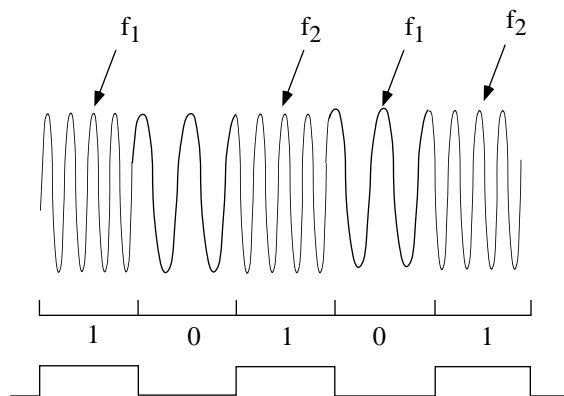




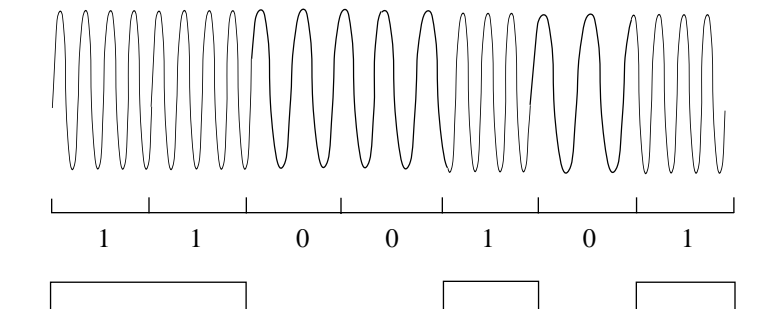
Experiment No. 2. Frequency Shift Keying (FSK) and Digital Communications

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One of the easiest and most useful modulation techniques for digital communication is Frequency Shift Keying (FSK). FSK means, quite simply, that one sends two frequencies over the air-waves; one frequency represents a digital “1” and one frequency represents a digital “0”. For example, if 101010... is desired, the signal becomes:



If 1100101 is desired, the signal becomes:



The data rate (or the frequency of the square wave or the time of a digital “1” or “0”) is called the FSK rate, and is in general several hundred times slower than f_1 or f_2 . For example, for $f_1 = 10$ MHz and $f_2 = 10.2$ MHz, the FSK rate could be 50 KHz.



Another excellent characteristic of FSK signals is that the transmit power is constant for both the digital "1" and "0" signals. For an antenna impedance of R_{ant} , the transmit power is:

$$P_t = \frac{V_{rms}^2}{R_{ant}} = \frac{Avg [\sqrt{2} V_{rms} \sin(\omega_1 t)]^2}{R_{ant}} =$$

$$\frac{Avg [\sqrt{2} V_{rms} \sin(\omega_2 t)]^2}{R_{ant}} = \frac{V_{rms}^2}{R_{ant}} \text{ (indep. of } \omega_1 \text{ and } \omega_2 \text{).}$$

There are several questions that may be on your mind about FSK and communications:

1. Why don't we send simply "1" and "0"? If we were in a coaxial cable, we would do just this. However, in transmit/receive systems, we must send electromagnetic waves (which are actually composed of sinusoids) using transmit antennas and capture them using receive antennas.
2. Who regulates the transmit/receive signals? Since all communication systems share the same medium, the "air-waves" are regulated very stringently by the FCC (Federal Communications Commission). This organization regulates the AM, FM, TV, police, hospital, citizen, satellite, wireless telephone, ... bands and sets the modulation technique/spectrum/bandwidth/power/harmonics/... in each band! In general, the law in the U.S. says that one can receive anything (even police and military communications!) but no one can transmit without getting a license from the FCC which specifies exactly the frequency, power, modulations, etc. of the transmitter. This is done so as to make sure that your transmitter does not interfere with other channels in the area. The license can be virtually free (for example, radio amateurs, etc. ...) or can cost several hundred million dollars for exclusive use (for example, wireless telephone operators in New York city).
3. Why don't we send f_1 for a digital "1" and nothing for a digital "0"? We could conserve on transmit power this way!
 - a. Remember that we are transmitting 10's or 100's of Watts in the time of the signal "f₁". This means that it takes a long time (msec!) to switch off a transmitter from "f₁" to "nothing" (digital "0"). Therefore, we cannot use a fast bit rate if "f₁" and "nothing" are sent!
 - b. When "nothing" is sent, it does not mean that one receives nothing! The receiver will tune and amplify the noise in the digital "0" area and may treat it as a digital "1". This will lead to a large bit-error rate (BER).
4. What is the optimum f_1 and f_2 vs. the data rate (FSK rate)? This is reserved for communication theory (EECS 455) which predicts that if we want to send a certain data stream with each bit taking a time T_b then the separation between f_1 and f_2 for optimum bandwidth and minimum error is:

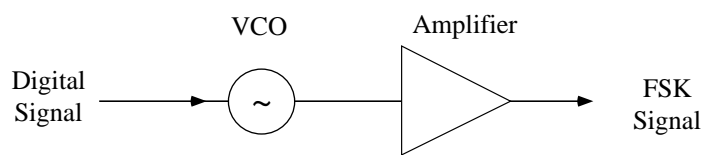
$$\Delta f = f_1 - f_2 = \frac{1.43}{T_b} \text{ where } T_b \text{ is the time allocated for a digital "1" or a digital "0".}$$



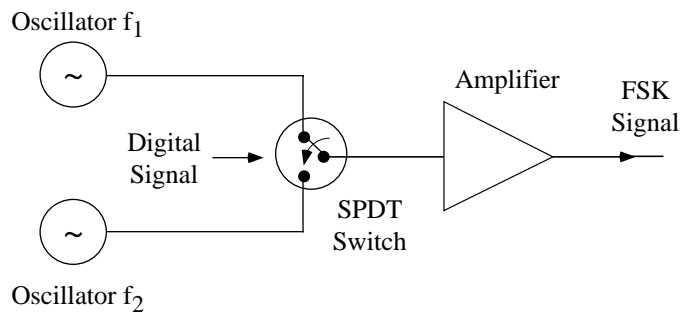
If we have a good detector, then the bit-error rate (BER) is about 10^{-6} for a signal-to-noise ratio of 15 dB at the receiver and 10^{-8} for a signal-to-noise ratio of 20 dB.

Generation of FSK Signals:

The best part of FSK modulation is the ease of signal generation. In the simplest technique, a voltage controlled oscillator (VCO) is used which oscillates at f_1 and f_2 depending on the control “digital” voltage. VCO are very easy to build and are composed of a transistor (or high-frequency op-amp) with a variable capacitor (actually a varactor diode) in the feedback circuit. The digital voltage controls the bias on the diode and hence the oscillation frequency. The VCO output is generally in the mW levels, so an amplifier is needed to boost the signal to the Watt levels.

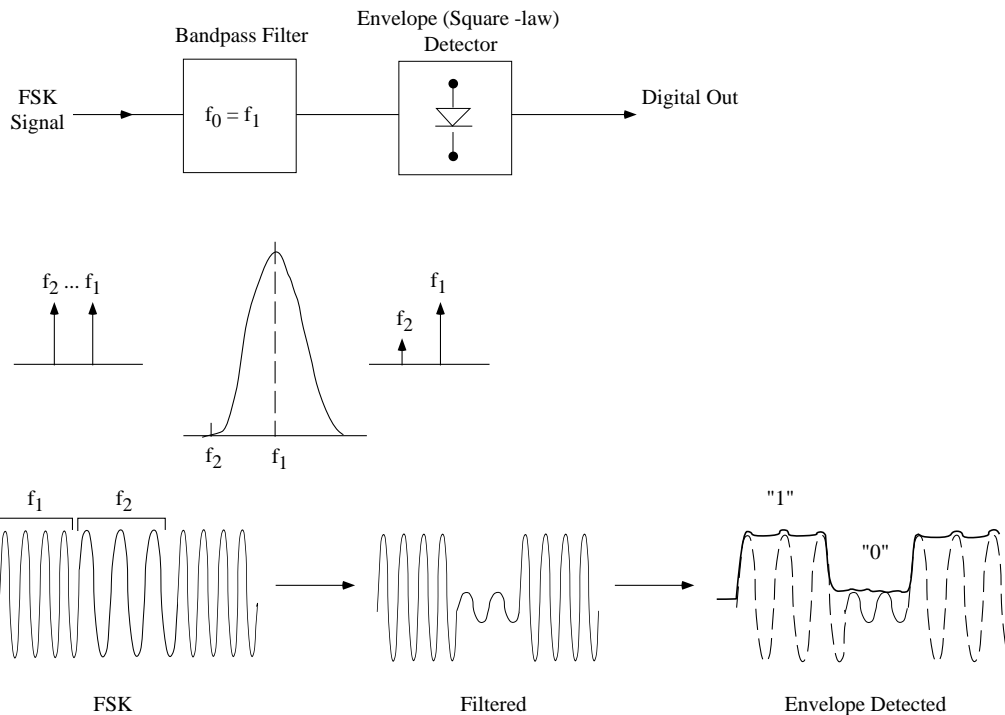


A low cost VCO takes a certain time to switch from f_1 to f_2 , typically around 1 μ sec, and therefore this technique is not suitable for very high data rate (MHz rates) signals. In this case, two oscillators are used, one at f_1 and one at f_2 , and the digital signal controls a single pole double throw (SPDT) switch which selects either f_1 or f_2 . It is common to have a switching time of nanoseconds and therefore this technique can handle 10's of MHz of data rates.



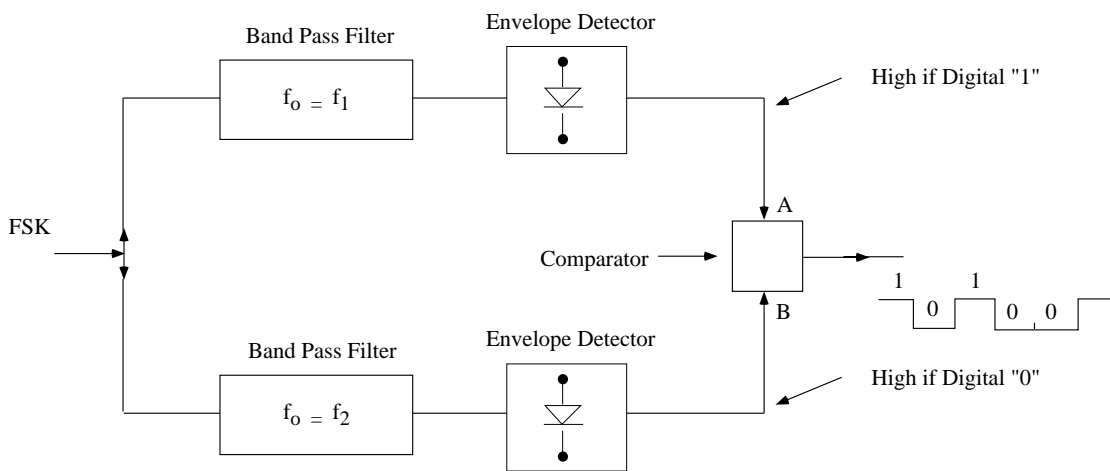
Detection of FSK Signals:

1. Single Filter: A single bandpass filter is used which is tuned to f_1 and attenuates f_2 by +30 dB. After the filter, the signal is passed through an envelope detector to smooth out the waveform (this is Experiment #4). The resulting signal is the digital waveform.



The main disadvantage of this technique is that noise peaks can affect the digital "0" area and it can be seen as a "1" resulting in a high BER (bit-error rate).

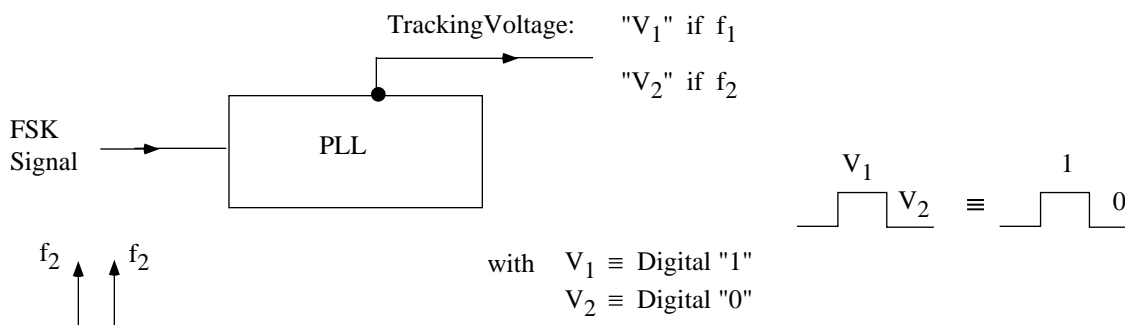
- 2. Matched Filters: Two bandpass filters are used. One is tuned to f_1 and one is tuned to f_2 . Each filter attenuates the other frequency by 30 dB or more. If the output of filter 1 (f_1) is high, then a digital "1" is being received. If the output of filter 2 (f_2) is high, then a digital "0" is being received. The comparator looks at both signals and outputs "1" if input A is high and "0" if input B is high. This can be easily done using op-amps or digital gates.



- 3. Phase Locked Loop (PLL): This is the modern and best way to detect FSK signals. A (PLL) is actually a voltage controlled oscillator (VCO) which tracks a signal in frequency and phase. The



PLL outputs a tracking DC signal (V_O) which is a direct indication of the input frequency. For example, V_O can be 1V for $f_2 = 10$ MHz and 1.5 V for $f_2 = 10.2$ MHz. The output voltage is then sent to an op-amp comparator which turns it into a digital "1" or "0". The FSK signal is therefore fed to a PLL and the tracking signal is monitored to get the digital information.



In a real communications system, some error correction is always applied. A bit error rate of 10^{-6} is probably ok for your personal e-mail, but totally unacceptable for a bank.

Band-Pass Filters

A general second-order bandpass filter has a transfer function given by:

$$H(s) = \frac{K(\omega_o/Q)s}{s^2 + (\omega_o/Q)s + \omega_o^2} \quad s = j\omega \quad (1)$$

with a gain of $|K|$, a "resonant" frequency of ω_o , and a quality factor Q . The frequency response is shown in Figure 1. The quality factor Q determines the 3-dB bandwidth ($\Delta\omega$) of the filter, and for $Q > 1$, $\Delta\omega = \omega_o/Q$. For $Q > 0.5$, ω_o is called the "resonant" frequency, while for $Q < 0.5$, ω_o is called the "mean" frequency, i.e., the mean frequency between two poles ω_1 and ω_2 ($\omega_o = \sqrt{\omega_1 \omega_2}$, see Fig.1). At $\omega = \omega_o$, $|H(\omega_o)| = K$, independent of Q , and $\angle H(\omega_o) = 0$. The roll-off at $\omega \ll \omega_o$ and $\omega \gg \omega_o$ is proportional to $1/\omega$ and is -20 dB/dec. It is easy to determine K , ω_o , and Q from the frequency domain measurements.

Determining K , ω_o , and Q from Time-Domain Measurements:

If a step function, $Au(t)$, is impressed on the bandpass filter, the output voltage is:

$$V_o(t) = \frac{2KA}{\sqrt{4Q^2 - 1}} e^{\frac{-\omega_o t}{2Q}} \sin\left(\sqrt{1 - \frac{1}{4Q^2}} \omega_o t\right) u(t) \quad (2)$$

Determining K:

Unfortunately, the final value of the output voltage is zero ($V_o(t \rightarrow \infty) = 0$), since the bandpass filter cannot pass the DC-component of the step-function. However, we can calculate the first peak which occurs for $\sin\left(\sqrt{1 - \frac{1}{4Q^2}} \omega_o t_p\right) = 1$. For $Q > 3$, this is equivalent to $\sin(\omega_o t_p) = 1$, and results in $\omega_o t_p = \pi/2$. Substituting this into (2), we find that the first peak value is:

$$V_{p_1} = \frac{2AK}{\sqrt{4Q^2 - 1}} e^{\frac{-\pi/2}{2Q}} \quad (3)$$

And again, for $Q > 3$;

$$V_{p_1} \approx \frac{AK}{Q} e^{\frac{-\pi/2}{2Q}} \quad \text{with less than 1.5\% error.} \quad (4)$$

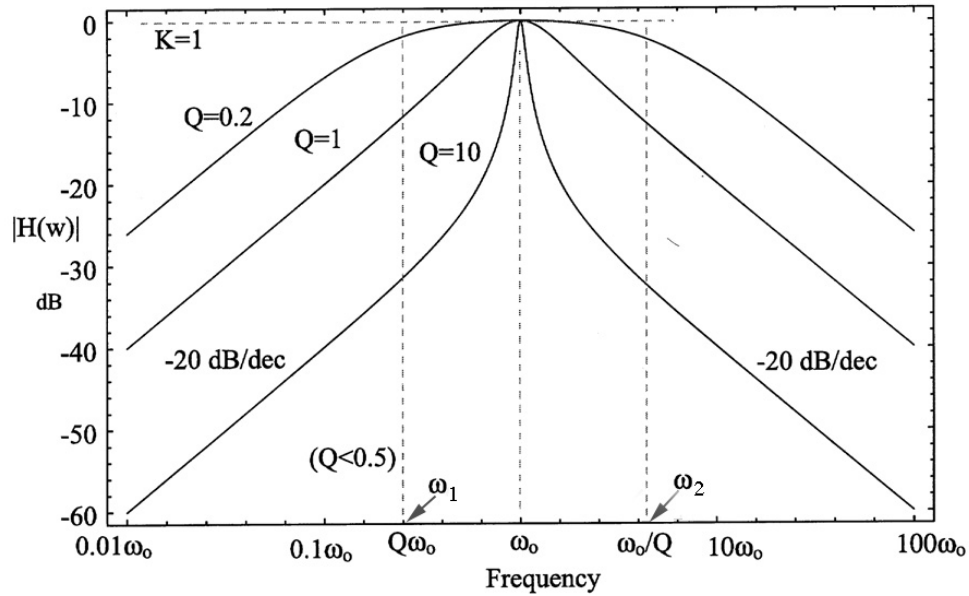


Figure 1a. Band-pass frequency response, magnitude.

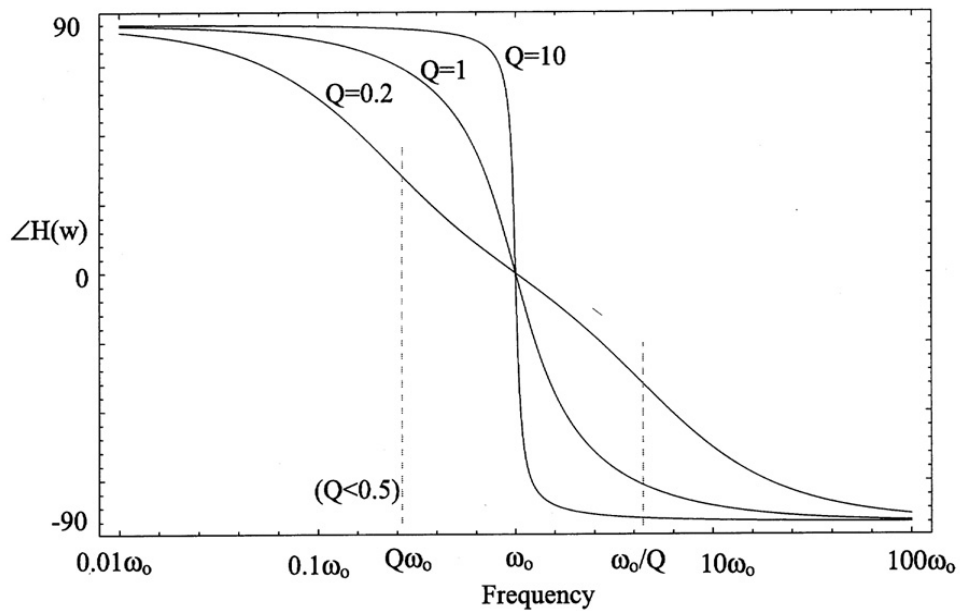


Figure 1b. Band-pass frequency response, phase.



Determining ω_0 :

The oscillation frequency of the decaying sinusoid is

$$\omega_R = 2\pi/t_R = \omega_o \sqrt{1 - \frac{1}{4Q^2}} \tag{5}$$

For $Q > 3$, $\omega_R \approx \omega_0$ with less than 1.5% error.

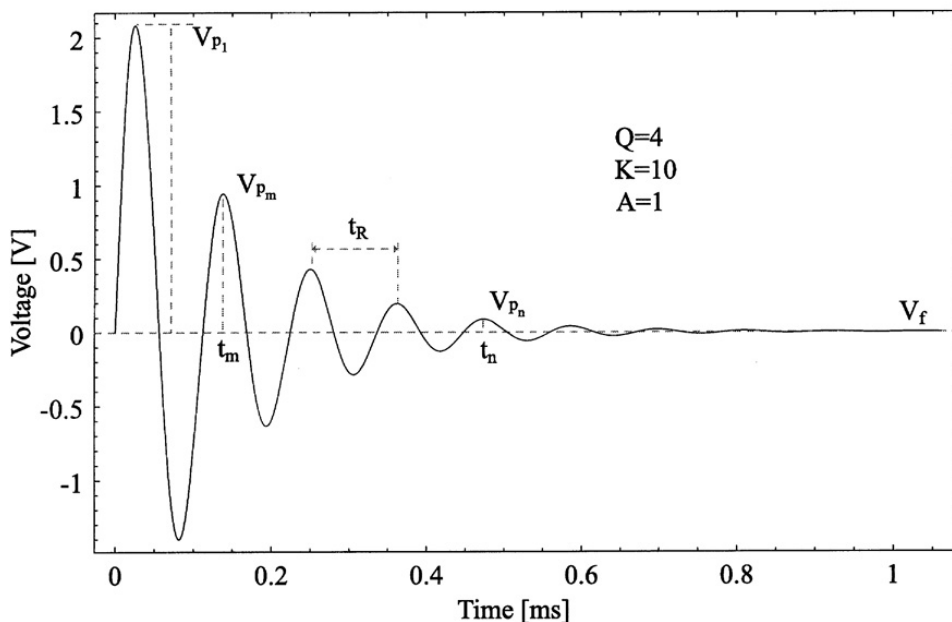


Figure 2. Band-pass time domain response.

Determining Q:

The same approach as Method 2 in Lab #1 is used (Decaying Peaks Approach). To repeat,

$$Q = \frac{\frac{1}{2} \omega_o \Delta t}{\ln\left(\frac{V_{pm}}{V_{pn}}\right)} \tag{6}$$

where $\Delta t = t_n - t_m$ and $V_f = 0$ (band-pass filter). Knowing Q, we can determine K from equations (3) and (4). As usual, for large Q, you need to take peaks which are far away from each other so as to minimize the effect of the measurement error.



What if $0.5 < Q < 3$?

In this case, the natural resonant frequency (ω_0) is different from the resonant frequency (ω_R), since

$$\omega_R = \omega_o \sqrt{1 - \frac{1}{4Q^2}}.$$

Q is determined using equation (6) but care must be taken to find the exact value and times of the peaks (which could be very broad). To avoid a complex equation between ω_0 and Q, an iterative approach is used:

Take a good guess at ω_o ($\omega_o \cong \omega_R$ at the beginning) and calculate Q using equation (6). Use this value of Q to calculate a new ω_0 using equation (5), then a new Q using equation (6). After 2 to 3 iterations, the exact values of Q and ω_0 are found.

To determine K, we have at the first peak, $\sin\left(\sqrt{1 - \frac{1}{4Q^2}} \omega_o t_p\right) = 1$ and

$$\omega_o t_p = \frac{\pi/2}{\sqrt{1 - \frac{1}{4Q^2}}} \tag{7}$$

$$V_{p1} = \frac{2AK}{\sqrt{4Q^2 - 1}} e^{\frac{-\pi/2}{2Q\sqrt{1 - 1/4Q^2}}}$$

Equation (7) can be used with the known values of ω_0 and Q to calculate AK (or K, knowing the input step function A).

What if $Q < 0.5$?

For $Q < 0.5$, the step response is dominated by the first pole-zero at $\omega_1 = \omega_o Q$, and the fall-time (not the rise-time) of the step response can be determined by ω_1 (problem 3 in pre-lab). We cannot really determine Q and ω_2 from time domain measurements alone, but this is not important. The system performance is limited by ω_1 , and we know how to find its value!



Experiment No. 2. Active Bandpass Filters; Frequency Shift Keying Communications

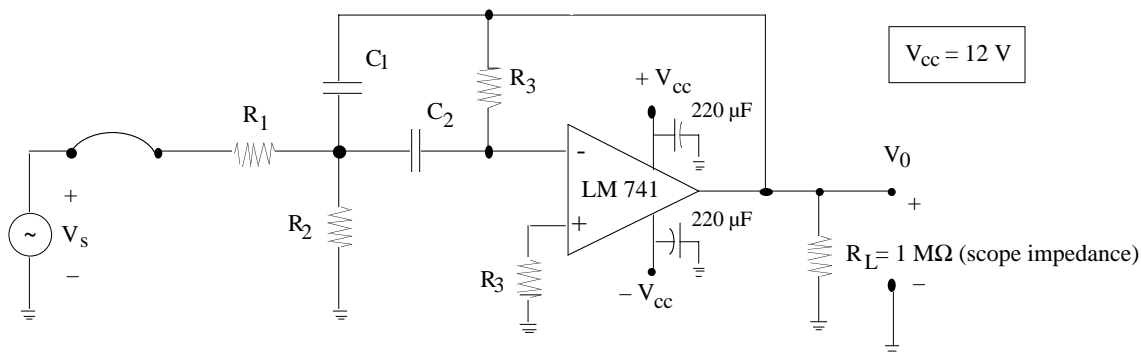
Goal: The goal of this experiment is to measure the response of a bandpass filter for different Q, in time and frequency domain. Also, to use the bandpass filter as a simple FSK receiver in a digital system.

- Read this experiment and answer the pre-lab questions before you come to the lab.

- Equipment:**
- Agilent E3631A Triple output DC power supply
 - Agilent 33120A Function Generator
 - Agilent 34401A Multimeter
 - Agilent 54645A Oscilloscope

1.0 Active Bandpass Filters:

An active multiple-loop bandpass filter is shown below:



The filter response is given by:

$$\frac{V_0}{V_s} = H(s) = \frac{-\left(\frac{1}{R_1 C_2}\right) s}{s^2 + \left(\frac{1}{C_1} + \frac{1}{C_2}\right) \frac{1}{R_3} s + \frac{1}{R_3 C_1 C_2} \left(\frac{1}{R_1} + \frac{1}{R_2}\right)} \quad s = j\omega$$



with

$$|K| = \frac{R_3}{R_1} \frac{C_1}{C_1 + C_2} \qquad Q = \frac{\sqrt{\frac{R_3}{R_2} \left(1 + \frac{R_2}{R_1}\right)}}{\sqrt{\frac{C_1}{C_2}} + \sqrt{\frac{C_2}{C_1}}}$$

$$\omega_0 = \sqrt{\frac{1}{R_3 C_1 C_2} \left(\frac{1}{R_1} + \frac{1}{R_2}\right)} \qquad \Delta\omega = \frac{1}{R_3} \left(\frac{1}{C_1} + \frac{1}{C_2}\right)$$

Notice the negative sign in the numerator. This is an *inverting* active filter.

where

$$\begin{aligned} |K| &= \text{Gain at resonance } (\omega_0) \\ \omega_0 &= \text{Resonant frequency} \\ \Delta\omega &= \text{3- dB Bandwidth} \\ Q &= \text{Quality factor } \approx \frac{\omega_0}{\Delta\omega} \text{ (for small } \Delta\omega) \end{aligned}$$

In order to simplify the design, choose $C_1 = C_2 = C$ and:

$$R_1 = \frac{Q}{|K| \omega_0 C}, \qquad R_3 = \frac{2Q}{\omega_0 C}$$

$$R_2 = \frac{Q}{(2Q^2 - |K|) \omega_0 C} \qquad \text{with} \qquad |K| < 2Q^2$$

which results in:

$$|K| = \frac{1}{2} \frac{R_3}{R_1}, \qquad \omega_0 = \frac{1}{C} \left(\sqrt{\frac{1}{R_3} \left(\frac{1}{R_1} + \frac{1}{R_2} \right)} \right), \qquad \Delta\omega = \frac{2}{R_3 C}$$



and

$$Q = \frac{\sqrt{\frac{R_3}{R_2} \left(1 + \frac{R_2}{R_1}\right)}}{2}$$

This filter topology is good for low ($Q \sim 1$) to medium Q ($Q < 50$). For high Q , this topology gives a very large spread of resistance values and the value of ω_0 and Q become very sensitive to component values. Notice also that the filter peak gain, $|K|$, is always smaller than $2Q^2$. This means that low Q filters using this topology will have a gain around -3 dB to +3 dB.

For $C_1 = C_2 = C$ and relatively large Q and K ($R_2 \ll R_1$ and $R_3 \gg R_1$), we have:

$$|K| = \frac{1}{2} \frac{R_3}{R_1} \qquad Q \approx \frac{1}{2} \sqrt{\frac{R_3}{R_2}}$$

$$\omega_0 \approx \frac{1}{C \sqrt{R_2 R_3}} \qquad \Delta\omega = \frac{2}{R_3 C}$$

Low-Q Design:

1. Assemble the circuit on the breadboard using $C_1 = C_2 = C = 2.2$ nF and:

$$R_1 = 9.1 \text{ K}\Omega$$

$$R_2 = 9.1 \text{ K}\Omega$$

$$R_3 = 20 \text{ K}\Omega$$

This should result in $f_0 \sim 7.6$ KHz, $Q \sim 1$ and $|K| \sim 1$.

Connect the power supply (± 12 V) and make sure to use the 220 μ F noise canceling capacitors.


- Draw the circuit in your lab notebook.
- 2. Connect the Agilent 33120A waveform generator to the input and set it to $V_{ppk} = 1$ V. Connect the output to the Agilent scope.
 - Make a fast frequency sweep (with the knob) and determine f_0 and $|K|$.
- 3. Make an accurate frequency response measurement of this filter (you can do it using the V_{ppk} readout, or the FFT mode) from $\sim 0.1 f_0$ (500 Hz) to $\sim 10 f_0$ (100 KHz).

Be careful! With bandpass filters, you should not measure at 20 Hz, 50 Hz, 100 Hz, etc. as we did with standard amplifiers! The correct way to measure filters is to determine the frequency where the response falls by 3 dB, 6 dB, 10 dB, 15 dB, 20 dB, etc. ...

(You will find that the response is very board. However, this filter is still sharper than the typical midband audio tone controllers which have a Q of around 0.5.)

- 4. Calculate the 3-dB bandwidth and the Q of the filter, and show it to the T.A.



5. Measure the phase delay of V_O/V_S at $0.5 f_0$, f_0 , $2 f_0$ (you need to connect the output to Channel 1 and the input to Channel 2 and use the phase delay measurement under the Measure  menu).

High-Q Design:

1. Change the resistances to:

$$R_1 = 15 \text{ K}\Omega \quad R_2 = 240 \text{ }\Omega \quad R_3 = 300 \text{ K}\Omega$$

(keep $C_1 = C_2 = C = 2.2 \text{ nF}$).

This should result in $f_0 \sim 8.5 \text{ KHz}$, $Q \sim 18$, $|K| \sim 10$.

Set the Agilent 33120A to give a $V_{ppk} = 500 \text{ mV}$.

- Draw the circuit in your lab notebook.
2. Make a fast frequency sweep (with the knob) and determine f_0 and $|K|$.
(Be careful, the amplitude will change very quickly around 8-9 KHz and a high frequency resolution is needed). Determine f_0 to a 10 Hz resolution.
3. Make an accurate frequency response measurement from $\sim 0.1 f_0$ ($\sim 1 \text{ KHz}$) to $\sim 10 f_0$ ($\sim 100 \text{ KHz}$). Again, determine the frequency where the response falls by 3 dB, 6 dB, 10 dB, 15 dB, 20 dB and after this at 5 dB steps (25, 30, 35, 40 dB, etc...until you reach $\sim 100 \text{ KHz}$).

(Make sure that you are not clipping the waveform on the display and that you are not measuring V_{ppk} using a "small" waveform on the scope display. For highest accuracy, always choose a vertical setting to "fill" your screen.)

You will find that the response is quite sharp and that the filter response varies quickly around f_0 .
4. Calculate the 3-dB bandwidth and the Q of the filter, and show it to the T.A.
5. Measure the phase delay of V_O/V_S at $0.8 f_0$, f_0 and $1.25 f_0$.

2.0 Time Domain Response of High-Q Bandpass Filters:

1. a. Set the Agilent 33120A to give a square-wave at f_0 ($\sim 8.5 \text{ KHz}$) and $V_{ppk} = 2 \text{ V}$. Plot V_i and V_o and label the ppk values. Do you know why it is a sinusoid?
b. Measure V_i and V_o in dB in the frequency domain (f_0 , $3f_0$ and $5f_0$.)
2. Change the input to a square-wave frequency of 200 Hz and $V_{ppk} = 1\text{V}$. Connect the sync. signal of the Agilent 33120A to the Ext. trigger of the scope, and set the trigger to External. Set the timebase to see at least one complete cycle of V_o . Plot V_i and V_o , and pay attention to the slope of output signal at the transition.
3. From the ripples, measure the oscillation frequency. Is it close to f_0 ?



4. From the ripples, take several (4 separated points) peak voltages (V_p) and time (t) measurements. This is needed to calculate the decay time or the Q of the filter.
5. Measure the final voltage of V_O (after the ripple dies down).

3.0 FSK Modulation and Digital Communication Systems:

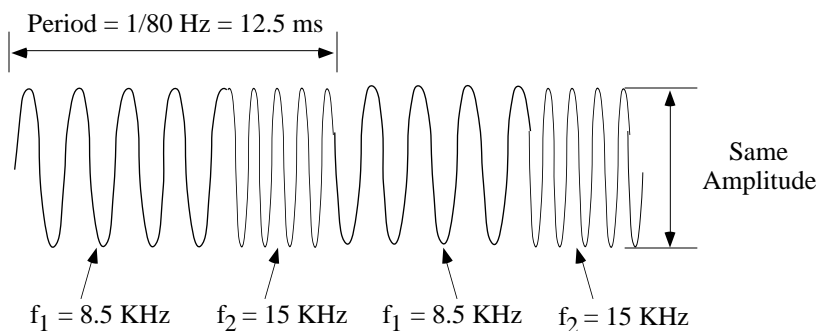
Set the Agilent 33120A to give an FSK signal with $V_{ppk} = 500$ mV.

$f_1 = f_0$ (~ 8.5 KHz) [First Frequency]

$f_2 = 1.7 f_0$ (~ 15 KHz) [Hop Frequency]

and FSK Rate = 80 Hz [FSK Rate]

The FSK signal should look like:



To Set an FSK Waveform:

1. First set f_1 normally for ~8.5 KHz (whatever your f_0 is) and $V_{ppk} = 500$ mV.
2. Press to enter into FSK mode.

To Set the "Hop" Frequency:

3. a. Press Recall Menu. You will see
 - b. Move down one level: Press . You will see .
 - c. Enter 14 KHz (or whatever your frequency is) using the command.

To Set the FSK Rate:

4. a. Press Recall Menu. You will see
 - b. Move across one level: Press . You will see .



- c. Move down one level: Press . You will see .
- d. Enter 80 Hz using the command.

Connect the SYNC output of the Agilent 33120A to the scope external trigger input. Set the scope trigger to "External". Now, look at the waveform with a long enough timebase to see the FSK signal. You should see clearly the FSK signal.

- 5. Connect the FSK signal to the input of the Q = 18, K = 10 filter.

The ~9 KHz signal should pass unattenuated and the ~15 KHz should be strongly attenuated (around -30 dB). Thus the 9 KHz is equivalent to a digital "1" and the 15 KHz is equivalent to a digital "0".

- 6. Draw the resulting waveform on your notebook and label the average peak voltage for "1" and the average peak voltage of "0".

Congratulations; you have built a simple but very useful digital FSK receiver. In this case, you are sending a stream of 101010... (square-wave) but it could be anything such as an ASCII code with 11000101...



Experiment No. 2.

Active Bandpass Filters; Frequency Shift Keying Communications

Pre-Lab Assignment

- Draw the bandpass filter circuit for $f \ll f_0$ and $f \gg f_0$, and calculate V_O/V_S at these frequencies.
 - Knowing that capacitors are “open” for $\omega \rightarrow 0$, and “short” for $\omega \rightarrow \infty$, calculate the input impedance of the active filter as seen by the source at $f \ll f_0$ and $f \gg f_0$.
 - Why is there a resistor R_3 connected to the (+) input of the op-amp (again, think non-ideal op-amp properties).
- Make sure that the values of R's and C's given in the experiment do result in the quoted f_0 , Q, K for the low-Q and high-Q cases.
- The response of a general bandpass filter with gain |K| at resonance is given by:

$$\frac{V_o(s)}{V_s} = H(s) = \frac{K(\omega_0/Q)s}{s^2 + (\omega_0/Q)s + \omega_o^2} \quad \begin{array}{l} (s = j\omega) \\ f_o = 10 \text{ KHz} \end{array}$$

$$\text{and } |H(\omega)| = \frac{K \left(\frac{\omega_0}{Q} \right) (\omega)}{\sqrt{[\omega_0^2 - \omega^2]^2 + \left[\left(\frac{\omega_o}{Q} \right) \omega \right]^2}} \quad (\text{Magnitude of } H(\omega))$$

- Derive $|H(\omega)|$
- For $K = 1$ and using MATLAB, plot on the same graph (dB, log f) the response of this filter for $Q = 0.4$, $Q = 2$ and $Q = 20$. The horizontal scale should be from $0.01 f_0$ to $100 f_0$. The vertical scale should be from 0 to -60 dB.
- From the plot, determine the 3-dB bandwidth ($\Delta\omega$) for each case, and determine Q from $Q = \omega_0/\Delta\omega$. What do you notice?
- Derive the equation of the phase of $H(\omega)$ and plot on the same graph the phase for $Q = 0.4$, $Q = 2$ and $Q = 20$ from $0.01 f_0$ to $100 f_0$. What is the phase of $H(\omega)$ as $\omega \rightarrow 0$, $\omega \rightarrow \infty$ and $\omega = \omega_0$?
- Prove that, at resonance ($\omega = \omega_0$), $|H(\omega_0)| = K$ and the phase of $H(\omega_0)$ is equal to 0° .



Experiment No. 2.

Active Bandpass Filters; Frequency Shift Keying Communications

Lab Report

- Draw the filter circuit and neatly summarize all your measured data for the low-Q and high-Q cases. Putting the data in a table form will result in an excellent presentation and you should do it. Comment on your measured vs. calculated values for K, f_0 , Q.
 - Plot on the same Bode Plot (dB, $\log f$) the filter response for $Q \sim 1$ and $Q \sim 18$. The vertical scale should be around +20 dB to -40 dB and the horizontal scale should be from around 500 Hz to 100 KHz (depending on your measured values). Clearly label the resonant frequency (f_0), the gain at resonance (K) and the bandwidth (Δf) of the filters.
- For the high-Q filter, calculate f_0 and Q from time-domain measurements. For the Q calculations, in one case use two peaks which are close to each other, and in another case, use two peaks which are farthest away from each other. Compare the results with the frequency domain measurements comment.
- For a square-wave of $V_{pk} = 1V$ ($V_{ppk} = 2V$) and $f = f_0$ (~ 8.5 KHz), calculate the fundamental, third, fifth and seventh harmonic levels in Volts and dB before and after it passes by the high-Q filter. Compare with frequency domain measurements for V_i and V_o .
 - Calculate the THD (total harmonic distortion) of the output sinusoidal waveform. Do you consider this waveform to be a "clean" sinusoid?
- Using the formulas given in the experiment, design a filter with $Q = 40$, $K = 100$ and $f_0 = 8.5$ KHz. Choose $C = 2.2nF$. What is the main difference in resistances with the design of $Q = 18$, $K = 10$?
 - What component(s) would you change in the $Q = 18$, $K = 10$ filter to get an $f_0 \sim 1.8$ KHz? What will happen to Q and K for the new 1.8 KHz filter?
 - Drawing of the filter circuit above with the calculated values for R's and C's. Circle the components that:
 - determine the bandwidth,
 - determine Q,
 - determine ω_0 .
- A wideband (200 KHz) FSK signal is composed of a 10 MHz signal (f_1) and a 10.2 MHz signal (f_2). Determine the optimum time T_b for a digital bit (1 or 0).
 - Determine the minimum Q of the filter needed so as to pass the 10 MHz signal and attenuate the 10.2 MHz by at least 20 dB. (Hint: Q is very large, $Q > 100$). Either calculate Q analytically (and show it in detail) or include an expanded view of a Matlab plot (9.6-10.4 MHz) showing the attenuation at 10.2 MHz and the calculated Q.