EE 422G - Signals and Systems Laboratory

Lab 8 Band-Pass Modulation

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November 7, 2011

Objectives:

- Understand the models for signal modulation 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. Channel bandwidth is determined by properties of the physical medium or set by a standard protocol (such as is done by the FCC). Signals supported by frequencies falling almost entirely within the channel bandwidth can be transmitted reliably. Frequency content outside this range is attenuated resulting in signal 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 available/allocated channel bandwidth. Modulation shifts baseband signals to a particular frequency range. In many cases modulation is implemented by multiplying a sinusoidal waveform with the baseband signals.

Modulation for real signal can be understood from the following trigonometric identity:

$$\sin(2\pi ft)\cos(2\pi f_c t) = \frac{1}{2}\left(\sin(2\pi(f+f_c)t) + \sin(2\pi(f-f_c)t)\right) \tag{1}$$

where f_c is the modulating frequency and f is the baseband signal frequency. Note the signal being modulated is shifted up and down the spectrum (double-sided) by the modulating frequency. For modulating a complex sinusoid with another complex sinusoid, the signal is only shifted in one direction:

$$\exp(j2\pi ft)\exp(j2\pi f_c t) = \exp(j2\pi(f_c + f)t) \tag{2}$$

All real-valued signal have double sided spectra (positive and negative frequencies) with even symmetry for the magnitude and odd symmetry for the phase. Therefore, because of this redundancy only one side is needed to represent the unique signal information. A complex signal can have a one-sided spectrum if the real and imaginary parts of the time domain signal are related through the Hilbert transform.

A circuit implementation of a complex signal is usually accomplished by creating 2 channels where one is sifted 90 degrees with respect to the other. One channel is the real component and the other is the imaginary component. The advantage of using a complex representation is that the signal can be represented by a one-sided spectrum, effectively cutting its bandwidth in half.. The one-sided spectrum concept is used in single-sideband modulation systems. While the complex signal system requires less bandwidth, it also requires more complexity for both the transmitter and receiver.

One of the simplest modulation schemes is Amplitude Modulation (AM), which is used in commercial radio (AM band). The frequency ranges of radio stations are controlled by government regulations. 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 broadcast bandwidth to 40 kHz. Since the audio signal for radio ranges from about 20 Hz to 20 kHz, the radio transmitter *modulates* the baseband signal (limited to 20 kHz) to a higher frequency by effectively multiplying it with a sinusoid (i.e. at 740 kHz). This results in a signal within the 720 to 760 kHz range, and contains the same information as the original baseband signal. A radio receiver, on the other hand, *demodulates* the signal by multiplying it with a sinusoid to shift signal down in frequency and recover the original baseband signal through filtering out the higher frequency spectral images.

2. Pre-Laboratory Exercises

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

   $$s(t) = 500\,\mathrm{sinc}(500t)\cos(2\pi 4000t)$$

   b) Sketch the spectrum of the demodulated waveform:

   $$r(t) = s(t)\cos(2\pi 4000t)$$

   c) Describe the problems that would occur with the above modulation and demodulation process if the carrier frequency was only 400 Hz (rather than 4000 Hz).

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

   $$\hat{s}(t) = 500\,\mathrm{sinc}(500t)\exp(-j2\pi 4000t)$$

   b) Sketch the spectrum of the demodulated waveform:

   $$\hat{r}(t) = \hat{s}(t)\exp(j2\pi 4000t)$$

   c) Would this modulation/demodulation have similar problems as in the real modulation of Problem 1 if the carrier frequency was only 400 Hz (rather than 4000 Hz)?

3. Laboratory Exercises

Include all code generated to do the following exercises in an appendix. Make sure the lines are commented so reader can understand the purpose of code sections and overall procedure.

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 pre-lab Problems 1 and 2, which used the sinc function as the test signal. It would be best to have a symmetric time axis (i.e. positive and negative time values) so the sinc function appears symmetric about $t = 0$. 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 explain similarities and differences with the pre-lab sketches in the **discussion 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. Since modulation shifts the waveform on the frequency axis, these waveforms will be upsampled to 96kHz to raise the Nyquist (aliasing) frequency to 48kHz. So any signal frequencies shifted above 48kHz will fold back in the spectrum causing distortion. Avoid this. The *resample* command in Matlab can be use for both upsampling and downsampling. You need to ensure the Nyquist frequency is high enough to prevent aliasing of the signal after modulation.

   You will 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 10 mW of white Gaussian noise so the received signal will sound noisy (note the weaker the input signal is, relative to 10mW, the noisier the received signal will sound). 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 upsample the waveform to 96kHz as done in the *simpmod.m* script to limit aliasing during the modulation and demodulation process. Use this sampling rate for the following problems: