ECE 4117

Experiment #2

Amplitude Modulation
and Demodulation

The expression for an AM waveform with carrier frequency ω_c, modulated by a message signal m(t), is

$$\phi_{AM}(t) = A[1+m(t)]\cos(\omega_c t)$$

In the equation below, the "A" part of this will be the DC that is added to either the audio oscillator signal or the voice signal before multiplication by the carrier $\cos(\omega_c t)$, The modulation index "m" mentioned in the manual is the ratio $(|m(t)|_{max})/A$. For example, in 2(b), below, A = 1 and peak voltage of m(t) = 1 results in m = 1. (In Lathi's textbook, the symbol μ is used in place of m for modulation index.)

When the modulation index m = 1 and m(t) is a low frequency sinusoid $\cos(\omega_m t)$ ($\omega_m << \omega_c$), we have

$$\phi_{AM}(t) = A[1+ \cos(\omega_m t)]\cos(\omega_c t)$$

$$= A \cos(\omega_c t)+A/2 \cos[(\omega_{c+}\omega_{m)t]+}A/2 \cos[(\omega_{c-}\omega_{m)t]}$$

This represents, in order, the carrier, a sideband above carrier by frequency ω_m, and a sideband below carrier by the same amount. For m = 1 the spectral lines representing these sidebands are exactly half the height of the one for the carrier, as we see by factors A and A/2, above. If m(t) is reduced by 50%, the sideband spectra will have a fourth of the amplitude of the carrier. (We call this 50% modulation). If m(t) is doubled (100% overmodulation, definitely not desirable!) the sidebands will have the same amplitude as the carrier.

We might ask questions. Why add a carrier to a DSBSC signal? Doesn't the DSBSC radio transmission contain all the information of importance?.

The answer is that with DSBSC, in order to demodulate (that is, recover m(t)), we have to "mix" (multiply) it by a replica of the carrier, which is to say, a sinusoid with the same (within very tight limits) frequency and phase. Thus, we must have a "local oscillator" at the receiver site, precisely following every cycle of the sinusoid that was use to move the baseband up to higher frequency. It's often a difficult task, (at least in the sense of requiring more complex and expensive electronics, and in the sense of reliable performance) because transmission through the atmosphere -- or other medium -- is subject to time-varying random phase shifts caused by fluctuations in temperature or density of the atmosphere or other medium of transmission..

So-called "ordinary AM", the subject of this experiment, is sometimes called "DSBWC", meaning "double sidebands with carrier". Adding a carrier with voltage amplitude greater than the maximum *voltage* variations of m(t) produces a time domain signal where a positive version of m(t) can be seen as an upper "envelope" going through the positive peaks of the total signal, and again, with opposite phase, going through the negative peaks. Now it becomes possible to rectify to eliminate either the below-axis part or the above-axis part of the waveform. A simple RC circuit removes the radio frequency component, capacitor coupling blocks the DC, and what was previously an envelope is now our desired recovered signal m(t). We call this process "envelope demodulation". It is very inexpensive and highly reliable. The radio transmitter has to generate much more power than it would for DSBSC, but millions of listeners to programming from that or other transmitters benefit by very low cost of their receivers. The system as a whole is, therefore, highly cost-effective.

As you can see from the m=1 or m=0.75 cases in Fig. 7 this will work fine if m<1. If m>1, however, (that is, overmodulation), we get distortion (see the m = 1.5 case of Fig. 7). Do not confuse "m" with "m(t)".

1. Basically for much of this experiment we follow Amplitude Modulation, p.47, A1, parts T1 through T14 except that for baseband, m(t), set the audio oscillator at **2** kHz (this will give better separation of the spectral lines on the spectrum analyzer). Use the 100 kHz carrier and again a lower frequency carrier of around 15-17 kHz (from the VCO), as explained below. The setup is much the same as for DSBSC, except that DC is added to the baseband signal. This is exactly equivalent to adding carrier itself, because this DC together with m(t) is used to multiply $\cos(\omega_c t)$. In place of T1-T14 it would be best to follow the summary below, since various changes are made.

(**a**) Figures 2 and 6 provide block diagrams for your use. Figures 1, 3 and 5 show waveforms of the type you should be seeing, except that when the 18 kHz carrier is used the black "blur" becomes visible as a high frequency under the envelope of m(t).

(b) Set the DC voltage to 1 volt (voltmeter not necessary, the scope is sufficiently accurate), and set the audio oscillator voltage (after going through a buffer amplifier) to 2 v peak to peak. (A DC voltage of -1 volt works just as well as 1 volt; the modulated waveforms will look exactly the same). Multiply by the 100 kHz signal as per Fig. 6. (Multiplier set to accept DC). The output of the multiplier should look much like Fig. 7, m = 1. Record a two channel display as in Fig. 1.

(c) Now connect the VCO output ("LO" setting, and f_o as high as possible without triggering the overload LED) to a buffer amplifier, and replace the 100 kHz carrier with this VCO output. Adjust amplitude until the oscilloscope display again show an "m=1" condition. Record a two channel display as in (b), and record a spectral analysis displayed on the 39 kHz setting. You should get a display something like Example 1, below, but not necessarily with the same amplitudes and spacings. (Don't worry about the very low amplitude peaks at higher frequencies in the spectrum; some are due to harmonic distortion in the modules and others may be sampling artifacts.)

The sidebands should be spaced at 2 kHz to the right and left of the carrier, and they should be 6 dB below the carrier in amplitude. Vary the frequency of the oscillator; you should see the sidebands move toward the carrier at lower frequency and away from it for higher frequency.

(d) Substitute voice for the local oscillator and adjust to obtain approximately 2 v p-p on maximum excursions. Record a sample of the spectrum when the sideband levels are reasonably prominent (be sure to use the "average" setting under Tools-Options). See "Example 2" below. Temporarily remove DC; again you will have a DSBSC signal (Example 2 without the central peak.)

(e) Connect the AM signal (DC back in place) to the "rectifier" on the Utilities module, and connect the output of the rectifier to the input of the 60 kHz LPF. This should give you a clean replica of m(t) with the high frequency removed (provided the LPF cutoff is below your carrier frequency. Alternatively you can go back to a 100 kHz carrier, then the LPF cutoff frequency is guaranteed to be low enough.) Compare the two on Picoscope; they should be essentially the same.

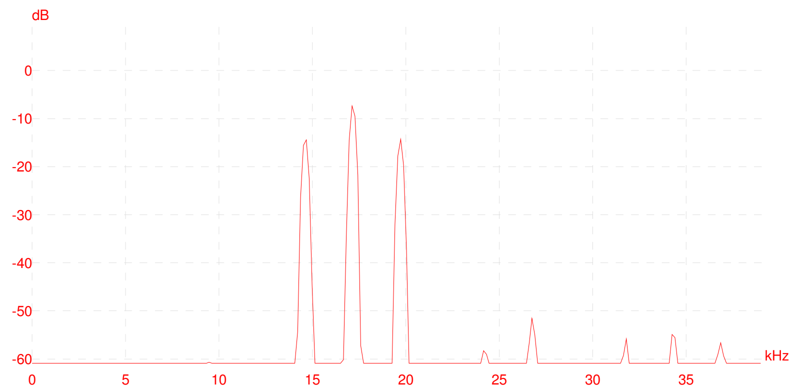
(f) While using the audio oscillator for m(t), raise the gain of the buffer amplifier and watch the spectrum. You will start seeing extra sidebands; these will be due to the distortion resulting from the fact that the upper (or lower) half of the AM envelope (see for example m = 1.5 in Fig. 7) no longer contains a faithful replica of the audio signal. Record a sample spectrum showing this distortion.

(g) In your report, include appropriate explanatory comments on what you are seeing and learning. Answer Tutorial Questions on p. 59 except for Q2

(h) The following statement must be entered at the end of your report.

<u>Certification:</u> I hereby certify that performance of the experiment above and the writing of this report was entirely by me (or by me and my lab partner). I understand that if this certification is false, I am in violation of Academic Honesty rules and may be subject to serious penalty in accordance with those rules.

Signature ______________________________________Date_______________



Example 1

Example 2