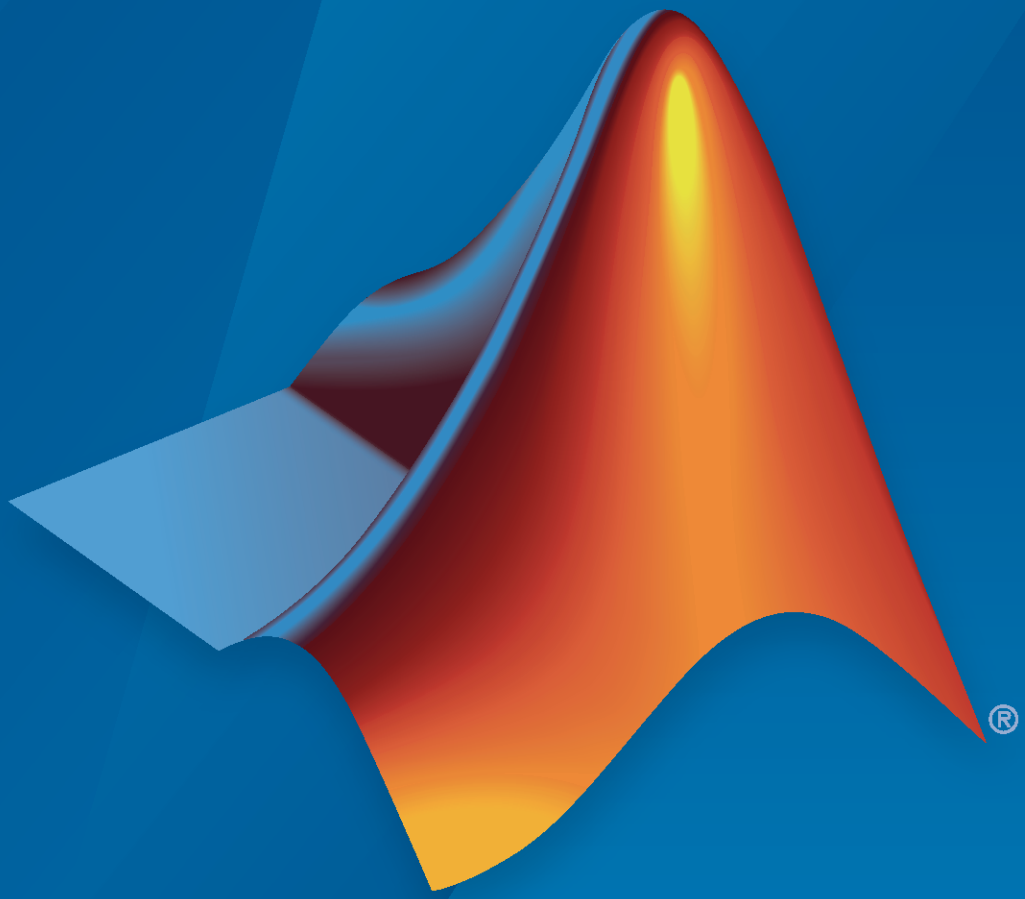
Audio Toolbox™ Release Notes



MATLAB® & SIMULINK®

MathWorks®

How to Contact MathWorks

Latest news: `www.mathworks.com`

Sales and services: `www.mathworks.com/sales_and_services`

User community: `www.mathworks.com/matlabcentral`

Technical support: `www.mathworks.com/support/contact_us`

Phone: 508-647-7000

The MathWorks, Inc.
1 Apple Hill Drive
Natick, MA 01760-2098

Audio Toolbox™ Release Notes

Contents

R2021a

R2020b

R2020a

R2019b

R2019a

R2018b

R2018a

R2017b

R2017a

R2016b

R2016a

R2021a

Version: 3.0

New Features

Bug Fixes

OpenL3 Pretrained Network: Extract deep audio embeddings with pretrained OpenL3 convolutional neural network

`openl3` produces audio embeddings that can be used as input to deep learning classification networks or general machine learning systems. The layers of `openl3` can also be incorporated as part of a larger network.

- Use the `openl3` function to return the pretrained neural network.
- Use the `openl3Preprocess` function to preprocess the audio signal and generate the input to the network.
- Use the `openl3Features` function to combine preprocessing and network inference to generate deep audio embeddings.

The `openl3` and `openl3Features` functions require Deep Learning Toolbox™.

CREPE: Deep learning pitch estimation

This release introduces the CREPE convolutional deep learning neural network and associated pre and postprocessing functions. Depending on the setup, CREPE can be integrated into a larger deep learning model or used as a feature extractor.

- Use the `crepe` function to return the pretrained neural network.
- Use the `crepePreprocess` function to preprocess the audio signal into a format acceptable by `crepe`.
- Use the `crepePostprocess` function to postprocess the output of the CREPE network and convert it to a pitch estimate.
- Use the `pitchnn` function to incorporate audio pre and postprocessing with network inference and produce the pitch estimation.

The `crepe` and `pitchnn` functions require Deep Learning Toolbox.

ivectorSystem: Produce compact representations of audio signals

Use `ivectorSystem` object to produce compact representations of audio signals for applications such as speaker verification, speaker identification, speaker diarization, speech emotion recognition, and acoustic machine fault detection.

The `ivectorSystem` object enables a streamlined workflow including system training, label enrolling and unenrolling, label verification, label identification, as well as open-set classification threshold determination.

vggishPreprocess and yamnetPreprocess

This release introduces functions that preprocess audio signals for classification tasks using the VGGish (`vggish`) or YAMNet (`yamnet`) pretrained neural networks. With the `vggishPreprocess` and `yamnetPreprocess` functions, users can process large audio sets through the networks more efficiently.

Enhanced audio workflow within Signal Labeler

The **Signal Labeler** app has been enhanced with new audio functionality that enables data importing from audio files and folders.

acousticRoughness: Measure perceived roughness of an acoustic signal

Apply the `acousticRoughness` function for perceptual sound quality analysis or noise vibration harshness evaluation.

Using algorithms based on the Zwicker loudness method (ISO 532-1), the `acousticRoughness` function quantifies perceived amplitude modulations (between approximately 10 Hz and 400 Hz) due to amplitude or frequency modulations in the audio signal.

The `acousticRoughness` function returns roughness in asper units and specific roughness in asper/Bark.

Octave Filter Bank Block: Octave and fractional-octave filter bank

The Octave Filter Bank Simulink® block simplifies the design and visualization of ANSI S1.11-2204 compliant filter banks. The Octave Filter Bank block maintains the properties of the `octaveFilterBank` object, with additional enhancements enabling individual frequency band access, as well as specifying or inheriting sample rate.

Sidechain input capability for dynamic range objects and blocks

The Sidechain input lets you use one audio input signal to provide dynamic range compression or expansion of a second, separate audio signal. Use the Sidechain input for channel-linked stereo audio processing, ducking, and de-essing.

The Sidechain input applies to four objects and their associated blocks:

- The `compressor` object and Compressor block
- The `expander` object and Expander block
- The `limiter` object and Limiter block
- The `noiseGate` object and Noise Gate block

GPU code acceleration for audioFeatureExtractor object

The `audioFeatureExtractor` object now supports `gpuArray` objects.

You must have Parallel Computing Toolbox™ to use `gpuArray` objects with supported functions. See Run MATLAB Functions on a GPU for more details and GPU Support by Release to see which GPUs are supported.

New rectification option for cepstralCoefficients

The `cepstralCoefficients` function now allows users to implement custom rectification outside of the function.

- `'log'` — Performs logarithmic 'Rectification'.
- `'cubic-root'` — Performs cubic-root 'Rectification'.
- `'none'` — No 'Rectification' is carried out. This option enables you to implement your own rectification algorithms outside of the function.

Extended JUCE project support

`generateAudioPlugin` function can now generate C/C++ source code and JUCE project files suitable for use with JUCE 5.3.2 to 6.0.1. This functionality requires a MATLAB® Coder™ license.

Set nondefault time stretching and pitch shifting parameters using audioDataAugmenter

`audioDataAugmenter` has been enhanced with two new object functions allowing default parameter value modification of time stretching and pitch shifting augmentation algorithms.

Additional examples for deep learning and psychoacoustics

Design and Use Deep Learning Systems

- “Speaker Recognition Using x-vectors”
- “Speaker Diarization Using x-vectors”
- “Train Spoken Digit Recognition Network Using Out-of-Memory Audio Data”
- “Train Spoken Digit Recognition Network Using Out-of-Memory Features”
- “Speaker Identification Using Custom SincNet Layer and Deep Learning”
- “Dereverberate Speech Using Deep Learning Networks”
- “Speech Command Recognition in Simulink”

Deploy Deep Learning Systems

- “Keyword Spotting in Noise Code Generation with Intel MKL-DNN”
- “Keyword Spotting in Noise Code Generation on Raspberry Pi”

Psychoacoustics

- “Effect of Soundproofing on Perceived Noise Levels”

R2020b

YAMNet Pretrained Network: Classify sounds with pretrained YAMNet neural network

Use the `classifySound` function to preprocess, perform sound classification using YAMNet, and then postprocess the network outputs. You can specify smoothing and thresholding parameters for sound classification, and analyze the output by viewing the most likely sounds for a whole signal, or a table of detected sounds and the corresponding decision confidence. YAMNet is a pretrained neural network that predicts 521 audio event classes.

Use the `yamnet` function to interact with the network object directly. The network is returned as a `SeriesNetwork` object. You can use this pretrained model for classification and transfer learning.

Use the `yamnetGraph` function to explore the audio event class ontology.

The YAMNet functionality requires Deep Learning Toolbox.

VGGish Pretrained Network: Extract audio embeddings with pretrained VGGish neural network

Use the `vggishFeatures` function to extract semantically meaningful 128-dimensional feature vectors (embeddings). You can use the feature vectors as input for a classification model.

Use the `vggish` function to interact with the network object directly. The network is returned as a `SeriesNetwork` object. You can use this pretrained model for feature extraction and transfer learning.

The VGGish functionality requires Deep Learning Toolbox.

Extract cepstral coefficients from spectrograms and auditory spectrograms

Use the `cepstralCoefficients` function to extract cepstral coefficients from spectrograms and auditory spectrograms. The `cepstralCoefficients` function supports streaming and nonstreaming audio signals.

You can now use modular functions such as `stft`, `designAuditoryFilterBank`, `cepstralCoefficients`, and `audioDelta` in combination for efficient extraction of audio features such as mel frequency cepstral coefficients (MFCC), gammatone frequency cepstral coefficients (GTCC), and Bark frequency cepstral coefficients (BFCC). You can also use these modular functions to explore, extend, or modify default implementations.

Compute delta of audio features

Use the `audioDelta` function to compute the delta of any audio feature. The `audioDelta` function supports streaming and nonstreaming audio signals.

You can now use modular functions such as `stft`, `designAuditoryFilterBank`, `cepstralCoefficients`, and `audioDelta` in combination for efficient extraction of audio features such as mel frequency cepstral coefficients (MFCC), gammatone frequency cepstral coefficients (GTCC), and Bark frequency cepstral coefficients (BFCC). You can also use these modular functions to explore, extend, or modify default implementations.

Enhanced control of designAuditoryFilterBank

The `designAuditoryFilterBank` function now includes parameters to enable the following functionality:

Parameter	New Functionality
`'OneSided'`	Specify whether the filter bank returned is one-sided or two-sided. Depending on your processing pipeline, using a two-sided filter bank may increase speed.
`'FilterBankDesignDomain'`	Specify whether the filter shapes are designed in the linear (Hz) or warped (mel or Bark) domain. This parameter is only applicable when the frequency scale is mel or Bark.

Enhanced control of audioFeatureExtractor and Extract Audio Features

The `audioFeatureExtractor` object and the **Extract Audio Features** task now include parameters to enable the following functionality:

`audioFeatureExtractor` Parameter	Extract Audio Features UI label	New Functionality
`WindowNormalization`	Window normalization	Specify whether to normalize a spectrum by the time-domain window energy. This functionality applies when using or extracting a linear, mel, Bark, or ERB spectrum.
`FilterBankDesignDomain`	Filter bank domain	Specify whether the filter shapes are designed in the linear (Hz) or warped (mel or Bark) domain. This functionality applies when using or extracting a mel or Bark spectrum.

Enhanced control of melSpectrogram

The `melSpectrogram` function now includes parameters to enable the following functionality:

Parameter	New Functionality
`'Window'`	Specify the desired time-domain window to apply prior to the discrete Fourier transform (DFT).
`'WindowNormalization'`	Specify whether or not to normalize the spectrum by the window energy.

Parameter	New Functionality
`'FilterBankNormalization'`	Specify the type of normalization applied to each filter in the filter bank. You can choose to ignore normalization, or normalize by the individual filter bandwidth or area.

Enhanced control of time-domain windowing for mfcc and gtcc

The `mfcc` and `gtcc` functions include a new parameter, `Window`, that enables you to specify the window applied in the time domain.

Extract spectral flux from streaming signals

To extract spectral flux from streaming signals, you can now pass state in and out of the `spectralFlux` function.

Generate MATLAB function from audioFeatureExtractor

Use `generateMATLABFunction` to create a MATLAB function from the `audioFeatureExtractor` object. The generated MATLAB function performs equivalent audio feature extraction as the object, and is compatible with C/C++ code generation.

You can also generate a MATLAB function that is optimized for frame-based, streaming audio signals. When you generate a MATLAB function for stream processing, required states are maintained by the function.

GPU code acceleration for new and existing features

The following new features support `gpuArray` objects:

- `audioDelta`
- `cepstralCoefficients`

The following existing features now support `gpuArray` objects:

- `stretchAudio`
- `shiftPitch`
- `spectralCentroid`
- `spectralCrest`
- `spectralDecrease`
- `spectralEntropy`
- `spectralFlatness`
- `spectralFlux`
- `spectralKurtosis`
- `spectralRolloffPoint`
- `spectralSkewness`

- `spectralSlope`
- `spectralSpread`

You must have Parallel Computing Toolbox to use `gpuArray` objects with supported functions. See Run MATLAB Functions on a GPU for more details and GPU Support by Release to see which GPUs are supported.

GPU code generation support for the melSpectrogram function

The `melSpectrogram` function now supports code generation for graphical processing units (GPUs). If you have GPU Coder™, you can generate optimized CUDA® code from the `melSpectrogram` function for machine learning and deep learning systems.

Measure perceived acoustic fluctuation strength

Use `acousticFluctuation` to measure perceived fluctuation in accordance with the Zwicker model of fluctuation and ISO 532-1:2017(E) (the Zwicker loudness method). The `acousticFluctuation` function returns fluctuation in vacil and specific fluctuation in vacil/Bark.

Enhanced control over sound pressure level (SPL) metering

You can now use the `FrequencyRange` property of the `splMeter` object to specify the frequency range to analyze.

Improved low-frequency and high-frequency octave filtering using octaveFilter

The `octaveFilter` object has been enhanced so that you can specify a larger range of center frequencies. This enhancement includes more accurate filtering at low and high frequencies.

Improved low-frequency octave filtering using octaveFilterBank

The `octaveFilterBank` object has been enhanced to provide more accurate filtering at low frequencies.

Enhanced Audio Test Bench workflow

The **Audio Test Bench** app has been enhanced to maintain settings applied to scopes between sessions.

Report audio plugin latency to host

Use the `setLatencyInSamples` method in your `audioPlugin` class definition to report latency to your host application. Typically, digital audio workstations (DAWs) use this information to compensate for algorithmic latency and align tracks.

Refresh audio device list from audio I/O blocks

The Audio Device Reader block and the Audio Device Writer block now include the ability to refresh a list of available audio devices. This functionality enables you to add and remove audio devices during a single MATLAB session.

Additional examples for deep learning

Deploy Deep Learning Systems

- Speech Command Recognition Code Generation on Raspberry Pi
- Speech Command Recognition Code Generation with Intel MKL-DNN

Functionality being removed or changed

Specify window for mfcc, gtcc, and melSpectrogram functions

Behavior change in future release

The `WindowLength` parameter will be removed from `mfcc`, `gtcc`, and `melSpectrogram` in a future release. Use the `Window` parameter instead.

For example, in releases prior to R2020b, you could only specify the length of the time-domain window. The window was always designed as a periodic Hamming window. You can replace instances of the following code:

```
a = mfcc(audioIn,fs,'WindowLength',1024);
b = gtcc(audioIn,fs,'WindowLength',1024);
c = melSpectrogram(audioIn,fs,'WindowLength',1024);
```

With this code:

```
a = mfcc(audioIn,fs,'Window',hamming(1024,'periodic'));
b = gtcc(audioIn,fs,'Window',hamming(1024,'periodic'));
c = melSpectrogram(audioIn,fs,'Window',hamming(1024,'periodic'));
```

Delta computation for mfcc, gtcc, and audioFeatureExtractor

Behavior change

The delta and delta-delta calculations are now computed using the `audioDelta` function, which has a different startup behavior than the previous algorithm. The default `DeltaWindowLength` has changed from 2 to 9. A delta window length of 2 is no longer supported.

Window normalization parameter renamed for audioFeatureExtractor

Behavior change in future release

The `FilterBankNormalization` parameter replaces the `Normalization` parameter when using or extracting the `melSpectrum`, `barkSpectrum`, or `erbSpectrum`. The `Normalization` parameter will be removed in a future release.

For example, replace instances of the following code:

```
afe = audioFeatureExtractor('melSpectrum',true,'barkSpectrum',true,'erbSpectrum',true);
setExtractorParams(afe,'melSpectrum','Normalization','none')
setExtractorParams(afe,'barkSpectrum','Normalization','none')
setExtractorParams(afe,'erbSpectrum','Normalization','none')
```

With this code:

```
afe = audioFeatureExtractor('melSpectrum',true,'barkSpectrum',true,'erbSpectrum',true);
setExtractorParams(afe,'melSpectrum','FilterBankNormalization','none')
setExtractorParams(afe,'barkSpectrum','FilterBankNormalization','none')
setExtractorParams(afe,'erbSpectrum','FilterBankNormalization','none')
```

SOS returned instead of FOS from octaveFilterBank
Behavior change

The `coeffs` function of `octaveFilterBank` now returns the filter in second-order sections (SOS) instead of fourth-order sections (FOS). This new format reflects an updated internal representation, which has been enhanced to remain stable at very low frequencies.

cepstralFeatureExtractor will be removed
Still runs

`cepstralFeatureExtractor` will be removed. Use the `mfcc` and `gtcc` functions to compute the same features for batch signals. For time-critical processing, use a combination of `designAuditoryFilterBank`, `cepstralCoefficients`, and `audioDelta` to compute the same features. If you are extracting multiple audio features, use the `audioFeatureExtractor`.

designAuditoryFilterBank scaling changed for ERB filter banks
Behavior change

The half-sided ERB filter bank returned from `designAuditoryFilterBank` is now scaled by 2. This change provides consistent results when applying one-sided or two-sided filtering, without requiring multiplications in the processing loop.

R2020a

Version: 2.2

New Features

Measure perceived loudness according to ISO 532-1 or ISO 532-2

Use `acousticLoudness` to measure perceived loudness in accordance with ISO 532-1:2017(E) (the Zwicker method) and ISO 532-2:2017(E) (the Moore-Glasberg method). Both methods can return loudness in sone and specific loudness. When using the Zwicker method, you can also measure the percentile loudness of time-varying signals, as described in ISO 532-1:2017(E).

Measure perceived sharpness according to DIN 45692

Use `acousticSharpness` to measure perceived sharpness of an acoustic signal, as specified by the ISO 532-1:2017(E) and DIN 45692 standards.

Determine calibration factor for microphone

Use `calibrateMicrophone` to determine a calibration factor for your audio input system.

Convert between acoustic loudness units phon and sone

Audio Toolbox now includes conversion functions for loudness units phon and sone.

- `phon2sone` –– Convert from phon to sone
- `sone2phon` –– Convert from sone to phon

Detect boundaries of speech in audio

Use `detectSpeech` to determine the start and end indices of regions of speech in audio.

Speech detection is a common preprocessing step for machine learning and deep learning workflows. See Classify Gender Using LSTM Networks and Keyword Spotting in Noise Using MFCC and LSTM Networks for examples.

Streamline audio feature extraction in the Live Editor

Use the **Extract Audio Features** task to configure an optimized feature extraction pipeline by selecting features and parameters graphically. You can reuse the output from the **Extract Audio Features** task to apply feature extraction to entire data sets.

GPU code generation support for the mfcc function

The `mfcc` function now supports code generation for graphical processing units (GPUs). If you have GPU Coder, you can generate optimized CUDA code from the `mfcc` function for machine learning and deep learning systems.

Audio Datastore: Write data from audio datastore using writeall

You can now write data from an `audioDatastore` object to files on disk using the `writeall` function.

GPU code acceleration for mfcc and melSpectrogram functions

The `mfcc` and `melSpectrogram` functions now support `gpuArray` objects. You must have Parallel Computing Toolbox to use `gpuArray` objects with supported functions. See Run MATLAB Functions on a GPU for more details and GPU Support by Release to see which GPUs are supported.

Text-to-speech conversion using third-party speech API

To perform text-to-speech conversion in MATLAB, use the `text2speech` function available on File Exchange. The function enables you to interface third-party text-to-speech APIs, including:

- Google® Speech API
- IBM® Watson Speech API
- Microsoft® Azure Speech API

The File Exchange submission includes a tutorial to help get you started.

Cubic root rectification for MFCC and GTCC calculations

You can now choose between cubic-root and log rectification when extracting mel frequency cepstral coefficients and gammatone cepstral coefficients. This enhancement applies to the `mfcc` and `gtcc` functions, the `cepstralFeatureExtractor` and `audioFeatureExtractor` objects, and the Cepstral Feature Extractor block.

Enhanced look and feel for Audio Test Bench

The **Audio Test Bench** app has been enhanced with a modern look and feel. The app now supports custom UI layouts and parameter styles defined in your plugin interface.

Enhanced visualization for loaded plugins

The `parameterTuner` function now renders parameter-specific widgets for plugins loaded using `loadAudioPlugin`.

Generate standalone executable from audio plugin

You can now use the -exe option of the generateAudioPlugin function to generate a binary standalone executable from an audio plugin. When you evaluate the generated code, the UI you defined in your audio plugin opens. You can tune parameters and control the input and output of the plugin using the UI.

Generate sine, square, and sawtooth waveforms in Simulink

Use the Audio Oscillator block in Simulink to generate tunable sine, square, and sawtooth waveforms. You can tune parameters of the waveforms, such as frequency, amplitude, and DC offset, using input ports or a UI.

Generate periodic signal from single-cycle waveforms in Simulink

Use the Wavetable Synthesizer block in Simulink to generate a periodic signal from a single-cycle waveform you define. You can tune parameters of the signal, such as frequency, amplitude, and DC offset, using input ports of a UI.

Additional input ports for Audio Toolbox blocks

The table describes the new optional input ports for tuning your block parameters.

Block	New Optional Input Ports
Reverberator	Pre-delay (s), Highcut frequency (Hz), Diffusion, Decay factor, High frequency damping, Wet/dry mix
Voice Activity Detector	silence-to-speech probability, speech-to-silence probability
Crossover Filter	Crossover frequency (Hz), Crossover order

Additional examples for deep learning and machine learning

- Train Generative Adversarial Network (GAN) for Sound Synthesis
- Speaker Verification Using Gaussian Mixture Model
- Speaker Verification Using i-Vectors

R2019b

Version: 2.1

New Features

AU Plugin Generation: Generate AU plugins for macOS

You can now generate an Audio Unit (AU) v2 audio plugin binary using `generateAudioPlugin`.

Custom Plugin UI: Generate VST and AU plugins with custom UIs

Define custom user interfaces (UIs) using the updated `audioPluginInterface` and `audioPluginParameter` functions, and the new `audioPluginGridLayout` function. The custom UI is visible once the plugin is generated or when using `parameterTuner` in MATLAB. Customization abilities include:

- Define parameter controls that render as knobs, sliders, rocker switches, toggle switches, check boxes, or drop-downs.
- Position controls and labels on a grid layout.
- Define custom background color, background image, or both.
- Define custom images for controls (generated plugins only).

See Design User Interface for Audio Plugin for more information.

Enhanced Parameter Tuner UI

`parameterTuner` now renders parameters as knobs, sliders, rocker switches, toggle switches, check boxes, or drop-downs when used with objects that inherit from `audioPlugin`. You can also define a custom background color, background image, or both. Custom images for controls are not supported by `parameterTuner`.

Audio Data Augmentation: Enlarge your dataset using audio-specific augmentation

`audioDataAugmenter` enables you to enlarge your audio dataset using audio-specific augmentation techniques like pitch shifting, time-scale modification, time shifting, noise addition, and volume control. You can create a cascaded augmentation pipeline to apply multiple algorithms probabilistically, or a parallel augmentation pipeline to apply algorithms deterministically. You can also add custom augmentation algorithms to `audioDataAugmenter`.

See Keyword Spotting in Noise Using MFCC and LSTM Networks for an example.

Audio Feature Extraction: Streamline audio feature extraction

Use `audioFeatureExtractor` to extract multiple audio features using an efficient processing pipeline. `audioFeatureExtractor` encapsulates the extraction pipeline so that your code is cleaner and more modular.

See Sequential Feature Selection for Speech Emotion Recognition and Classify Gender Using LSTM Networks for examples.

Pitch shifting: Increase or decrease the pitch of audio signals

Use `shiftPitch` to increase or decrease the pitch of an audio signal by a given number of semitones. To achieve better fidelity with the original audio, you can optionally apply phase locking and formant preservation.

Time stretching: Stretch the time scale of audio signals

Use `stretchAudio` to apply time-scale modification (TSM) to an audio signal. You can speed up or slow down audio while preserving the original pitch.

For streaming applications, use `audioTimeScaler` to apply TSM. The `audioTimeScaler` enables you to tune the speedup factor while streaming.

Auditory Filter Banks: Design common frequency-domain auditory filter banks

Use `designAuditoryFilterBank` to design a mel, Bark, or ERB filter bank. You can use the filter bank to apply computationally efficient frequency-domain filtering.

Enhanced MFCC extraction

The `mfcc` function now accepts frequency-domain input so that you can reuse your DFT computation.

The `mfcc` function now enables you to specify the bandedges of the filter bank. You can use this ability to fine-tune the MFCC feature extraction or mimic other implementations.

Enhanced GTCC extraction

The `gtcc` function now accepts frequency-domain input so that you can reuse your DFT computation.

Audio Labeler App: Automatically label regions of speech

The **Audio Labeler** app now provides automatic labeling of detected regions of speech.

Audio Labeler App: Speech-to-text transcription using third-party speech API

The **Audio Labeler** app now supports the `speech2text` function available on File Exchange. Through the Audio Labeler app, you can interface with third-party speech-to-text APIs, including:

- Google Speech API
- IBM Watson Speech API
- Microsoft Azure Speech API

The `speech2text` entry on File Exchange includes a tutorial includes a tutorial to help get you started.

Pink Noise: Generate noise signals common to audio applications

Use `pinknoise` to generate a single channel or multiple independent channels of pink noise that is bounded between –1 and 1. The power spectral density of pink noise is inversely proportional to frequency and falls off at 10 dB/decade (3 dB/octave). Pink noise is commonly used to test and equalize loudspeakers and to mimic background noise encountered in real-world situations.

Tune reverberator parameters graphically

You can now tune parameters of the `reverberator` object graphically using `parameterTuner`.

Specify coefficient orientation output from designParamEQ, designShelvingEQ, and designVarSlopeFilter

`designParamEQ`, `designShelvingEQ`, and `designVarSlopeFilter` now enable you to specify the orientation of the returned filter coefficients. Specify the orientation as `'row'` for interoperability with **FVTool**, `dsp.DynamicFilterVisualizer`, and `dsp.FourthOrderSectionFilter`.

Visualize and analyze the filters designed by weightingFilter and octaveFilter

The `weightingFilter` object and the `octaveFilter` object now include additional filter analysis tools.

Additional examples for deep learning, active noise control, positional audio, and time-frequency masking

- Keyword Spotting in Noise Using MFCC and LSTM Networks
- Sequential Feature Selection for Speech Emotion Recognition
- Active Noise Control: From Modeling to Real-Time Prototyping
- Binaural Audio Rendering Using Head Tracking
- Time-Frequency Masking for Harmonic-Percussive Source Separation

R2019a

Version: 2.0

New Features

Compatibility Considerations

Modified Discrete Cosine Transform (MDCT)

Audio Toolbox enables you to transform to and from a compact frequency domain representation with perfect reconstruction:

- `mdct` –– Transform a signal into a compact frequency-domain representation using the modified discrete cosine transform (MDCT)
- `imdct` –– Transform a signal from a compact frequency-domain representation to the time domain using the inverse MDCT.
- `kbdwin` –– Create a Kaiser-Bessel derived window. This window enables perfect reconstruction when used with `mdct` and `imdct`.

Gammatone Filter Bank: Mimic the human auditory system

Use `gammatoneFilterBank` to decompose a signal by passing it through a bank of gammatone filters equally spaced on the equivalent rectangular bandwidth (ERB) scale. Gammatone filter banks are designed to model the human auditory system.

Mel-Spaced Spectrogram: Transform signals into perceptually-spaced compact time-frequency representations

Use `melSpectrogram` to compute the mel spectrogram of an audio signal. Mel spectrograms are popular features in deep-learning applications.

See Speech Command Recognition Using Deep Learning (Deep Learning Toolbox) and Acoustic Scene Recognition Using Late Fusion for examples.

Feature Extraction: Gammatone cepstral coefficients (GTCC)

Use `gtcc` to extract gammatone cepstral coefficients from audio signals. You can specify the window length and overlap length used for analysis, and optionally return the delta and delta-delta features calculated with look-ahead. Gammatone cepstral coefficients are a biologically inspired modification to mel frequency cepstral coefficients (`mfcc`), which have been shown to be robust to noise when used in machine-learning applications.

See Classify Gender Using Long Short-Term Memory Networks for an example.

Feature Extraction: Characterize level of harmonicity in audio signals

Use `harmonicRatio` to describe how much of the total energy of a signal is harmonic.

Feature Extraction: Characterize spectral shape of audio signals

Audio Toolbox now includes a suite of features that describe spectral shape, or timbre:

- `spectralCentroid`
- `spectralCrest`
- `spectralDecrease`

- `spectralEntropy`
- `spectralFlatness`
- `spectralFlux`
- `spectralKurtosis`
- `spectralRolloffPoint`
- `spectralSkewness`
- `spectralSlope`
- `spectralSpread`

See Spectral Descriptors for an overview of spectral descriptors and common applications.

Feature Extraction: Enhancements to cepstral feature extractors

`cepstralFeatureExtractor` and the Cepstral Feature Extractor block can now return gammatone cepstral coefficients (GTCC). Use `cepstralFeatureExtractor` in MATLAB and the Cepstral Feature Extractor block in Simulink when computing cepstral features for streaming audio.

Enhancements to Audio Datastore: Combine datastores and define custom read functions

`audioDatastore` has been enhanced to include the following functions:

- `transform` –– Define a custom read function on a datastore
- `combine` –– Combine data from multiple audio datastores into a single datastore

Convert between Hz, Bark, ERB, and mel domains

Audio Toolbox now includes conversion functions between Hz and popular auditory scales: Bark, equivalent rectangular bandwidth (ERB), and mel.

- `erb2hz` –– Convert from ERB scale to Hz
- `hz2erb` –– Convert from Hz to ERB scale
- `bark2hz` –– Convert from Bark scale to Hz
- `hz2bark` –– Convert from Hz to Bark scale
- `mel2hz` –– Convert from mel scale to Hz
- `hz2mel` –– Convert from Hz to mel scale

Generate JUCE projects from your audio plugins

`generateAudioPlugin` can now generate C/C++ source code and JUCE project files suitable for use with JUCE 5.3.2. This functionality requires a MATLAB Coder license.

Octave Filter Bank: Decompose signal into octave or fractional-octave subbands

Use `octaveFilterBank` to decompose signals into octave or fractional-octave subbands.

Graphically tune audio plugins and Audio Toolbox objects while streaming

Use `parameterTuner` to graphically tune parameters of audio plugins while streaming. `parameterTuner` is compatible with classes that inherit from `audioPlugin` and define tunable parameters.

You can also tune parameters of Audio Toolbox objects, including:

- `compressor`
- `expander`
- `limiter`
- `noiseGate`
- `octaveFilter`
- `crossoverFilter`
- `multibandParametricEQ`
- `graphicEQ`
- `audioOscillator`
- `wavetableSynthesizer`

Enhanced Parametric Equalization in Simulink

The Parametric EQ Filter block has been renamed as Parametric EQ and enhanced to use the `designParamEQ` algorithm. Instances of the Parametric EQ Filter block in existing models will not be automatically updated.

Improved Swept Sine Generation and Impulse Response Estimation

`sweeptone` and `impzest` have been improved to calculate more accurate impulse response estimations. The output of the `sweeptone` function has changed. `impzest` can be used with recordings using `sweeptone` from early releases as long as the corresponding excitation is specified to `impzest`.

New examples for deep learning, active noise control, pitch tracking, and MIDI

Examples for machine learning and deep learning:

- Cocktail Party Source Separation Using Deep Learning Networks
- Voice Activity Detection in Noise Using Deep Learning
- Acoustic Scene Recognition Using Late Fusion

- Spoken Digit Recognition with Wavelet Scattering and Deep Learning

Examples for active noise control:

- Active Noise Control with Simulink Real-Time

Example for pitch tracking:

- Pitch Tracking Using Multiple Pitch Estimations and HMM

Example for MIDI:

- Convert MIDI Files into MIDI Messages

R2018b

Version: 1.5

New Features

Audio Labeler App: Interactively define and visualize ground-truth labels for audio datasets

Use the **Audio Labeler** app for interactive audio labeling. The Audio Labeler app enables you to:

- Visualize the time-domain waveform during playback.
- Assign labels at the file level and region level.
- Create label definitions for consistent and fast labeling.
- Record audio.

Audio Datastore: Handle large collections of audio recordings for batch processing or machine and deep learning applications

Use `audioDatastore` to handle large collections of audio signals and associated labels that are too large to fit in memory. With audio datastores, you can:

- Point to a collection of audio files in a specified location.
- Associate labels with audio files.
- Split datastores according to label definitions and specified proportions.
- Randomize the order of audio files.
- Read files consecutively while monitoring progress.
- Process audio files in parallel when using a machine with multiple cores (requires Parallel Computing Toolbox).

Octave Level Metering: Measure sound pressure level for octave and fractional-octave bands of audio signals

The `splMeter` System object™ now enables you to measure the sound pressure level (SPL) of octave and fractional-octave bands.

HRTF Interpolation: Compute Head-Related Transfer Functions (HRTF) for arbitrary positions from space-discrete datasets

Use `interpolateHRTF` to interpolate between HRTFs that were measured at known positions.

Impulse Response Measurements: Estimate impulse responses of acoustical systems using MATLAB code

Use `impzest` to estimate the impulse response of an audio system given a known excitation signal and a recorded signal. The `impzest` function supports the maximum length sequence (MLS) technique and the exponential sine sweep (ESS) technique for impulse response estimation. Use `mls` and `sweeptone` to generate the excitation signals.

Audio Test Bench enhancements

The **Audio Test Bench** is a graphical debugging and testing suite for audio processing modules.

With the Audio Test Bench, you now can:

- Open custom visualizations for audio plugins.
- Use the visualization and tuning capabilities of the Audio Test Bench without writing audio to a device or file.

Additional examples for machine learning, deep learning, and spatial audio

Examples for machine learning and deep learning:

- Classify Gender Using Long Short-Term Memory Networks –– Use MFCC, pitch, harmonicity, and spectral centroid with a multilayer LSTM network to classify speaker gender.
- Denoise Speech Using Deep Learning Networks –– Use a spectrogram transformation and a deep CNN network to solve an audio regression problem.
- Music Genre Classification Using Wavelet Time Scattering –– Use wavelet time scattering and the audio datastore to classify music by genre.

Examples for spatial audio:

- Ambisonic Binaural Decoding –– Decode ambisonic audio into binaural audio using virtual loudspeakers.
- Ambisonic Plugin Generation –– Create ambisonic plugins using higher order ambisonic (HOA) demo functions.

R2018a

Version: 1.4

New Features

Impulse Response Measurer App: Interactively measure impulse and frequency responses of acoustic systems

The **Impulse Response Measurer** app enables you to acquire, analyze, and export impulse response and frequency response measurements through a user interface.

MIDI Message Interface: Send and receive MIDI messages of any type in MATLAB

You can now send and receive MIDI messages using the following features:

- `mididevice` –– Interface to a MIDI device in MATLAB. `mididevice` acts as a conduit between the MATLAB environment and your real-world MIDI device.
- `midimsg` –– Create a MIDI message in MATLAB
- `midisend` –– Send a MIDI message to an external MIDI device
- `midireceive` –– Receive a MIDI message from an external MIDI device

See MIDI Device Interface for a walkthrough of sending and receiving MIDI messages in MATLAB.

Voice Activity Detection: Automate the detection of speech content in audio signals

The `voiceActivityDetector` System object returns a confidence metric indicating the presence of speech in streaming audio signals. The input to the object can be time-domain or frequency-domain signals.

In the Simulink environment, use the Voice Activity Detector block.

Feature Extraction: Compute features of audio signals, such as pitch and MFCC

Detect the fundamental frequency of audio signals using the `pitch` function. You can choose between pitch detection algorithms, including the pitch estimation filter, normalized correlation function, cepstrum, log-harmonic summation, and summation of residual harmonics.

The Audio Toolbox enables batch and streaming approaches to cepstral feature extraction:

- `cepstralFeatureExtractor` –– Use the `cepstralFeatureExtractor` System object to process frame-based audio signals. You can specify your input in the time or frequency domain. This feature enables you to fine-tune the extracted features using the `BandEdges`, `FilterBankNormalization`, and `FilterBankDesignDomain` properties.

  In the Simulink environment, use the Cepstral Feature Extractor block.
- `mfcc` –– Use the `mfcc` function to process whole audio signals. You can specify the window length and overlap length used for analysis, and optionally return the delta and delta-delta features calculated with look-ahead.

Sound Pressure Level (SPL) Metering: Measure the level of acoustic signals in decibels relative to a standard perceptual reference

Use the `splMeter` System object to compute fast or slow, A-weighted or C-weighted, equivalent continuous, peak, and maximum sound pressure level measurements. You can calibrate your `splMeter` for your hardware and environment using the `CalibrationFactor` and `TimeInterval` properties. You can calculate your `CalibrationFactor` according to standards using the `calibrate` method.

Improved Audio Test Bench: Persistent I/O Settings and Bypass Mode

The **Audio Test Bench** is a graphical debugging and testing suite for audio processing modules.

New abilities of the **Audio Test Bench** include:

- Persistent input and output settings across sessions.
- A/B test your algorithm by bypassing the object under test.
- Visualize, analyze, and play unprocessed audio by not specifying an object under test.

Multichannel Support for RaspberryPi and STM Discovery Hardware

New multi-channel support for Mic In, Line In, and Speaker Out blocks for hardware support packages. See the hardware package release notes for more details:

- Release Notes for Embedded Coder Support Package for STMicroelectronics Discovery Boards (Embedded Coder Support Package for STMicroelectronics Discovery Boards)
- Release Notes for Simulink Support Package for Raspberry Pi Hardware (Simulink Support Package for Raspberry Pi Hardware)

Additional examples for word recognition and dataset recording

New examples include:

- Deep Learning Speech Recognition

  Use an auditory-based spectrogram to train a speaker-independent small vocabulary isolated word recognition system. Use `audioexample.Datastore` to manage large datasets. After training, you can run a streaming version for real-time word recognition. This example requires the Neural Network Toolbox™.

- Record Audio Datasets

  Record and label audio datasets using a user interface.

Speech-to-Text Transcription Using 3rd-Party Speech API

To perform speech-to-text transcription in MATLAB, use the `speech2text` function available on File Exchange. The function enables you to interface third-party speech-to-text APIs, including:

- Google Speech API

- IBM Watson Speech API
- Microsoft Azure Speech API

The File Exchange submission includes a tutorial to help get you started.

R2017b

Version: 1.3

New Features

AU Plugin Hosting: Run and test Audio Units (AU) plugins in MATLAB on macOS

You can now load Audio Units (AU) plugins into MATLAB using the `loadAudioPlugin` function. You can interact with the hosted plugin graphically using the **Audio Test Bench**.

Graphic Equalization: Boost and cut standard octave or fractional octave frequency bands in MATLAB and Simulink

You can use the `graphicEQ` System object to perform equalization. The graphic equalizer uses the ANSI S1.11-2004 and ISO 266:1997(E) standards to determine and label the center frequencies of individual bandpass filters.

In the Simulink environment, use the Graphic EQ block.

Real-World Parameter Values for Hosted Plugins: Set and get values of hosted plugin parameters directly, using standard dot notation

Plugins loaded into the MATLAB environment using `loadAudioPlugin` are now populated with properties with real-world values. You can interact with the properties directly using standard dot notation.

See Host External Audio Plugins for a description of how normalized parameter values are heuristically interpreted as real-world values.

VST and AU plugins continue to support interaction through normalized parameter values.

MATLAB Code Generation from Audio Test Bench: Automatically generate MATLAB code for real-time audio streaming and processing

You can now generate MATLAB code from the **Audio Test Bench**. The generated MATLAB code is the script implementation of your **Audio Test Bench**. Settings in the **Audio Test Bench**, such as plugin parameter values and scopes opened through the test bench, are also ported to the MATLAB code.

Direct Access to ASIO Configuration Panel: Open configuration panel of ASIO drivers directly from MATLAB

You can now open an ASIO settings user interface directly from MATLAB using the `asiosettings` function.

Additional input ports for Audio Toolbox blocks

The table describes the new optional input ports for tuning your block parameters.

Block	Parameters Tuned by New Optional Input Ports
Compressor	**Threshold (dB)**, **Ratio**, **Knee width (dB)**, **Attack time (s)**, **Release time (s)**
Expander	**Threshold (dB)**, **Ratio**, **Knee width (dB)**, **Attack time (s)**, **Release time (s)**, **Hold time (s)**
Limiter	**Threshold (dB)**, **Knee width (dB)**, **Attack time (s)**, **Release time (s)**, **Hold time (s)**
Noise Gate	**Threshold (dB)**, **Attack time (s)**, **Release time (s)**, **Hold time (s)**
Octave Filter	**Center frequency (Hz)**

Additional examples for machine learning, spatial audio, device measurements, and deployment to android

New examples include:

- Speaker Identification Using Pitch and MFCC
- Acoustic Beamforming Using a Microphone Array
- Measure Frequency Response of an Audio Device
- Parametric Audio Equalizer for Android Devices

Enhancements to existing examples include:

- Measure Impulse Response of an Audio System
- Measure Audio Latency

R2017a

Version: 1.2

New Features

Enhanced VST Workflow in Audio Test Bench: Interactively tune hosted VST plugins and test MATLAB objects in VST mode

The **Audio Test Bench** is a graphical debugging and testing suite for audio processing modules.

New abilities of the **Audio Test Bench** include:

- Host external VST and VST3 plugins. You can now interact with external plugins using the graphical UI of the **Audio Test Bench**, as you would in a DAW.
- Run audio plugins created in MATLAB as VST plugins.

Synchronized Playback and Acquisition: Play back and acquire audio signals synchronously in MATLAB via a single audioPlayerRecorder object

The `audioPlayerRecorder` System object reads and writes from an audio device simultaneously, enabling real-time system measurements when using ASIO, Core Audio, or ALSA drivers. Combining the play and record into a single object enables easy configuration and consistent latency between input and output.

WASAPI Driver Support on Windows: Stream signals from and to audio devices equipped with WASAPI drivers

The `audioDeviceWriter` and `audioDeviceReader` System objects and the Audio Device Reader and Audio Device Writer blocks now support WASAPI drivers on Windows machines.

File browsing in Audio Test Bench

You can now browse for audio input files and the object under test directly from the **Audio Test Bench**.

Additional fractional bandwidth option for octave filtering

The `octaveFilter` System object and Octave Filter block now support 2/3 octave bandwidth. This octave bandwidth is popular in graphic equalizer designs.

configureMIDI support for hosted audio plugins

You can now use the `configureMIDI` function to quickly synchronize your hosted audio plugins with MIDI devices.

Tab completion for parameter names and options

You can now use tab completion to complete parameter names and options for all System objects in Audio Toolbox. Tab completions also work for the `validateAudioPlugin`, `loadAudioPlugin`, and `generateAudioPlugin` functions.

Additional audio plugin examples

The Audio Plugin Gallery contains audio plugin example code that can be used as building blocks in larger systems, as models for design patterns, or as benchmarks for comparison.

New plugins in the gallery include:

- `audiopluginexample.FastConvolver`
- `audiopluginexample.Phaser`
- `audiopluginexample.MultiNotchFilter`
- `audiopluginexample.SpeechPitchDetector`
- `audiopluginexample.BeatDetector`

Enhancements to existing plugins in the gallery include:

- The `audiopluginexample.SpectralSubtractor` example now includes an analysis and synthesis buffering object. The `audiopluginexample.private.AnalysisAndSynthesisBuffer` object enables easy input/output buffering for the audio plugin API so that you can concentrate on your algorithm.

R2016b

Version: 1.1

New Features

Audio Plugin Hosting: Run and test VST plugins directly in MATLAB

The `loadAudioPlugin` function enables you to host external VST and VST3 plugins in MATLAB. You can process audio using the algorithm of the hosted plugin. You can interact with the hosted plugin programmatically by getting and setting parameters.

Improved Audio Test Bench: Choose from a wider range of input signals, and generate VST plugins directly from the app

The Audio Test Bench is a graphical debugging and testing suite for audio processing modules.

New abilities of the **Audio Test Bench** include:

- Switch the object under test in a single instance of the test bench.
- New input choices: `wavetableSynthesizer`, `audioOscillator`, `dsp.Chirp`, and `dsp.ColoredNoise`.
- Validate and generate VST plugins directly from the test bench.
- Track overrun and underrun in frames, seconds, or samples.

Loudness Metering: Measure standard-compliant loudness parameters

Measure integrated loudness and loudness range of an audio signal using the `integratedLoudness` function.

Measure momentary loudness, short-term loudness, integrated loudness, loudness range, and true-peak of streaming audio using the `loudnessMeter` System object. You can also open an 'EBU-Mode' visualization for loudness metering.

Measure momentary loudness, short-term loudness, and true-peak in the Simulink environment using the Loudness Meter block.

Octave-Band Filters: Select octave and fractional-octave signal bands using standard-compliant digital filters

Perform octave-band and fractional octave-band filtering for arbitrary center frequency using the `octaveFilter` System object. With this object, you can tune center frequency and bandwidth while the simulation is running. To check your compliance to the ANSI S1.11-2004 standard, use the `isStandardCompliant` method. To visualize and validate your filter response, use the `visualize` method.

In the Simulink environment, use the Octave Filter block.

Audio Weighting Filters: Compensate signal magnitude for perceptual measurements using standard-compliant A-, C-, and K-weighted filters

Perform frequency-weighted filtering using the `weightingFilter` System object. With this object, you can design A-weighted and C-weighted filters based on the ANSI S1.42-2001 standard, or K-weighted filters based on the ITU-R BS.1770-4 standard. To check your compliance to the IEC

61672-1:2002 standard, use the `isStandardCompliant` method. To visualize and validate your filter response, use the `visualize` method.

In the Simulink environment, use the Weighting Filter block.

Plugin class creation and MIDI support for multiband parametric equalizer

New functionality for the `multibandParametricEQ` System object includes:

- Plugin class creation using `createAudioPluginClass`
- MIDI support using `configureMIDI`

`multibandParametricEQ` is now enabled for the **Audio Test Bench**.

Simpler way to call System objects

Instead of using the `step` method to perform the operation defined by a System object, you can call the object with arguments, as if it were a function. The `step` method will continue to work. This feature improves the readability of scripts and functions that use many different System objects.

For example, if you create a `weightingFilter` System object named `Cweight`, then you call the System object as a function with that name.

```
Cweight = weightingFilter('C-weighting');
Cweight(x)
```

The equivalent operation using the `step` method is:

```
Cweight = weightingFilter('C-weighting');
step(Cweight,x)
```

When the `step` method has the System object as its only argument, the function equivalent has no arguments. This function must be called with empty parentheses. For example, `step(sysobj)` and `sysobj()` perform equivalent operations.

R2016a

Version: 1.0

New Features

VST plugin generation for digital audio workstations

Audio Toolbox enables the design and generation of VST plugins.

For more information, see Export a MATLAB Plugin to a DAW.

Interfaces to ASIO, ALSA, CoreAudio, and Windows Direct Sound

Audio Toolbox enables real-time audio processing using low-latency audio drivers.

For more information, see Audio I/O: Buffering, Latency, and Throughput.

Interfaces to MIDI controls for real-time tuning of MATLAB and Simulink simulations

Audio Toolbox enables real-time tuning in MATLAB and Simulink using MIDI controls.

For more information, see `configureMIDI` and Musical Instrument Digital Interface (MIDI).

Audio processing algorithms, sources, and measurements for MATLAB and Simulink

Audio Toolbox provides algorithms and tools for the design, simulation, and desktop prototyping of audio processing systems.

For more information, see Audio Processing Algorithm Design.

Audio test bench to automatically generate an interactive audio simulation environment

Audio Toolbox provides an all-in-one graphical debugging and testing suite.

For more information, see Audio Test Bench and Use the Audio Test Bench.

Support for C code generation

You can use MATLAB Coder to generate efficient C and C++ code for most Audio Toolbox functions, classes, and System objects.

For a list of supported functions and objects, see Audio System Toolbox.

For a guide to developing code capable of C code generation, see MATLAB Programming for Code Generation.

Support for MATLAB Compiler

You can use MATLAB Compiler™ to share MATLAB programs as standalone applications.

For an example, see Deploy Audio Applications with MATLAB Compiler.