University of Kentucky

EE 422G - Signals and Systems Laboratory

Lab 7 – Band-Pass Modulation

Objectives:

- Understand the modulation of signals with real and complex sinusoids.
- Apply modulation to efficiently use available bandwidth for transmission.
- Implement an AM modulation system and examine impact of modulation index.

1. Background

Communication channels are restricted by their inherent physical properties, which are measured in terms of bandwidth and noise. Only frequencies within the channel's bandwidth can pass through with minimal distortion. Therefore, in the process of sending signals from one point in physical space (source) to another (sink), the signals must be modified according to the channel bandwidth. Channel bandwidth is determined by properties of the physical medium or set by a standard protocol (such as is done by the FCC). Modulation shifts baseband signals to a particular frequency range, and in many cases is implemented by multiplying a sinusoidal waveform with the baseband signals.

Amplitude Modulation (AM), as used in commercial radio, is an example of using modulation. Radio station frequency ranges are constrained by government regulators. For example a radio station can purchase a license to broadcast radio signals in the range of 720 kHz to 760 kHz, which limits the bandwidth to 40 kHz. Since the audio signal for radio ranges from about 20 Hz to 20 kHz, the radio transmitter uses the baseband signal to modulate the amplitude of a high frequency sinusoid (i.e. at 740 kHz) (**modulation**) resulting in a signal in the 720 to 760 kHz range. A radio receiver, on the other hand, must **demodulates** the signal to recover the original baseband signal.

2. Pre-Lab

1. a) Based on the modulation property, sketch the spectrum of the modulated waveform:

   $$s(t) = 1000\,\mathrm{sinc}(1000t)\cos(2\pi 5000t)$$

   b) Sketch the spectrum of the demodulated waveform:

   $$r(t) = s(t)\cos(2\pi 5000t)$$

   c) Describe the problems that would occur with the above modulation and demodulation process if the carrier frequency was only 400 Hz (rather than 5000 Hz).

2. a) Based on the complex modulation property, sketch the spectrum of the modulated waveform:

$$\hat{s}(t) = 1000\,\mathrm{sinc}(1000t)\exp(-j2\pi 5000t)$$

b) Sketch the spectrum of the demodulated waveform:

$$\hat{r}(t) = \hat{s}(t)\exp(j2\pi 5000t)$$

c) Describe the problems that would occur with the above modulation and demodulation process if the carrier frequency was only 400 Hz (rather than 5000 Hz).

3. **Lab Exercise**

Include all code generated to do the following exercises in an appendix. Make sure they are commented.

1. For this exercise you may need an mfile function, named *simpmod.m* located at: http://www.engr.uky.edu/~donohue/ee422/mfiles/simpmod.m
   Modify the *simpmod.m* script to verify your sketches in the Prelab Problems 1 and 2, which used the sinc function as the test signal. It would be best to have a symmetric time axis for this so the sinc function is symmetric without having to shift it in time. Do not use a PSD (averaging over a hopping window), as in the original script. Just use the FFT magnitude over a single segment of the waveform. PWELCH will do this if you specify your window length to be the same as the signal length. Explain the main features of your code in the procedure section, present the plots in the results section, and in the discussion section explain similarities and differences you see between the prelab sketches and the plots in the results section.

2. For this exercise you will need to download an audio file sampled at 48kHz. There are 3 such files, located at:

   http://www.engr.uky.edu/~donohue/ee422/Data/mfopt.wav
   http://www.engr.uky.edu/~donohue/ee422/Data/twNspec.wav
   http://www.engr.uky.edu/~donohue/ee422/Data/impNtrans.wav

   You just need to use one of these for this exercise.

   You will also need a hidden mfile function, referred to as a pfile, located at:
   http://www.engr.uky.edu/~donohue/ee422/mfiles/bbchan.p

   The file *bbchan.p* is an executable Matlab file that operates as an mfile function, except the contents cannot be viewed or edited. It was written to simulate transmission over a band-pass channel for which the pass-band parameters are unknown. The channel also adds noise to the signal to result in a 20 dB SNR, so the received signal will sound noisier. The wave files can be loaded into the Matlab workspace with:

   ```
   >> [y, fs] = wavread('filename');
   ```

   The audio file samples are included in vector *y* and sampling rate is in scalar *fs*. The audio file can be played with the command:

   ```
   >>  soundsc(y,fs)
   ```

For this exercise:

(a) determined the passband for the channel
(b) modulate the audio signal in *y* to another signal, (i.e. *ym*) so that it will efficiently pass through the channel with the least distortion
(c) send the modulated signal over the channel (i.e. *r* = *bbchan(ym,fs)* and demodulate.
(d) Filter the received signal, *r*, to reduce noise.

In the procedure section of the write-up explain the process used to find the pass-band range for the channel, the modulation scheme (use equations to limit words and ambiguity), and the filter used to reduce noise in the received signal. In the results section (at least) present the frequency limits you found for the channel, the PSD for the original audio signal, the modulated signal the demodulated signal and the filtered signals. Also listen to the transmitted and received waveform and qualitatively describe the differences. In the discussion section, explain how well your system worked relative to the goal of transmitting and receiving the signal with highest possible fidelity with the lowest level of noise.

3. An AM waveform can be represented mathematically by:

$$s(t) = A_c[1 + \delta\, m(t)]\cos(2\pi f_c t)$$

where $m(t)$ is the message (baseband signal), and $A_c \cos(2\pi f_c t)$ is the carrier. The δ parameter controls the percent modulation (related to the modulation index), which allows for a certain percentage of the carrier to be present in the modulation signal. The purpose for modulating this way is to allow for inexpensive (simple demodulators). For radio broadcast, a region may have a few transmitters (using more complex modulation systems) but many radio receivers. So the most cost effective approach is to make the receivers as inexpensive as possible. If $1 > |\delta\, m(t)|$ for all *t*, then a simple demodulation scheme can be used, given by:

$$r(t) = h(t) * |s(t)|$$

where * denotes convolution and *h*(*t*) is a low-pass filter with a cutoff frequency high enough to include the baseband signal without significant distortion. This demodulator can be implemented with a simple circuit using a coil, a diode, and a capacitor. For this exercise, amplitude-modulate one of the audio signals with a 16kHz carrier. Then demodulate it by simply applying a low-pass filter to the rectified (absolute value) of the modulated signal. In the procedure explain how this modulation scheme was implemented. Be sure to completely describe the low-pass filter used and the δ value selected. In the results section, show plots of the original waveform, the modulated waveform, and the demodulated waveform. Also include the PSDs for each and play the signals before and after demodulation. Identify any differences heard between the signals. Repeat this exercise for a δ that violates the $1 > |\delta\, m(t)|$ condition. In the discuss section