demodulate using M-ary PAM method
<code>comm.PAMModulator*</code>	(Not recommended) Modulate using M-ary PAM method
<code>comm.PhaseFrequencyOffset*</code>	Apply phase and frequency offsets to input signal
<code>comm.PhaseNoise*</code>	Apply phase noise to baseband signal
<code>comm.PNSequence*</code>	Generate a pseudo-noise (PN) sequence

<code>comm.PreambleDetector*</code>	Detect preamble in data
<code>comm.PSKCoarseFrequencyEstimator*</code>	(To be removed) Estimate frequency offset for PSK signal
<code>comm.PSKDemodulator*</code>	Demodulate using M-ary PSK method
<code>comm.PSKModulator*</code>	Modulate using M-ary PSK method
<code>comm.PSKTCMDemodulator*</code>	Demodulate convolutionally encoded data mapped to M-ary PSK signal constellation
<code>comm.PSKTCMModulator*</code>	Convolutionally encode binary data and map using M-ary PSK signal constellation
<code>comm.QAMCoarseFrequencyEstimator*</code>	(To be removed) Estimate frequency offset for QAM signal
<code>comm.QPSKDemodulator*</code>	Demodulate using QPSK method
<code>comm.QPSKModulator*</code>	Modulate using QPSK method
<code>comm.RaisedCosineReceiveFilter*</code>	Apply pulse shaping by decimating signal using raised cosine filter
<code>comm.RaisedCosineTransmitFilter*</code>	Apply pulse shaping by interpolating signal using raised cosine filter
<code>comm.RayleighChannel*</code>	Filter input signal through a Rayleigh multipath fading channel
<code>comm.RBDSWaveformGenerator*</code>	Generate RDS/RBDS waveform
<code>comm.RectangularQAMDemodulator*</code>	(To be removed) Demodulate using rectangular QAM signal constellation
<code>comm.RectangularQAMModulator*</code>	(To be removed) Modulate using rectangular QAM signal constellation
<code>comm.RectangularQAMTCMDemodulator*</code>	Demodulate convolutionally encoded data mapped to rectangular QAM signal constellation

<code>comm.RectangularQAMTCMModulator*</code>	Convolutionally encode binary data and map using rectangular QAM signal constellation
<code>comm.RicianChannel*</code>	Filter input signal through a Rician fading channel
<code>comm.RSDecoder*</code>	Decode data using Reed-Solomon decoder
<code>comm.RSEncoder*</code>	Encode data using Reed-Solomon encoder
<code>comm.Scrambler*</code>	Scramble input signal
<code>comm.SphereDecoder*</code>	Decode input using sphere decoder
<code>comm.SymbolSynchronizer*</code>	Correct symbol timing clock skew
<code>comm.ThermalNoise*</code>	Add thermal noise to signal
<code>comm.TurboDecoder*</code>	Decode input signal using parallel concatenated decoding scheme
<code>comm.TurboEncoder*</code>	Encode input signal using parallel concatenated encoding scheme
<code>comm.ViterbiDecoder*</code>	Decode convolutionally encoded data using Viterbi algorithm
<code>comm.WalshCode*</code>	Generate Walsh code from orthogonal set of codes
<code>convenc</code>	Convolutionally encode binary data
<code>de2bi</code>	Convert decimal numbers to binary vectors
<code>deintrlv</code>	Restore ordering of symbols
<code>doppler*</code>	Construct Doppler spectrum structure
<code>dpskdemod</code>	Differential phase shift keying demodulation
<code>dpskmod</code>	Differential phase shift keying modulation
<code>dsp.ArrayPlot*</code>	Display vectors or arrays
<code>dsp.DCBlocker*</code>	Block DC component (offset) from input signal
<code>dsp.SpectrumAnalyzer*</code>	Display frequency spectrum of time-domain signals
<code>dsp.TimeScope*</code>	Time domain signal display and measurement

dvbs2ldpc*	Low-density parity-check codes from DVB-S.2 standard
dvbsapskdemod	DVB-S2/S2X/SH standard-specific amplitude phase shift keying (APSK) demodulation
dvbsapskmod	DVB-S2/S2X/SH standard-specific amplitude phase shift keying (APSK) modulation
finddelay	Estimate delay(s) between signals
fogpl*	RF signal attenuation due to fog and clouds
fspl*	Free space path loss
gaspl*	RF signal attenuation due to atmospheric gases
genqamdmod	General quadrature amplitude demodulation
genqammod	General quadrature amplitude modulation
gray2bin	Convert Gray-encoded positive integers to corresponding Gray-decoded integers
gsmCheckTimeMask*	Inspect GSM burst against time mask
gsmDownlinkConfig*	Create GSM downlink TDMA frame configuration object
gsmFrame	Create GSM waveform
gsmInfo	View GSM waveform information
gsmUplinkConfig*	Create GSM uplink TDMA frame configuration object
helscandeintrlv	Restore ordering of symbols in helical pattern
helscanintrlv	Reorder symbols in helical pattern
intrlv	Reorder sequence of symbols
iqcoef2imbal	Convert compensator coefficient to amplitude and phase imbalance
iqimbal	Apply I/Q imbalance to input signal
iqimbal2coef	Convert I/Q imbalance to compensator coefficient
istrellis	True for valid trellis structure
matdeintrlv	Restore ordering of symbols by filling matrix by columns and emptying it by rows
matintrlv	Reorder symbols by filling matrix by rows and emptying it by columns

mil188qamdemod	MIL-STD-188-110 B/C standard-specific quadrature amplitude demodulation
mil188qammod	MIL-STD-188-110 B/C standard-specific quadrature amplitude modulation (QAM)
oct2dec	Convert octal to decimal numbers
ofdmmod	Demodulate time-domain signal using orthogonal frequency division multiplexing (OFDM)
ofdmmod	Modulate frequency-domain signal using orthogonal frequency division multiplexing (OFDM)
pamdemod	Pulse amplitude demodulation
pammod	Pulse amplitude modulation
poly2trellis*	Convert convolutional code polynomials to trellis description
qamdemod	Quadrature amplitude demodulation
qammod	Quadrature amplitude modulation
rainpl*	RF signal attenuation due to rainfall
randi*	Uniformly distributed pseudorandom integers
rangeangle*	Range and angle calculation
rcosdesign*	Raised cosine FIR pulse-shaping filter design
rsgenpoly*	Generator polynomial of Reed-Solomon code
rsgenpolycoeffs*	Generator polynomial coefficients of Reed-Solomon code
tpcdec*	Turbo product code (TPC) decoder
tpcenc*	Turbo product code (TPC) encoder
vitdec	Convolutionally decode binary data using Viterbi algorithm
wgn*	Generate white Gaussian noise samples
zadoffChuSeq	Generate root Zadoff-Chu sequence

HDL Code Generation

- “HDL Code Generation Support for Communications Toolbox” on page 19-2
- “Find Blocks and System Objects Supporting HDL Code Generation” on page 19-5
- “Wireless Communications Design for FPGAs and ASICs” on page 19-7

HDL Code Generation Support for Communications Toolbox

In this section...
“Blocks” on page 19-2
“System Objects” on page 19-3

Blocks

You can find Communications Toolbox blocks that support HDL code generation, in the 'Communications Toolbox HDL Support' library, in the Simulink library browser. Alternately, you can type `commhdllib` in the MATLAB command prompt to open this library. The blocks in `commhdllib` have their parameters set for HDL code generation.

Error Correction

- Convolutional Encoder
- General CRC Generator HDL Optimized
- General CRC Syndrome Detector HDL Optimized
- Integer-Input RS Encoder HDL Optimized
- Integer-Output RS Decoder HDL Optimized
- Viterbi Decoder

Filtering

- DC Blocker
- Raised Cosine Receive Filter
- Raised Cosine Transmit Filter

Interleaving

- Convolutional Deinterleaver
- Convolutional Interleaver
- General Multiplexed Deinterleaver
- General Multiplexed Interleaver

Modulation

- BPSK Demodulator Baseband
- BPSK Modulator Baseband
- M-PSK Demodulator Baseband
- M-PSK Modulator Baseband
- QPSK Demodulator Baseband
- QPSK Modulator Baseband
- Rectangular QAM Demodulator Baseband
- Rectangular QAM Modulator Baseband

Sequence Generation

- PN Sequence Generator

Sinks and Visualization

These blocks can be used for simulation visibility in models that generate HDL code, but are not included in the hardware implementation.

- Constellation Diagram
- Eye Diagram
- Error Rate Calculation

System Objects

HDL Coder supports the following Communications Toolbox System objects for HDL code generation:

Error Correction

- `comm.HDLCRCDetector`
- `comm.HDLCRCGenerator`
- `comm.HDLRSDecoder`
- `comm.HDLRSEncoder`
- `comm.ViterbiDecoder`

Interleaving

- `comm.ConvolutionalDeinterleaver`
- `comm.ConvolutionalInterleaver`

Modulation

- `comm.BPSKDemodulator`
- `comm.BPSKModulator`
- `comm.PSKDemodulator`
- `comm.PSKModulator`
- `comm.QPSKDemodulator`
- `comm.QPSKModulator`

Find Blocks and System Objects Supporting HDL Code Generation

Blocks

In the Simulink library browser, you can find libraries of blocks supported for HDL code generation in the **HDL Coder**, **Communications Toolbox HDL Support**, and **DSP System Toolbox HDL Support** block libraries.

To create a library of HDL-supported blocks from all your installed products, enter `hdlLib` at the MATLAB command line. This command requires an HDL Coder license.

Refer to the "Extended Capabilities > HDL Code Generation" section of each block page for block implementations, properties, and restrictions for HDL code generation.

You can also view blocks that are supported for HDL code generation in documentation by filtering the block reference list. Click **Blocks** in the blue bar at the top of the Help window, then select the **HDL code generation** check box at the bottom of the left column. The blocks are listed in their respective products. You can use the table of contents in the left column to navigate between products and categories.

Documentation All Examples Functions **Blocks** Apps Search Help

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« Blocks

Category

DSP System Toolbox

- Signal Generation, Manipulation, and Analysis 21
- Filter Implementation 10
- Transforms and Spectral Analysis 3
- Statistics and Linear Algebra 3
- Fixed-Point Design 8

HDL Coder
HDL Verifier
LTE HDL Toolbox
Mixed-Signal Blockset
SerDes Toolbox
SimEvents
Simulink Test

Extended Capability

- C/C++ Code Generation 34
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DSP System Toolbox — Blocks

By Category | [Alphabetical List](#)

i Results are filtered

Signal Generation, Manipulation, and Analysis

Signal Operations

Downsample	Resample input at lower rate by deleting samples
Repeat	Resample input at higher rate by repeating values
Sample and Hold	Sample and hold input signal
Upsample	Resample input at higher rate by inserting zeros
DC Blocker	Block DC component

Signal Generation

Constant	Generate constant value
NCO	Generate real or complex sinusoidal signals
NCO HDL Optimized	Generate real or complex sinusoidal signals—optimized for HDL code generation
Sine Wave	Generate continuous or discrete sine wave

Scopes and Data Logging

Spectrum Analyzer	Display frequency spectrum
Time Scope	Display and analyze signals generated during simulation and log signal data to MATLAB
Matrix Viewer	Display matrices as color images
Waterfall	View vectors of data over time
To Workspace	Write data to MATLAB workspace

System Objects

To find System objects supported for HDL code generation, see Predefined System Objects (HDL Coder).

Wireless Communications Design for FPGAs and ASICs

In this section...

“From Mathematical Algorithm to Hardware Implementation” on page 19-7

“HDL-Optimized Blocks” on page 19-10

“Reference Applications” on page 19-10

“Generate HDL Code and Prototype on FPGA” on page 19-11

Deploying algorithmic models to FPGA hardware makes it possible to do over-the-air testing and verification. However, designing wireless communications systems for hardware requires design tradeoffs between hardware resources and throughput. You can speed up hardware design and deployment by using HDL-optimized blocks that have hardware-suitable interfaces and architectures, reference applications that implement portions of the LTE physical layer, and automatic HDL code generation. You can also use hardware support packages to assist with deploying and verifying your design on real hardware.

MathWorks HDL products, such as LTE HDL Toolbox, allow you to start with a mathematical model, such as MATLAB code from LTE Toolbox™ or 5G Toolbox™, and design a hardware implementation of that algorithm that is suitable for FPGAs and ASICs.

From Mathematical Algorithm to Hardware Implementation

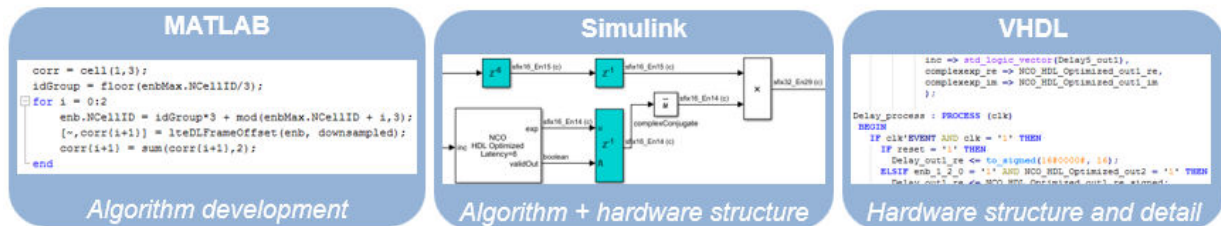
Wireless communications design often starts with algorithm development and testing using MATLAB functions. MATLAB code, which usually operates on matrices of floating-point data, is good for developing mathematical algorithms, manipulating large data sets, and visualizing data.

Hardware engineers typically receive a mathematical specification from an algorithm team, and reimplement the algorithm for hardware. Hardware designs require tradeoffs of resource usage for clock speed and overall throughput. Usually this tradeoff means operating on streaming data, and using some logic to control the storage and flow of data. Hardware engineers usually work in hardware description languages (HDLs), like VHDL and Verilog, that provide cycle-based modeling and parallelism.

To bridge this gap between mathematical algorithm and hardware implementation, use the MATLAB algorithm model as a starting point for hardware implementation. Make incremental changes to the design to make it suitable for hardware, and progress towards

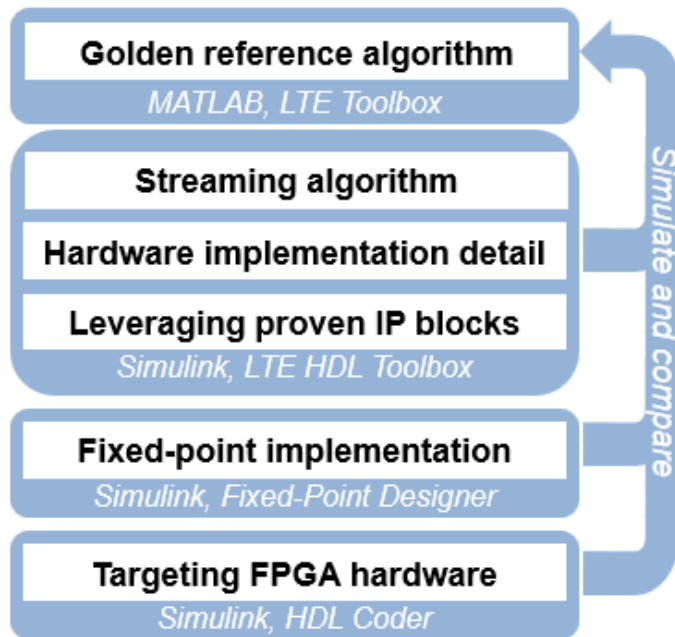
a Simulink model that you can use to automatically generate HDL code by using HDL Coder.

This diagram shows the design progression from mathematical algorithm in MATLAB, to hardware-compatible implementation in Simulink, and then the generated VHDL code.

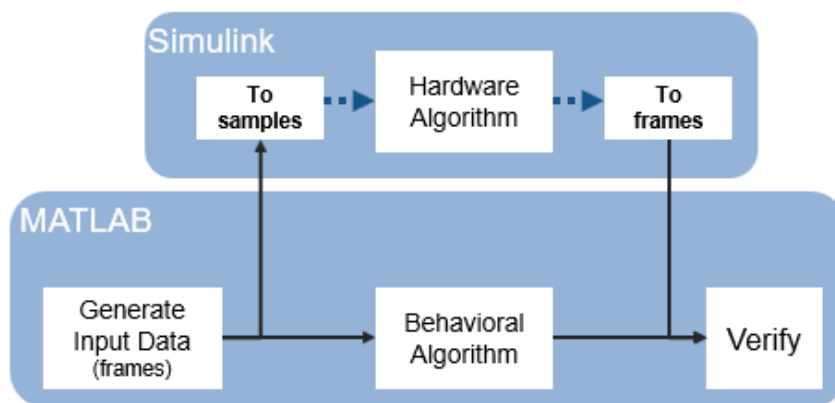


While both MATLAB and Simulink support automatic generation of HDL code, you must construct your design with hardware requirements in mind, and Simulink is better-suited for cycle-based modeling for hardware. It can represent parallel data paths and streaming data with control signals to manage the timing of the data stream. To aid in fixed-point type choices, it clearly visualizes data type propagation in the design. It also allows for easy pipelining of mathematical operations to improve maximum clock frequency in hardware.

While you create your hardware-ready design, use the MATLAB algorithm as a "golden reference" to verify that each version of the design still meets the mathematical requirements. The workflow shown in the diagram uses MATLAB and Simulink as collaboration and communication tools between the algorithm and hardware design teams.



For instance, when designing for LTE or 5G wireless standards, you can use LTE Toolbox and 5G Toolbox functions to create a golden reference in MATLAB. Then transition to Simulink and create a hardware-compatible implementation by using library blocks from LTE HDL Toolbox and blocks from Communications Toolbox and DSP System Toolbox that support HDL code generation. You can reuse test and data generation infrastructure from MATLAB by importing data from MATLAB to your Simulink model and returning the output of the model to MATLAB to verify it against the "golden reference".



HDL-Optimized Blocks

Library blocks from LTE HDL Toolbox implement encoders, decoders, modulators, demodulators, and sequence generators for use in an LTE, 5G, or general wireless communications system. These blocks use a standard streaming data interface for hardware. This interface makes it easy to connect parts of the algorithm together, and includes control signals that manage the flow of data and mark frame boundaries. These blocks support automatic HDL code generation with HDL Coder. You can also use blocks from Communications Toolbox and DSP System Toolbox that support HDL code generation.

The blocks provide hardware-suitable architectures that optimize resource use, such as including adder and multiplier pipelining to fit well into FPGA DSP slices. They also support automatic and configurable fixed-point data types. Using predefined blocks also allows you to try different parameter configurations without changing the rest of the design.

For lists of blocks that support HDL code generation, see [LTE HDL Toolbox Block List \(HDL Code Generation\)](#), [Communications Toolbox Block List \(HDL Code Generation\)](#), and [DSP System Toolbox Block List \(HDL Code Generation\)](#).

Reference Applications

LTE HDL Toolbox provides reference applications that contain hardware-ready implementations of large parts of the LTE physical layer. These designs are verified

against the "golden reference" functions provided by LTE Toolbox. They have also been tested on FPGA boards to confirm that they encode and decode over-the-air waveforms and use a reasonable amount of hardware resources. They are designed to be modular, scalable, and extensible so you can insert additional physical channels. The receiver design was tested using LTE waveforms captured off-the-air on three different continents.

The suite of LTE reference applications includes:

- Primary and secondary synchronization signal (PSS/SSS) generation and detection
- Downlink shared control channel detector and master information block (MIB) generation and recovery
- First system information block (SIB1) decoder
- Hardware-software interface models for MIB and SIB1 bit parsing and channel estimation data indexing
- Waveform generation for multiple-antenna transmission
- Support for FDD and TDD for both transmitter and receiver applications

These reference applications can be used as-is to deliver packet information to your unique application and to generate synthesizable VHDL or Verilog with HDL Coder. They also serve as examples to illustrate recommended practices for implementing communications algorithms on FPGA or ASIC hardware.

Generate HDL Code and Prototype on FPGA

LTE HDL Toolbox™ provides blocks that support HDL code generation. To generate HDL code from designs that use these blocks, you must have an HDL Coder license. HDL Coder produces device-independent code with signal names that correspond to the Simulink model. HDL Coder also provides a tool to drive the FPGA synthesis and targeting process, and enables you to generate scripts and test benches for use with third-party HDL simulators.

To assist with the setup and targeting of programmable logic on a prototype board, and to verify your wireless communications system design on hardware, download a hardware support package such as Communications Toolbox Support Package for Xilinx Zynq-Based Radio.

See Also

External Websites

- [LTE HDL Toolbox](#)
- [HDL Coder](#)

Simulation Acceleration

Simulation Acceleration Using GPUs

In this section...

“GPU-Based System objects” on page 20-2

“General Guidelines for Using GPUs” on page 20-3

“Transmit and decode using BPSK modulation and turbo coding” on page 20-3

“Process Multiple Data Frames Using a GPU” on page 20-4

“Process Multiple Data Frames Using NumFrames Property” on page 20-5

“gpuArray and Regular MATLAB Numerical Arrays” on page 20-6

“Pass gpuArray as an Input” on page 20-6

“System Block Support for GPU System Objects” on page 20-7

GPU-Based System objects

GPU-based System objects look and behave much like the other System objects in the Communications Toolbox product. The important difference is that the algorithm is executed on a Graphics Processing Unit (GPU) rather than on a CPU. Using the GPU can accelerate your simulation.

System objects for the Communications Toolbox product are located in the `comm` package and are constructed as:

```
H = comm.<object name>
```

For example, a Viterbi Decoder System object is constructed as:

```
H = comm.ViterbiDecoder
```

In cases where a corresponding GPU-based implementation of a System object exists, they are located in the `comm.gpu` package and constructed as:

```
H = comm.gpu.<object name>
```

For example, a GPU-based Viterbi Decoder System object is constructed as:

```
H = comm.gpu.ViterbiDecoder
```

To see a list of available GPU-based implementations enter `help comm` at the MATLAB command line and click **GPU Implementations**.

General Guidelines for Using GPUs

Graphics Processing Units (GPUs) excel at processing large quantities of data and performing computations with high compute intensity. Processing large quantities of data is one way to maximize the throughput of your GPU in a simulation. The amount of the data that the GPU processes at any one time depends on the size of the data passed to the input of a GPU System object. Therefore, one way to maximize this data size is by processing multiple frames of data.

You can use a single GPU System object to process multiple data frames simultaneously or in parallel. This differs from the way many of the standard, or non-GPU, System objects are implemented. For GPU System objects, the number of frames the objects process in a single call to the object function is either implied by one of the object properties or explicitly stated using the `NumFrames` property on the objects.

Transmit and decode using BPSK modulation and turbo coding

This example shows how to transmit turbo-encoded blocks of data over a BPSK-modulated AWGN channel. Then, it shows how to decode using an iterative turbo decoder and display errors.

Define a noise variable, establish a frame length of 256, and use the random stream property so that the results are repeatable.

```
noiseVar = 4; frmLen = 256;
s = RandStream('mt19937ar', 'Seed', 11);
intrlvrIndices = randperm(s, frmLen);
```

Create a Turbo Encoder System object. The trellis structure for the constituent convolutional code is `poly2trellis(4, [13 15 17], 13)`. The `InterleaverIndices` property specifies the mapping the object uses to permute the input bits at the encoder as a column vector of integers.

```
turboEnc = comm.TurboEncoder('TrellisStructure', poly2trellis(4, ...
    [13 15 17], 13), 'InterleaverIndices', intrlvrIndices);
```

Create a BPSK Modulator System object.

```
bpsk = comm.BPSKModulator;
```

Create an AWGN Channel System object.

```
channel = comm.AWGNChannel('NoiseMethod', 'Variance', 'Variance', ...  
    noiseVar);
```

Create a GPU-Based Turbo Decoder System object. The trellis structure for the constituent convolutional code is `poly2trellis(4, [13 15 17], 13)`. The `InterleaverIndices` property specifies the mapping the object uses to permute the input bits at the encoder as a column vector of integers.

```
turboDec = comm.gpu.TurboDecoder('TrellisStructure', poly2trellis(4, ...  
    [13 15 17], 13), 'InterleaverIndices', intrlvrIndices, ...  
    'NumIterations', 4);
```

Create an Error Rate System object.

```
errorRate = comm.ErrorRate;
```

Run the simulation.

```
for frmIdx = 1:8  
    data = randi(s, [0 1], frmLen, 1);  
    encodedData = turboEnc(data);  
    modSignal = bpsk(encodedData);  
    receivedSignal = channel(modSignal);
```

Convert the received signal to log-likelihood ratios for decoding.

```
receivedBits = turboDec(-2/(noiseVar/2))*real(receivedSignal));
```

Compare original the data to the received data and then calculate the error rate results.

```
errorStats = errorRate(data, receivedBits);  
end  
fprintf('Error rate = %f\nNumber of errors = %d\nTotal bits = %d\n', ...  
    errorStats(1), errorStats(2), errorStats(3))
```

Process Multiple Data Frames Using a GPU

This example shows how to simultaneously process two data frames using an LDPC Decoder System object. The `ParityCheckMatrix` property determines the frame size. The number of frames that the object processes is determined by the frame size and the input data vector length.


```

numframes = 2;

ldpcEnc = comm.LDPCDecoder;
ldpcGPUDec = comm.gpu.LDPCDecoder;
ldpcDec = comm.LDPCDecoder;

msg = randi([0 1], 32400,2);

for ii=1:numframes,
    encout(:,ii) = ldpcEnc(msg(:,ii));
end

%single ended to bipolar (for LLRs)
encout = 1-2*encout;

%Decode on the CPU
for ii=1:numframes;
    cout(:,ii) = ldpcDec(encout(:,ii));
end

%Multiframe decode on the GPU
gout = ldpcGPUDec(encout(:));

%check equality
isequal(gout,cout(:))

```

Process Multiple Data Frames Using NumFrames Property

This example shows how to process multiple data frames using the NumFrames property of the GPU-based Viterbi Decoder System object. For a Viterbi Decoder, the frame size of your system cannot be inferred from an object property. Therefore, the NumFrames property defines the number of frames present in the input data.

```

numframes = 10;

convEncoder = comm.ConvolutionalEncoder('TerminationMethod', 'Terminated');
vitDecoder = comm.ViterbiDecoder('TerminationMethod', 'Terminated');

%Create a GPU Viterbi Decoder, using NumFrames property.
vitGPUDecoder = comm.gpu.ViterbiDecoder('TerminationMethod', 'Terminated', ...
    'NumFrames', numframes );

msg = randi([0 1], 200, numframes);

for ii=1:numframes,

```

```
    convEncOut(:,ii) = 1-2*convEncoder(msg(:,ii));  
end  
  
%Decode on the CPU  
for ii=1:numframes;  
    cVitOut(:,ii) = vitDecoder(convEncOut(:,ii));  
end  
  
%Decode on the GPU  
gVitOut = vitGPUDecoder(convEncOut(:));  
  
isequal(gVitOut,cVitOut(:))
```

gpuArray and Regular MATLAB Numerical Arrays

A GPU-based System object accepts typical MATLAB arrays or objects created using the `gpuArray` class. A GPU-based System object supports input signals with double- or single-precision data types. The output signal inherits its data type from the input signal.

- If the input signal is a MATLAB array, the System object handles data transfer between the CPU and the GPU. The output signal is a MATLAB array.
- If the input signal is a `gpuArray`, the data remains on the GPU. The output signal is a `gpuArray`. When the object is given a `gpuArray`, calculations take place entirely on the GPU, and no data transfer occurs. Passing `gpuArray` arguments provides increased performance by reducing simulation time. For more information, see “Establish Arrays on a GPU” (Parallel Computing Toolbox).

Passing MATLAB arrays to a GPU System object requires transferring the initial data from a CPU to the GPU. Then, the GPU System object performs calculations and transfers the output data back to the CPU. This process introduces latency. When data in the form of a `gpuArray` is passed to a GPU System object, the object does not incur the latency from data transfer. Therefore, a GPU System object runs faster when you supply a `gpuArray` as the input.

In general, you should try to minimize the amount of data transfer between the CPU and the GPU in your simulation.

Pass gpuArray as an Input

This example shows how to pass a `gpuArray` to the input of the PSK modulator, reducing latency.

```
pskGPUModulator = comm.gpu.PSKModulator;  
x = randi([0 7], 1000, 1, 'single');
```

```
gx = gpuArray(x);  
  
o = pskGPUModulator(x);  
class(o)  
  
release(pskGPUModulator); %allow input types to change  
  
go = pskGPUModulator(gx);  
class(go)
```

System Block Support for GPU System Objects

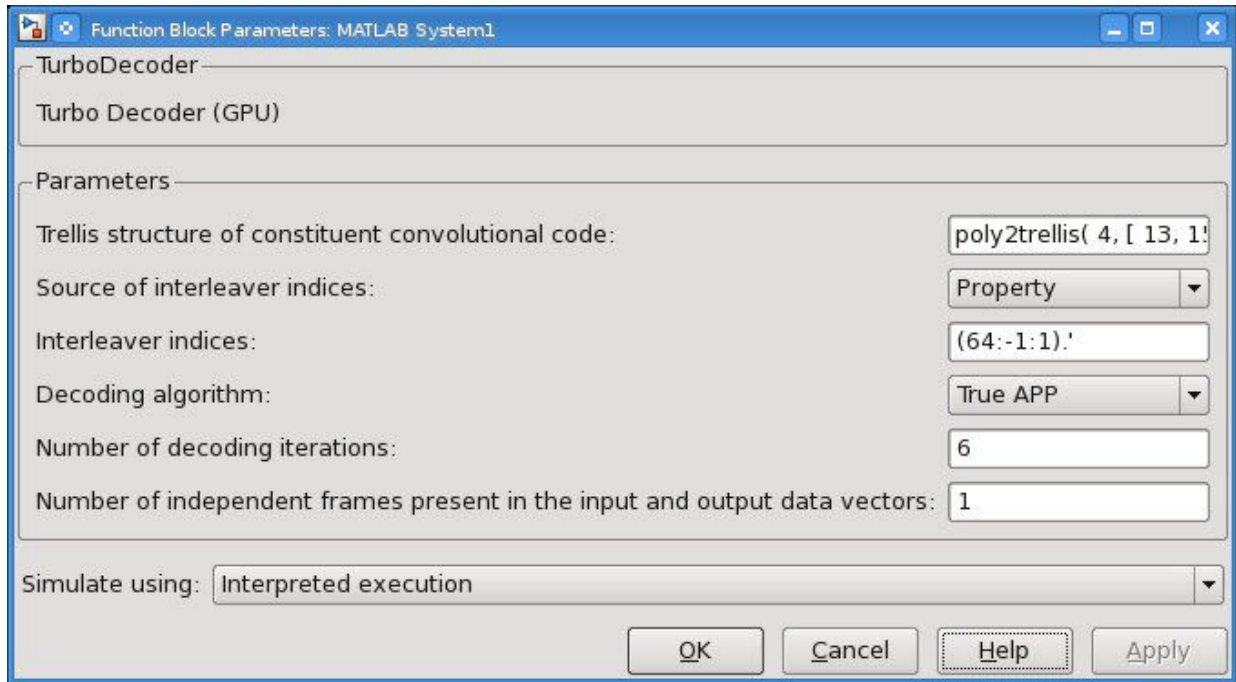
- “GPU System Objects Supported in System Block” on page 20-7
- “System Block Limitations for GPU System Objects” on page 20-7

GPU System Objects Supported in System Block

- `comm.gpu.AWGNChannel`
- `comm.gpu.BlockDeinterleaver`
- `comm.gpu.BlockInterleaver`
- `comm.gpu.ConvolutionalDeinterleaver`
- `comm.gpu.ConvolutionalEncoder`
- `comm.gpu.ConvolutionalInterleaver`
- `comm.gpu.PSKDemodulator`
- `comm.gpu.PSKModulator`
- `comm.gpu.TurboDecoder`
- `comm.gpu.ViterbiDecoder`

System Block Limitations for GPU System Objects

The GPU System objects must be simulated using Interpreted Execution. You must select this option explicitly on the block mask; the default value is Code generation.



See Also

More About

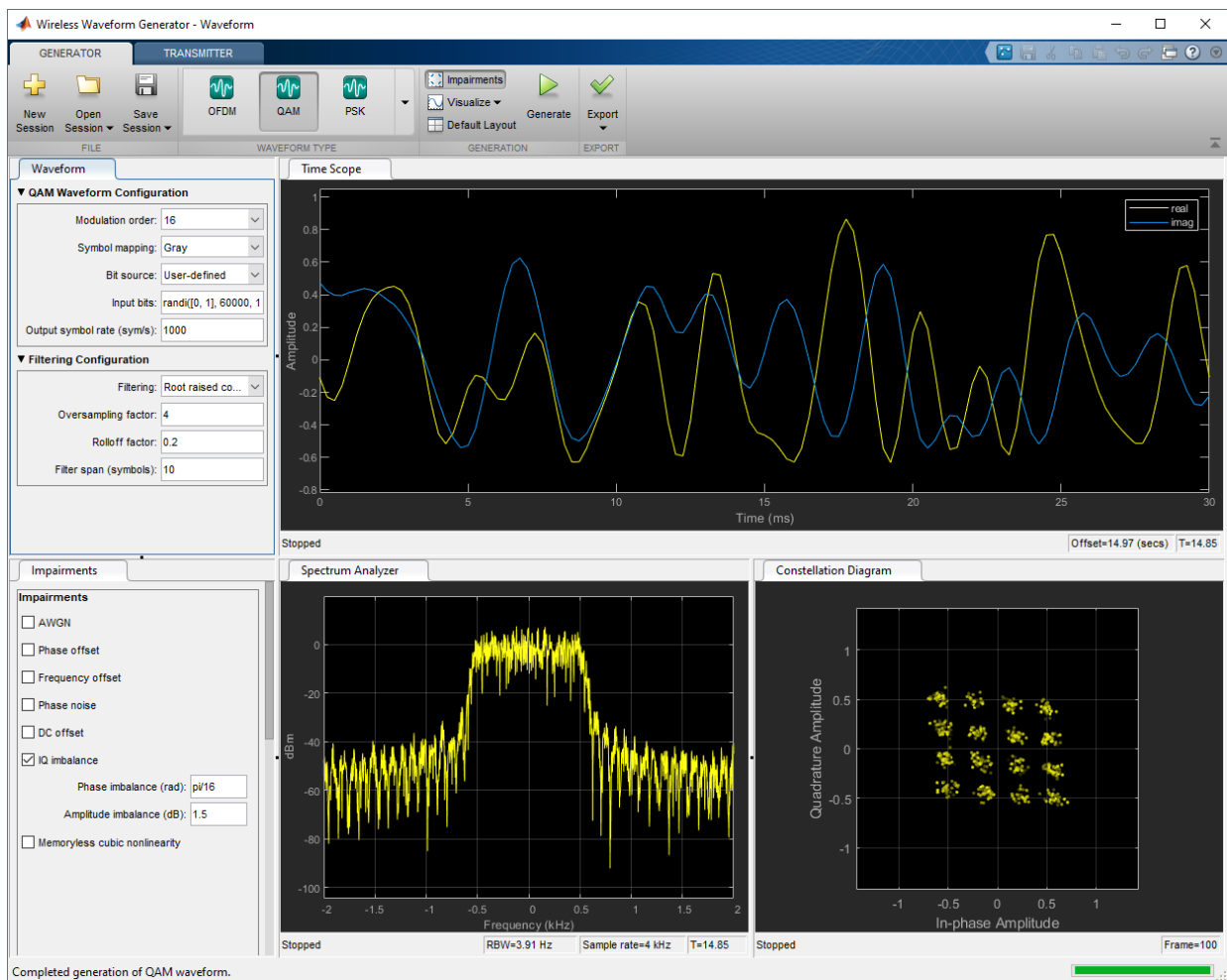
- "GPU Capabilities and Performance" (Parallel Computing Toolbox)

Wireless Waveform Generator App

Using Wireless Waveform Generator App

The **Wireless Waveform Generator** app is an interactive tool for creating, impairing, visualizing, and exporting modulated waveforms.

Start the app. On the **Apps** tab in the MATLAB toolstrip, under **Signal Processing and Communications**, click the app icon. You can also start the app by entering `wirelessWaveformGenerator` at the MATLAB command prompt.



These are the typical workflows when using the **Wireless Waveform Generator** app.

- Generate a waveform.
 - Select the desired waveform type from the options available in the **Waveform Type** section. Adjust the configuration parameters in the **Waveform** pane. For more information, see “Waveform Type” on page 21-3.
 - Select **Impairments** in the **Generation** section to open the **Impairments** pane. Adjust the configuration parameters in the **Impairments** pane. For more information, see “Add Impairments” on page 21-4.
 - To perform signal generation, click **Generate**. After generation, the waveform is displayed. You can adjust the waveform, filtering, and impairment configurations and then regenerate the waveform.
 - The default visualization plots vary based on the waveform type selected. Additional “Visualization Options” on page 21-4 can be opened by selecting them from **Visualize** in the **Generation** section.
- After generating a waveform you can export it by selecting the desired options from **Export** in the **Export** section. For more information, see “Export Waveform” on page 21-5.
- You can save the current session, open a previously saved session, or open a new session by selecting the desired option in the **File** section. For more information, see “Waveform Generator Session” on page 21-6.

Waveform Type

To generate the various available waveforms, the **Wireless Waveform Generator** app uses Communications Toolbox features. The supported waveform types include:

- OFDM — The app uses the `comm.OFDMModulator` System object to generate this type of waveform.
- QAM — The app uses the `qammod` function to generate this type of waveform.
- PSK — The app uses the `pskmod` function to generate this type of waveform.
- Sinewave — The app uses the `dsp.SineWave` System object to generate this type of waveform.
- Bluetooth — You can download the Communications Toolbox Library for the Bluetooth Protocol Add-On and generate Bluetooth modulated waveforms using features described in “Bluetooth”.

- WLAN — If you have the WLAN Toolbox™ you can also generate 802.11™ modulated waveforms using features in the “WLAN Toolbox”.
- LTE — If you have the LTE Toolbox you can also generate LTE modulated waveforms using features in the “LTE Toolbox”.

By default, generated waveforms have no filtering applied. To apply filtering to the waveform, select the desired filter option from the **Filtering** parameter on the **Waveform** pane. The available filter options vary based on the waveform type you select.

Add Impairments

You can add these impairments to the waveform that you generate.

- AWGN — The app uses the `awgn` function to impair the waveform.
- Phase offset — The app impairs the waveform by applying the specified phase offset as $y = xe^{j\phi}$, where ϕ is the phase offset in radians.
- Frequency offset — The app uses the `comm.PhaseFrequencyOffset` System object to impair the waveform.
- Phase noise — The app uses the `comm.PhaseNoise` System object to impair the waveform.
- DC offset — The app impairs the waveform by applying the specified DC offset as $y = x + dcOff$, where `dcOff` is the complex DC offset in Volts.
- IQ imbalance — The app uses the `iqimbal` function to impair the waveform..
- Memoryless cubic nonlinearity — The app uses the `comm.MemorylessNonlinearity` System object to impair the waveform.

Visualization Options

You can use these plot types to visualize waveforms that you generate.

- Spectrum Analyzer — The app plots the waveform in the frequency domain.
- OFDM Grid — For OFDM waveforms, the app plots the resource allocation of data and control channels.
- Time scope — The app plots the inphase and quadrature (IQ) waveform samples in the time domain.
- Constellation diagram — The app plots the constellation points of the modulation symbols.

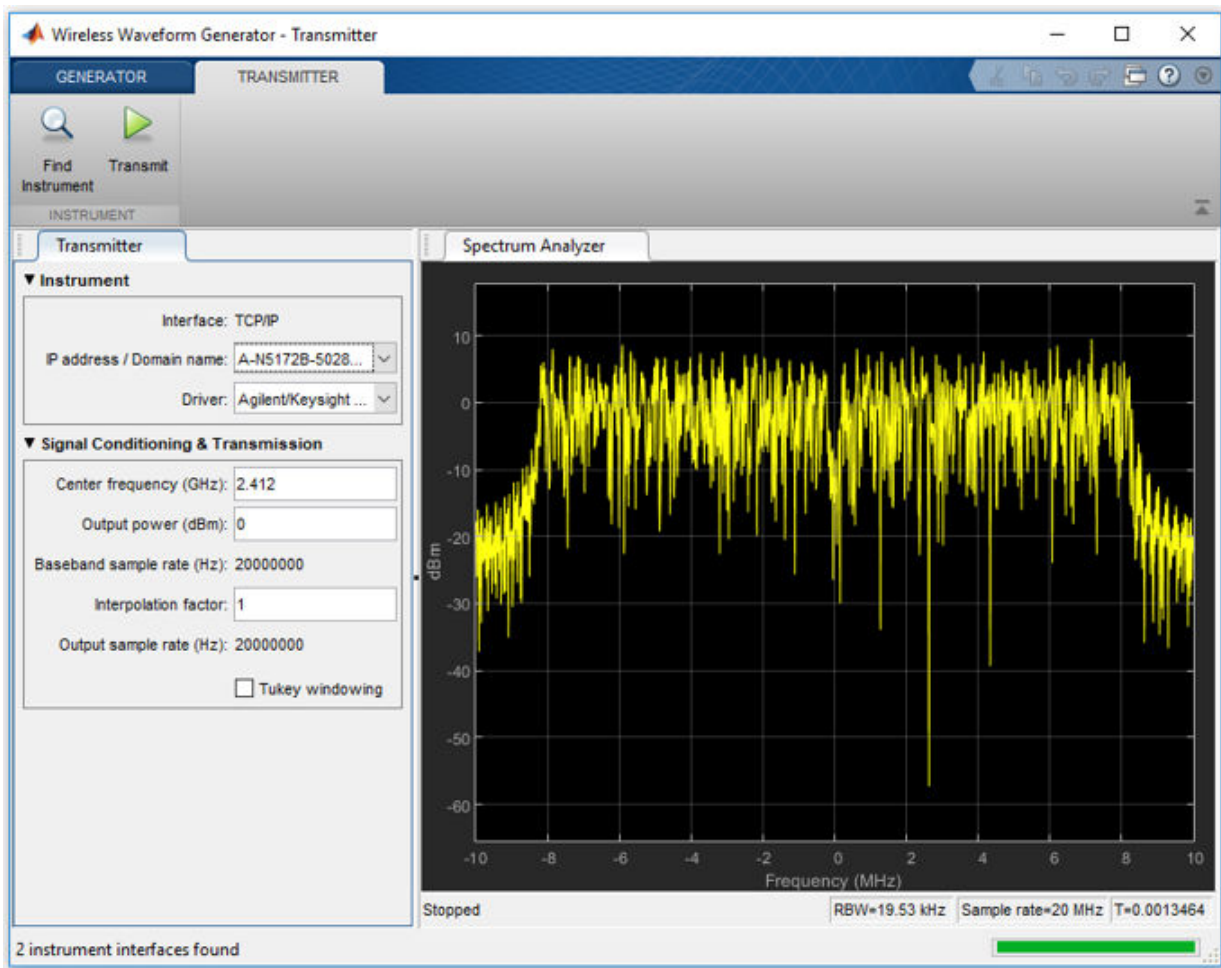
Export Waveform

You can export the waveform to a runnable MATLAB script, to your workspace, or to a signal file.

- Waveforms exported to a runnable MATLAB script can be used to reproduce your waveform outside of the **Wireless Waveform Generator** app by running the script.
- Waveforms exported to the workspace are saved as a structure containing these fields:
 - `type` — This field is a character vector indicating the waveform type.
 - `config` — This field is a structure or object containing fields that specify the configured waveform type.
 - `Fs` — This field is the signal sample rate in Hertz.
 - `waveform` — The field is the complex waveform samples output as an N_S -by-1 column vector or an N_S -by- N_T matrix. N_S is the number of time-domain samples, and N_T is the number of transmit antennas.
- Waveforms exported to a signal file can be saved as a `.mat` or `.bb` file.
 - MAT-files are binary MATLAB files that store workspace variables. For more information, see “MAT-File Versions” (MATLAB).
 - The app uses the `comm.BasebandFileWriter` System object to save `.bb` files.

Transmit Using Lab Test Instrument

Generate a waveform that you can transmit using a connected lab test instrument. The **Wireless Waveform Generator** app can transmit using instruments supported by the `rfsiggen` function.



Use of the transmit feature in the **Wireless Waveform Generator** app requires “Instrument Control Toolbox”.

Waveform Generator Session

You can save the current session, open a previously saved session, or open a new session by selecting the desired option in the **FILE** section. When you save a waveform generator

session, the session configuration is saved as a *.mat* file. For more information, see “MAT-File Versions” (MATLAB).

See Also

Wireless Waveform Generator

RF Propagation

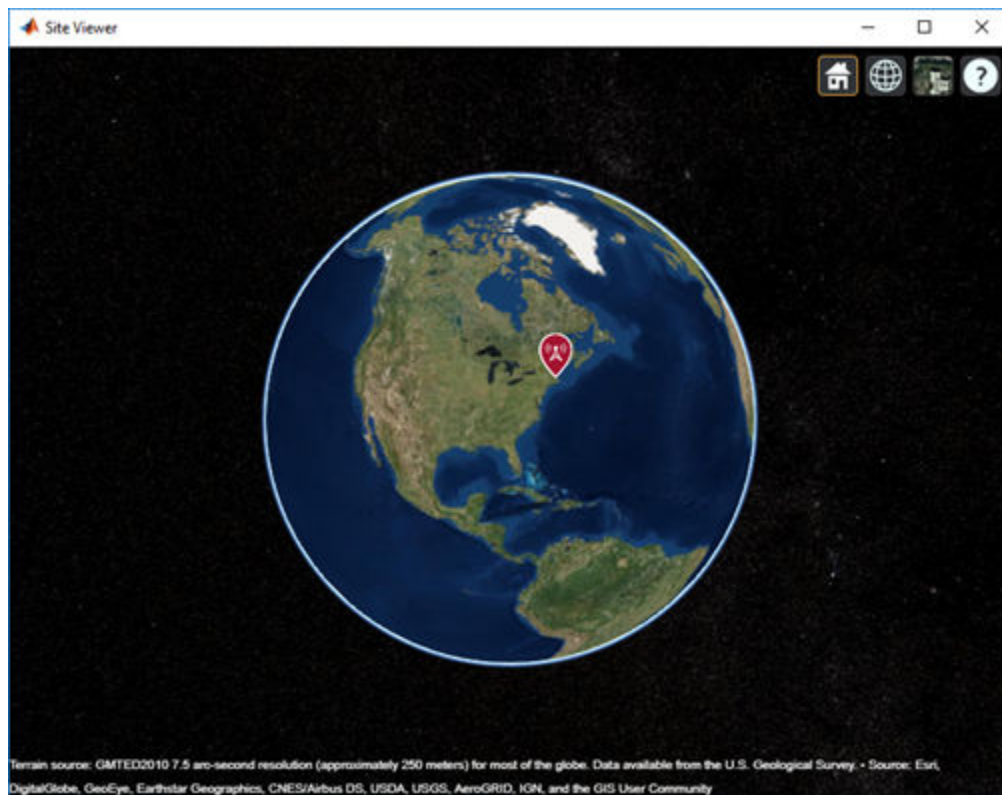
- “Site Viewer” on page 22-2
- “Troubleshooting Site Viewer” on page 22-11
- “Access Basemaps and Terrain in Site Viewer” on page 22-13
- “Access TIREM Software” on page 22-15

Site Viewer

In this section...
“Introduction” on page 22-2
“Dimension Picker” on page 22-3
“Basemaps” on page 22-4
“View Latitude and Longitude Coordinates of Location” on page 22-8
“Limitations for Terrain and Buildings” on page 22-9

Introduction

Transmitter and receiver site for RF propagation visualization involves a map display called Site Viewer. You can use the **Home** button to view the default home location of Site Viewer.



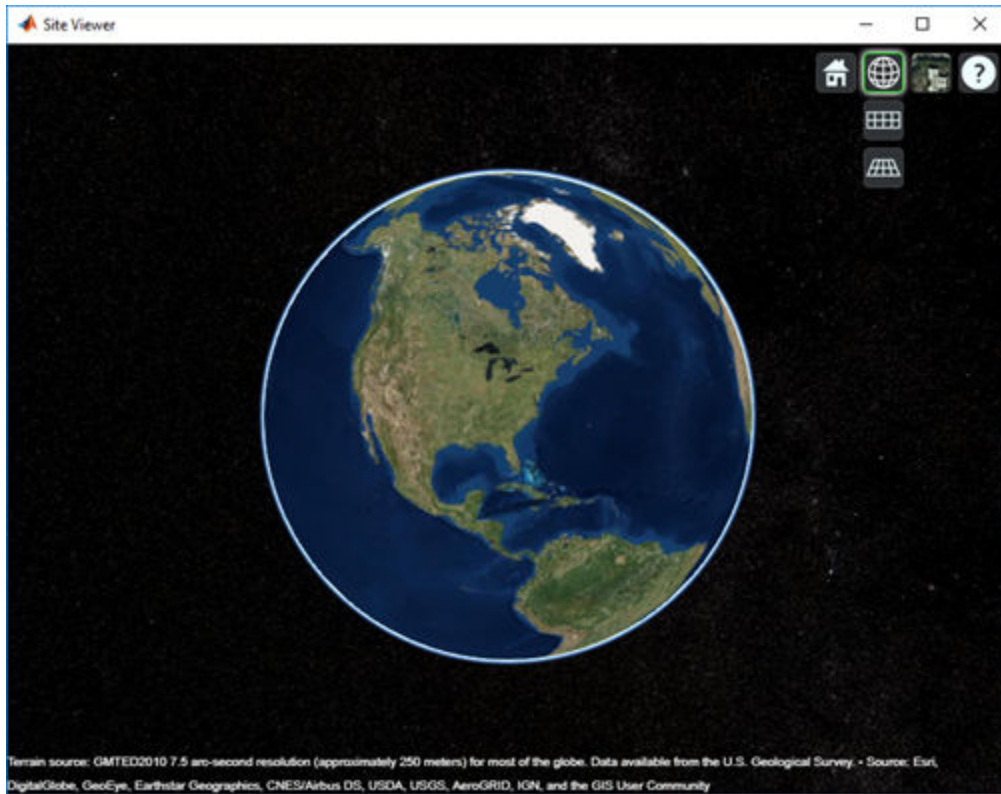
Site Viewer includes in-canvas pan, zoom, and rotation.

Site Viewer is a 3-D map display and requires hardware graphics support for WebGL™.

Opening Site Viewer using `siteviewer` allows customization of basemap, terrain, and buildings.

Dimension Picker

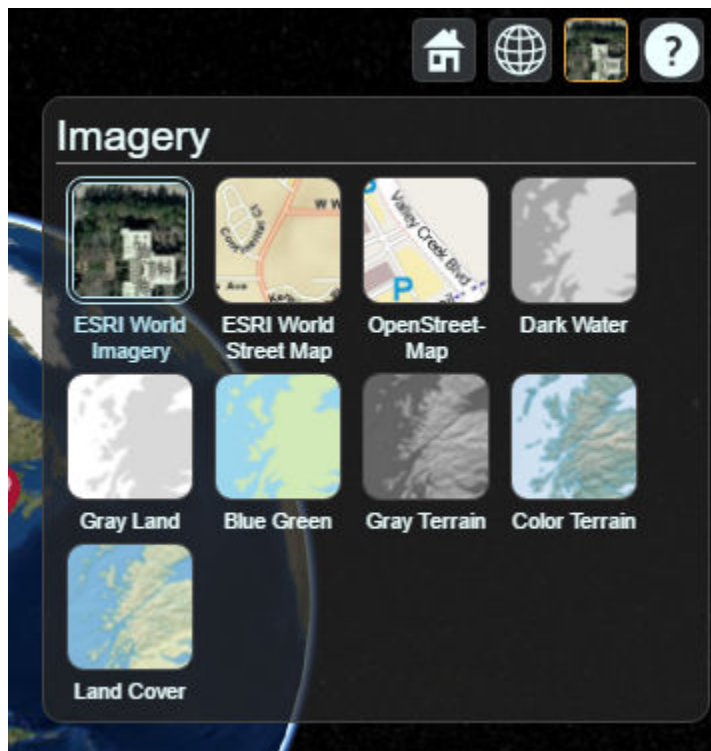
The dimension picker offers three choices: 3-D, 2-D, and Columbus View.



- 3-D View - Provides a smooth globe. This is the default view.
- 2-D View - Provides a flat map in Mercator projection.
- Columbus View - Provides a flat map in Mercator projection that allows tilt and rotation.

Basemaps

Basemaps are the underlying maps used by Site Viewer.



You can select the basemaps using the basemap picker. You need an active internet connection for all basemaps except Dark Water. The following options are available:




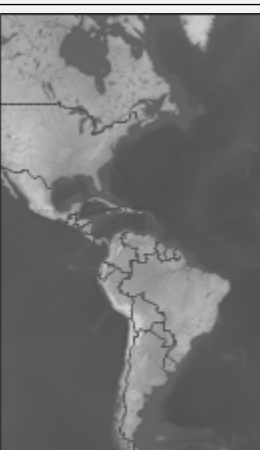
- Satellites
- Streets
- OpenStreetMap
- Dark Water
- Gray Land
- Blue Green
- Gray Terrain
- Color Terrain
- Land Cover



By default, Site Viewer uses **Satellites**. You need an active internet connection for all basemaps except **Dark Water**.

Use **Satellites** for high-quality satellite imagery. Use **Streets**, or **OpenStreetMap** for imagery of roads, labels, and political boundaries.

Use **The MathWorks® natural earth** basemaps for low resolution but more controls and color scheme options in basemaps.

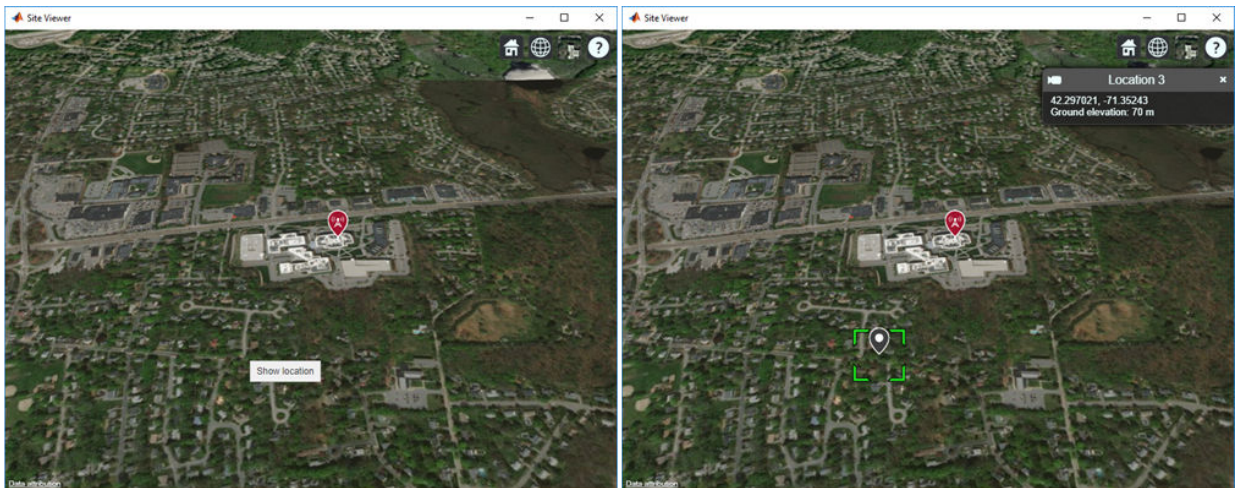
Basemaps

Two-Tone Basemaps	Description	Natural Earth Basemaps	Description
	<p>'darkwater'</p> <p>Land areas: light-to-moderate gray</p> <p>Ocean and water areas: darker gray</p> <p>This is the default map.</p>		<p>'colorterrain'</p> <p>Shaded relief combined with custom elevation colors based on climate—humid lowlands are green and arid lowlands brown.</p>
	<p>'grayland'</p> <p>Land areas: light-to-moderate gray land</p> <p>Ocean and water areas: white</p>		<p>'grayterrain'</p> <p>Worldwide terrain depicted monochromatically in shades of gray, combining shaded relief that emphasizes both high mountains and the micro terrain found in lowlands.</p>

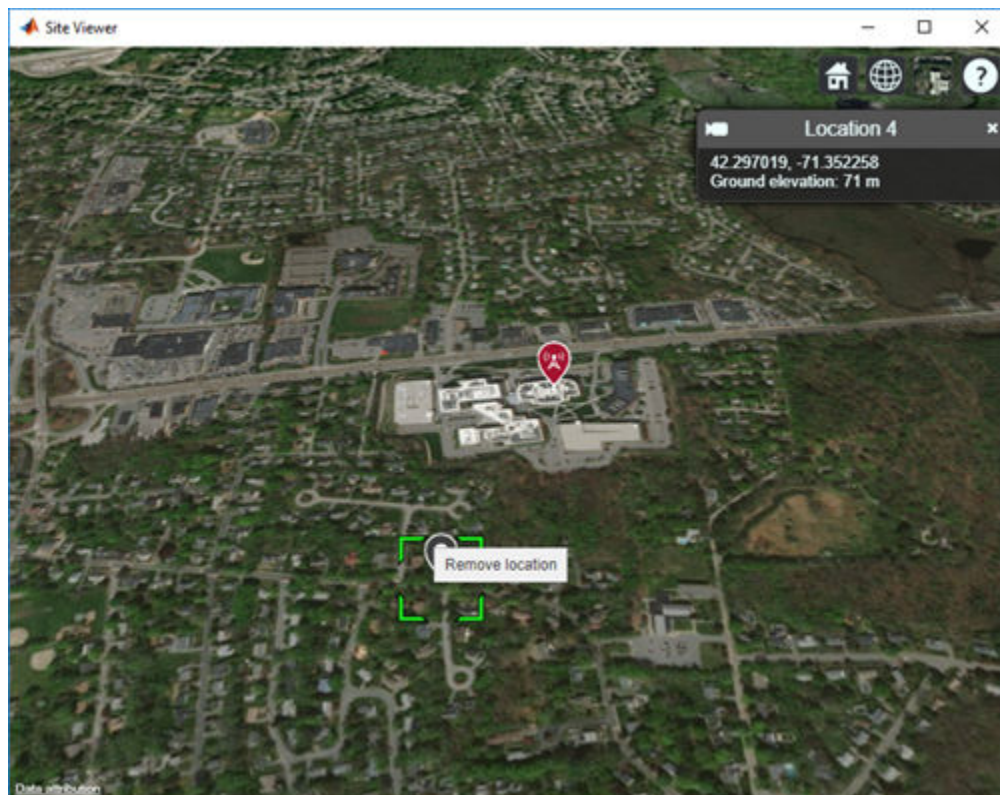
Two-Tone Basemaps	Description	Natural Earth Basemaps	Description
	<p>'bluegreen'</p> <p>Land areas: light green</p> <p>Ocean and water areas: light blue</p>		<p>'Landcover'</p> <p>Satellite-derived land cover data and shaded relief presented with a light, natural palette suitable for making thematic and reference maps (includes ocean-bottom relief).</p>

View Latitude and Longitude Coordinates of Location

Use Site Viewer to view the latitude and longitude coordinates of any location. To view the latitude and longitude, right-click the map and select **Show Location**.



To remove the location, right-click the location and select **Remove Location**.



Limitations for Terrain and Buildings

Terrain

- Default terrain access requires internet connection. If no internet connection exists, then Site Viewer automatically uses 'none' in the property Terrain.
- Custom DTED terrain files for use with `addCustomTerrain` must be acquired outside of MATLAB for example by using USGS EarthExplorer.
- When using custom terrain, analysis is restricted to the terrain region. For example, an error occurs if trying to show a txsite or rxsite outside of the region.

Buildings

- OpenStreetMap files obtained from <https://www.openstreetmap.org> represent crowd-sourced map data, and the completeness and accuracy of the buildings data may vary depending on the map location.
- When downloading data from <https://www.openstreetmap.org>, select an export area larger than the desired area to ensure that all expected building features are fully captured. Building features at the edge of the selected export area may be missing.
- Building geometry and features are interpreted from the file according to the recommendations of OpenStreetMap for 3-D buildings.

See Also

opengl

More About

- “Troubleshooting Site Viewer” on page 22-11
- “Access Basemaps and Terrain in Site Viewer” on page 22-13
- “System Requirements for Graphics” (MATLAB)
- “Resolving Low-Level Graphics Issues” (MATLAB)

Troubleshooting Site Viewer

In this section...

“Internet Connection Failure” on page 22-11

“Graphics Environment” on page 22-11

Internet Connection Failure

When you start Site Viewer up on showing a txsite or rxsite object, a check is made to make sure that you have an internet connection to retrieve the default basemap and terrain data.

If Site Viewer cannot connect to the Internet the following warning messages are displayed:

- *Warning: Unable to access the Internet, showing Dark Water instead of Satellites. See Access Basemaps and Terrain in Site Viewer.*
- *Warning: Unable to access terrain data. See Access Basemaps and Terrain in Site Viewer.*

If Site Viewer cannot connect to the Internet, then terrain data is not used and the "Dark Water" basemap is selected.

Graphics Environment

Site Viewer can fail to open because of two reasons:

- In MATLAB, OpenGL[®] is set to software graphics. An error message is displayed in the command window, notifying you to upgrade the graphics hardware driver or select hardware graphics using OpenGL.

For more information, see `opengl`, and “Resolving Low-Level Graphics Issues” (MATLAB).

- When starting Site Viewer, JavaScript[®] for WebGL support fails. An error message is displayed in the command window, notifying you to update the graphics hardware driver.

For more information, see “Resolving Low-Level Graphics Issues” (MATLAB)

See Also

opengl

More About

- “Site Viewer” on page 22-2
- “Access Basemaps and Terrain in Site Viewer” on page 22-13
- “System Requirements for Graphics” (MATLAB)
- “Resolving Low-Level Graphics Issues” (MATLAB)

Access Basemaps and Terrain in Site Viewer

In this section...

“Access and Download Basemaps” on page 22-13

“Access Terrain” on page 22-14

You need an active internet connection for all basemaps except Dark Water.

Access and Download Basemaps

To download MATLAB basemaps:

- 1 On the MATLAB **Home** tab, in the **Environment** section, click **Add-Ons > Get Add-Ons**.
- 2 In the Add-On Explorer, scroll to the **MathWorks Features** section, and click **show all** to find the basemap packages. You can also search for the basemap add-ons by name (listed in the following table) or click **Features** in **Filter by Type**.
- 3 Select the basemap data packages. For more information about basemaps, see geobubble.

Basemap Name	Basemap Data Package Name
'bluegreen'	MATLAB Basemap Data - bluegreen
'grayland'	MATLAB Basemap Data - grayland
'colorterrain'	MATLAB Basemap Data - colorterrain
'grayterrain'	MATLAB Basemap Data - grayterrain
'landcover'	MATLAB Basemap Data - landcover

In addition, Site Viewer also supports three external basemaps that you can select from the basemap picker. The following options are available:

- Satellites
- Streets
- OpenStreetMap

Note If basemap does not render correctly in Site Viewer (for example only the ocean is visible), check if the basemap server supports CORS (cross-origin resource sharing). Site Viewer does not support basemaps that do not support CORS.

Access Terrain

To access terrain data for Site Viewer, you need an active internet connection.

See Also

More About

- “Site Viewer” on page 22-2
- “Troubleshooting Site Viewer” on page 22-11
- “System Requirements for Graphics” (MATLAB)
- “Resolving Low-Level Graphics Issues” (MATLAB)

Access TIREM Software

The Terrain Integrated Rough Earth Model™ (TIREM™) is a propagation model for computing the path loss for irregular terrain and seawater scenarios. TIREM is developed, trademarked, and licensed by Alion Science. To use TIREM, you need to acquire it from Alion Science.

TIREM is designed to calculate the reference basic median propagation loss (path loss) based on the terrain profile along the great circle path between two antennas, for example, using digital terrain elevation data (DTED). You can use TIREM model to calculate the point-to-point path loss between sites over irregular terrain. The model combines physics with empirical data to provide path loss estimates. The TIREM propagation model can predict path loss at frequencies between 1 MHz and 1 THz.

Use `tiremSetup` to enable TIREM access from within MATLAB. The TIREM library folder contains the `tirem3` shared library. The full library name is platform-dependent:

Platform	Shared Library Name
Windows	<code>libtirem3.dll</code> or <code>tirem3.dll</code>
Linux	<code>libtirem3.so</code>
Mac	<code>libtirem3.dylib</code>

See Also

`tiremSetup`

