Elec 240 F18 Final Project: RF Transceiver System

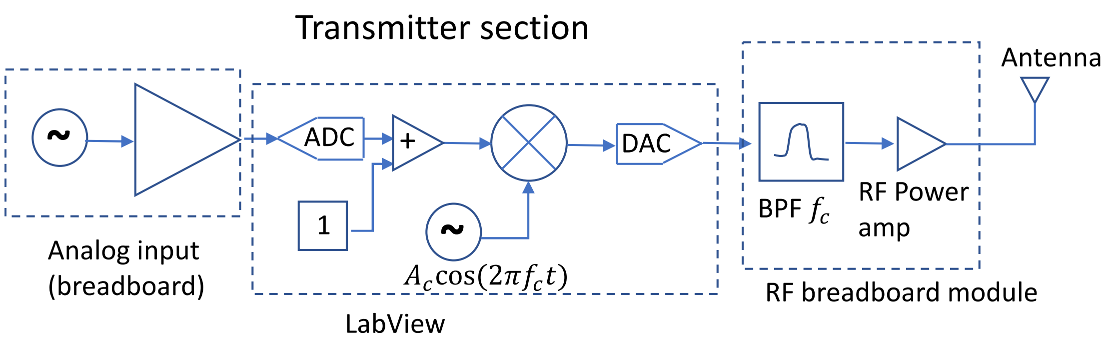
1 Overview

In this final project you will design and build a working two-way audio transmitter/ receiver system. The system will use amplitude modulation (AM) of a carrier in the range of 160-190 kHz. You will build and test various analog circuits and also digital signal processing blocks, get the whole system working, and present your results to the lab assistants. You will have three lab sessions (plus optionally Thanksgiving week) to work on your setup. You can divide up the work in any way you like during the 3 sessions. The lab assistants will schedule 5-minute presentation slots for each team which will take place during the last lab session of the semester. You will also need to upload a written report as usual.

2 Specifications & system overview

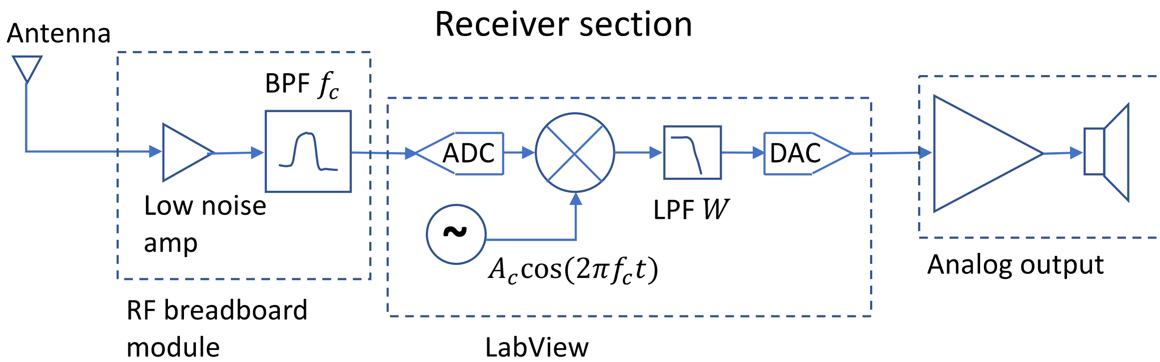
The specifications are simple: the system needs to be able to transmit and to receive audio signals across the lab. On the transmitter side, the blocks will be (as shown in the figure below):

- Analog input
- Modulation of carrier by analog signal (handled in the digital domain by LabView)
- RF bandpass, amplifier & transmitter (on the RF breadboard module)
- Antenna



On the receiver side the blocks will be (as shown in the figure below):

- Antenna
- RF amplifier and bandpass filter
- Demodulation of received signal & lowpass filtering
- Audio output



These blocks will be explained more fully below.

3 Notes on system design

3.1 Notes on radio design & the RF module

We choose the 160-190 kHz band for a few reasons. First, this portion of the RF spectrum is designated by the FCC as a so-called industrial/ scientific/ medical (ISM) band. FCC regulations allow anyone to use this frequency band as long as they transmit less than 1 watt of radiated power, roughly speaking. Second, the signal frequency is low enough that we don't have to worry about using RF circuit construction techniques.

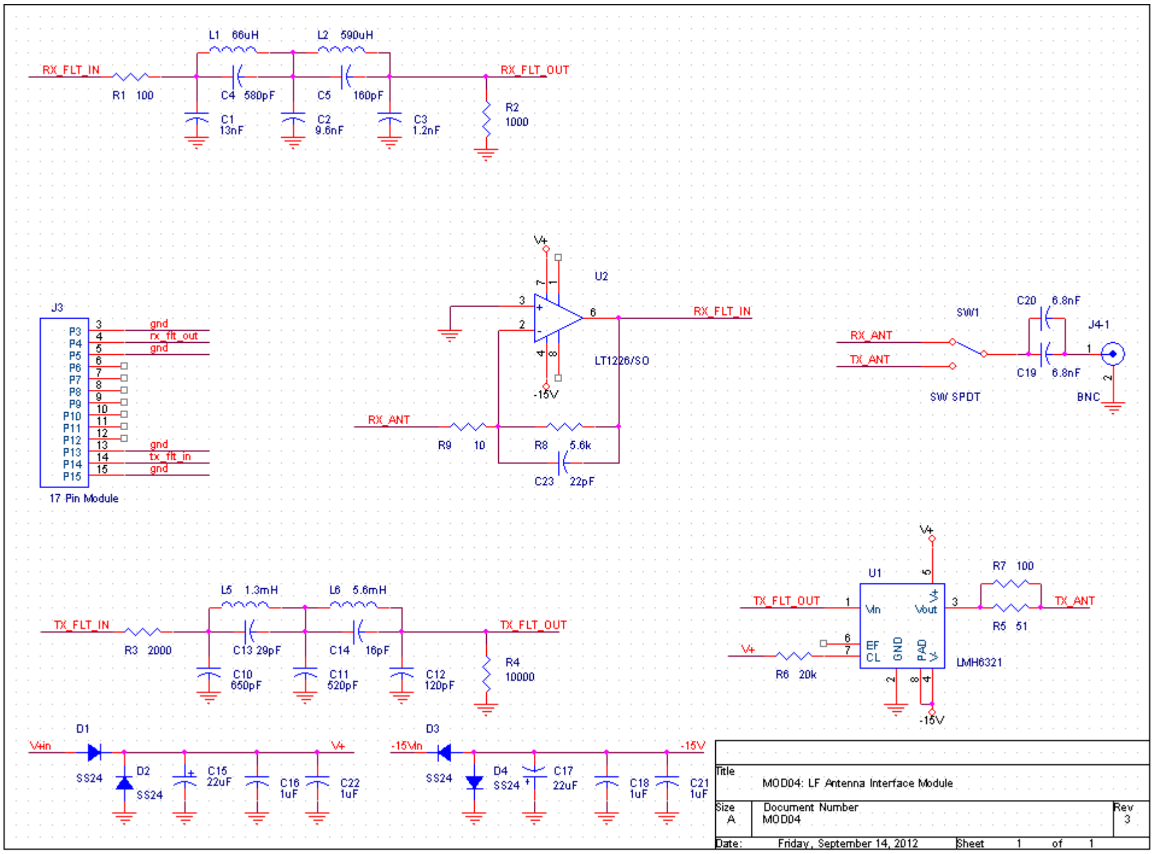
This ISM frequency band is not very convenient for most purposes – the wavelength λ of EM waves is $\lambda = \frac{c}{f}$ where the c is the speed of light (~3.0e8 m/s). Antennas ideally (if they are so-called electric dipoles) need to be about half a wavelength in size, which in this case would be ~ 3e8/ 1.5e5 /2 ~ 1 km. To get around this to some extent we use a loop antenna – a so-called current dipole instead of electric dipole. We can use many loops of wire in this way (our antenna has 8 loops inside its plastic case). If you take apart an AM radio (which is designed to operate at only about 10x higher frequencies, near 1 MHz) you will see an antenna coil inside the radio for this reason. By contrast, FM radios, which operate near 100 MHz, typically use electric dipole antennas on the order of 1.5 meters in size. WiFi routers, at 2.45 GHz, use electric dipole antennas of about 2.5 inches in size.

Because our antenna is so small compared to the optimal size, the efficiency of transmission of the antenna will be rather low. Since the receivers will all be in the same room though, we will still be able to demonstrate transmission & reception.

The RF module contains all the RF circuits for the transmitter and receiver so all we have to do is the baseband portion. Here is what the module looks like. You hook up one antenna to the BNC connector and switch between TX and RX modes using the toggle switch. The RX and TX signals appear on pins on the breadboard. If this module is installed in the normal slot (3rd from the left) on the breadboard then the transmitter input pin will be pin 39 and the receiver output pin will be pin 29 on the interface module. The adjacent pins are ground.

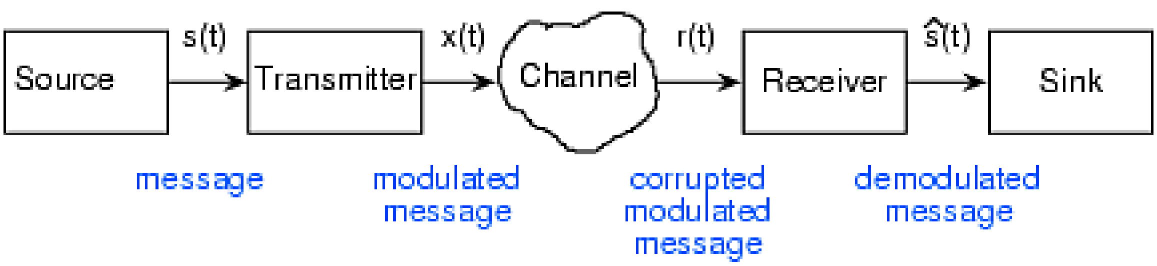
Below is the schematic for the RF module. Here is a brief explanation of how it works.

- Capacitors C10-C12, inductors L5 and L6, and resistors R3 and R4 form the lowpass filter for the transmitter signal. This is a 5th order elliptic filter with a 250 kHz cutoff frequency.
- The filtered transmitter signal is connected to the input of U1, an [LMH6321 300 mA high speed buffer.] This is basically a high power, wide bandwidth op amp wired as a unity gain buffer.
- With SW1 in the lower position, the output of U1 is connected to the antenna through C20 (C19 is not installed). C20 cancels the inductive component of the antenna impedance to provide a more efficient power transfer.
- With SW1 in the upper position, the antenna is connected to the input of the circuit consisting of U2, R8, R9, and C23. U2 is an [LM7171 high speed op-amp.]
- Capacitors C1-C5, inductors L1 and L2, and resistors R1 and R2 form another 5th order elliptic lowpass filter with a 250 kHz cutoff. This helps to reduce aliasing when the amplified receiver signal is digitized.
- Diodes D1-D4 and capacitors C15-C19 and C21-22 protect the board against damage should someone hookup the power supply leads backwards, and provide some filtering of power-supply noise.



3.2 AM signal modulation, transmission, and demodulation

We will build an AM signal processing system as follows. There will be an input audio signal $s(t)$. This signal will amplitude-modulate a carrier wave at frequency f_c; the resulting modulated carrier is $x(t)$. This signal will be sent over the channel (a wireless channel in this case). The received signal at the receiver is $r(t)$. The receiver will convert this signal back to a baseband audio signal, $\hat{s}(t)$.



The transmitted signal $x(t)$ is written simply as the product of a modulation signal $m(t)$ and a carrier wave at frequency f_c :

$$x(t) = A_c(1 + m(t))\cos(2\pi f_c t)$$

Where A_c is the amplitude of the carrier wave. The modulation signal $m(t)$ is related to the audio (so-called baseband) signal $s(t)$ by a simple scaling factor in this case. For our purposes the modulation signal should be close to but not exceed unity.

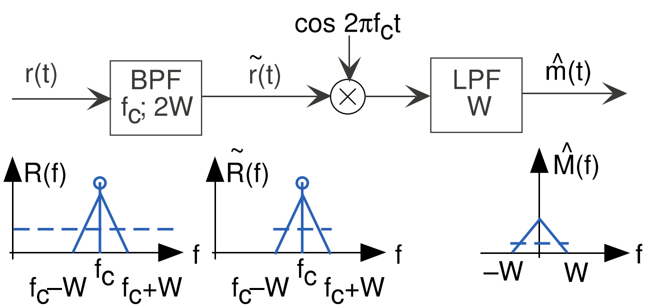
We assume that $m(t)$ has a corresponding Fourier transform $M(f)$, which has frequency content extending from about DC up to some maximum frequency W, the bandwidth of the signal. In the present case $m(t)$ corresponds to an audio signal so W is going to be a few kHz, or 20 kHz at most. More precisely, we say $m(t)$ is *band-limited:* $M(f) \approx 0$ for $|f| > W$. We further assume the carrier is at a much higher frequency than the bandwidth: $f_c \gg W$. In the frequency domain (for $f > 0$) the corresponding transmitted signal is:

$$X(f) \propto A_c[\delta(f - f_c) + M(f - f_c)]$$

The first term is energy in the carrier and the second is the information centered within a narrow band $f_c \pm W$.

The channel attenuates and phase shifts the signal but these are both linear operations, so the receiver eventually sees the received signal $r(t)$, from which the audio signal can be recovered.

Converting the transmitted signal back to a baseband $\hat{m}(t)$ is quite similar. The figure below describes the process. The received signal $r(t)$ ($R(f)$ in the frequency domain) is first passed through a bandpass filter (BPF) centered at the carrier frequency f_c, creating $\tilde{r}(t)$ ($\tilde{R}(f)$in the frequency domain.) The width of this filter needs to be (at least) $2W$ because it has to pass both the upper and lower sidebands surrounding the carrier. As shown in the picture below, $\tilde{R}(f)$ is pretty much the same as $R(f)$ except the noise outside the pass band has been eliminated. Then $\tilde{r}(t)$ is multiplied by another sinewave at the same carrier frequency, $\cos(2\pi f_c t)$. This moves the signal back down to baseband. Further low pass filtering (now with bandwidth W rather than $2W$) removes unwanted mixing products at $2f_c$. Thus we recover $\hat{m}(t)$ (or $\hat{M}(f)$ in the frequency domain).



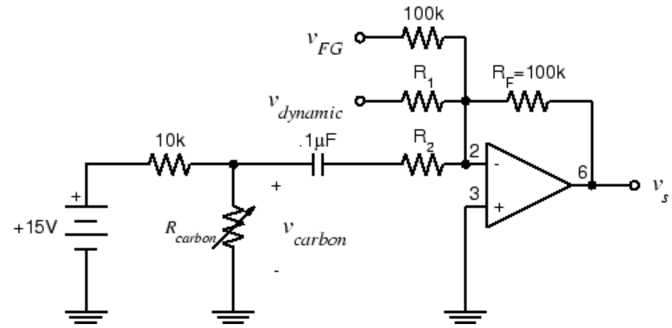
3.3 Modulation/ demodulation in the digital domain

We have to build a circuit that multiplies our carrier wave by the modulation signal. It is difficult to accurately multiply two signals together using analog components. However, it is trivial multiply two signals together in the digital domain, and filtering is much easier as well. So we will digitize our modulation signal, multiply it with a carrier in the digital domain in the LabView environment, and then convert it back to analog for transmission. We will do this both on the transmit and receive sides of the system.

4 Constructing subsystems

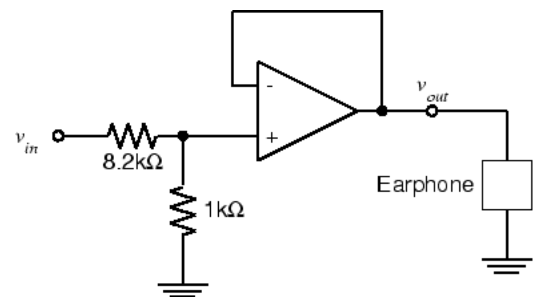
You can build and test these subsystems in any order. Test results for subsystems should all be collected in your final report.

4.1 Analog signal generation



If you have taken down your signal amplifier, if it needs to be cleaned up, now would be a good time to rebuild it. As a reminder the figure shows the signal source. You can connect signals from either the carbon or the dynamic mic or from a function generator or other source (e.g. analog audio from a sound card or phone). Verify that this circuit still works properly.

4.2 Analog speaker driver



Assemble (or re-assemble) the analog speaker driver from earlier in the semester. Verify that it works properly by verifying that you can hear a test signal in the earpiece.

4.3 Building LabView VIs

You should build the LabView VIs that will be used for modulation & demodulation. See the LabView VI Instructions for details on building and testing the LabView VIs.

4.4 Characterizing the RF modules

You should characterize the performance of the RF modules via Bode plots. For characterizing the Rx module, feed a low amplitude sine wave into the antenna port from the function generator on the Virtual Bench. Measure the output from the module (pin 29) and make a Bode plot (dB magnitude vs. log10 frequency – no need for keeping track of phase.) For characterizing the Tx module, hook up the antenna and feed a signal to be transmitted into the Tx port. Use a BNC T connector and scope probes or another BNC cable to measure the voltage at the antenna port. Make a Bode plot of the magnitude of the voltage gain vs. frequency. The easiest way to do this is to manually select a handful of frequencies at which to make the measurement.

Note: when making Bode plots you should sample the data at fairly closely spaced frequencies near the 3dB points, but away from such points you can get by with a frequency point per decade or so. For points where the amplitude is so low it's hard to measure on a scope, there are two tricks you can use. First, you can increase the amplitude of the driving sinewave: since you are measuring gain, the extra amplitude gets factored out at the end. Second, you can use the RMS measurement function on the Virtual Bench to improve the accuracy of your measurements (note for a sinewave, RMS amplitude is $\sqrt{2}$ lower than the peak amplitude.)

5 System testing

5.1 Self test

If each component of your system has passed the test given in its construction section, then your system should be functional. Because the antenna is switched between the transmitter and receiver, and because it is not possible to run both the transmitter and receiver VIs at the same time, it is not possible to test both parts of the system simultaneously. However, we can verify that both halves are functioning properly by using a second antenna connected to either the oscilloscope or the function generator.

Get two antennas. Connect one to the Antenna Interface and the other to the oscilloscope. Place the two antennas close to each other. Set the switch on the Antenna Interface to transmit. Connect the function generator and set it to produce a 1 kHz 1 V sine wave. Start the Transmitter VI. You should see an AM modulated signal on the scope.

Disconnect the function generator. Speak into the handset microphone in a normal voice and verify that full modulation is being achieved. If not, adjust R2 until it is. Plug in the dynamic microphone. Speak into it and adjust R1 if necessary to achieve full modulation.

Stop the transmitter VI. Remove the antenna from the oscilloscope and connect it to the function generator. Set the function generator frequency to 174 kHz. Set the switch on the antenna interface to Receive. Start the receiver VI. You should hear a 1 kHz tone from the earphone.

5.2 Full system test

The maximum range you can achieve in a self test is limited by the length of the antenna cable to about 5 feet. Since the specifications call for a range equal to the length of the lab (about 50 feet) you will need the cooperation of another lab group to perform a full system test.

Find another lab group to work as your partner for full system testing and characterization. If there are an odd number of groups, one of the labbies will serve as your partner. First verify that your systems work together at short range (e.g. at the same or adjacent lab stations). The full range test is very simple: with you and your partner group at the two test stations, you should be able to carry on an intelligible, two-way (half-duplex) conversation.

Although the antennas we are using are not highly directional in the sense that they do not produce a narrow beam, they are directional in the sense that they are more sensitive in some directions than others. This means that successful communication over long distances will require attention to orientation of the antennas. This may require a compromise between maximizing the received signal (with the transmit and receive antennas coplanar) and minimizing interference.

Since other groups may be trying to communicate at the same time you are, there is potential for interference. Since we have a 30 kHz band available and a speech signal has a bandwidth of only about 3 kHz, there are a number of different channels available.

6 Measuring system performance

When your system (and your partner group's) has passed the full range test, you should characterize it by getting quantitative performance characteristics at three different ranges: 5 ft, 15 ft, and 50 ft. Characterization will consist of 2 pairs of measurements made at each of the 3 ranges (a total of 2 x 2 x 3 = 12 measurements).

6.1 Preparation

Set the function generator to produce a 1 kHz sine wave. Unplug the dynamic microphone and the handset. Adjust the function generator amplitude to produce a fully modulated signal. With your DMM set to AC volts, measure and record the function generator output voltage. Set the volume control to maximum loudness.

6.2 Procedure

At each test distance perform the following procedure. At the receiving end disconnect the function generator. Carefully orient the transmitter and receiver antennas for maximum received output. Measure the *RMS* value of the receiver output. If you are using the prefabricated receiver VI, there is a display of the RMS value of the demodulated output in the lower left corner. If you have made your own receiver, you can add an RMS display using the Amp and Level block from the Signal Analysis palette. Or you can measure the voltage voutvout at the output of the earphone driver. In this case you must use a true RMS meter (available from the shop) for this measurement, not your DMM or scope. The voltage you measure will be the *signal plus noise* voltage.

Without moving either antenna, have your partner group switch their antenna interface to Receive. Again measure the RMS value of the receiver output. This is the *noise* voltage. Compute the *signal plus noise to noise* ratio

$$20\log_{10}\left(\frac{v_{s+n}}{v_n}\right)$$

Repeat these two measurements using your system as the transmitter and your partner group's as the receiver. Record both sets of measurements. Repeat these measurements at the two other test distances.

Note: Be sure that the function generator is *disconnected* at the receiving end for all measurements.

7 Presentation

In addition to the lab writeup, you should present your results to the labbies. The presentation should be about 5 minutes and include one or a few slides. More details on the presentation expectations will be given in a separate document.