Double Sideband (DSB) and Amplitude Modulation (AM)

Modules: Audio Oscillator, Wideband True RMS Meter, Multiplier, Adder, Utilities, Phase Shifter, Tuneable LPF, Quadrature Utilities, Noise Generator, Speech, Headphones

0 Pre-Laboratory Reading

Double sideband (DSB) is one of the easiest modulation techniques to understand, so it is a good starting point for the study of modulation. A type of DSB, called binary phase-shift keying, is used for digital telemetry. Amplitude modulation (AM) is similar to DSB but has the advantage of permitting a simpler demodulator, the envelope detector. AM is used for broadcast radio, aviation radio, citizens' band (CB) radio, and short-wave broadcasting.

0.1 Double Sideband (DSB) Modulation

A message signal $x(t)$ can be DSB modulated onto a carrier with a simple multiplication. The modulated carrier $y(t)$ can be represented by

$$y(t) = x(t) \cdot \cos(2\pi f_c t) \tag{1}$$

where f_c is the carrier frequency. The Fourier transform $Y(f)$ of the modulator output is related to the Fourier transform $X(f)$ of the message signal by

$$Y(f) = \frac{1}{2}X(f - f_c) + \frac{1}{2}X(f + f_c) \tag{2}$$

For a real $x(t)$, $|X(f)|$ is symmetric about $f = 0$ and $|X(f - f_c)|$ is symmetric about $f = f_c$. For the present discussion, it is only necessary to consider that part of $Y(f)$ that lies in the positive half of the frequency axis. The signal content that lies in the frequency domain below f_c is the lower sideband. The signal content that lies in the frequency domain above f_c is the upper sideband. This is the origin of the term *double sideband*.

When studying and testing analog modulation schemes, it is convenient to use a sinusoid as the message signal. This is a good choice for several reasons. First, when testing a system in the laboratory, it is desirable to use a periodic signal since a stable oscilloscope display with continuous signal capture is then possible. Second, the mathematics are usually simpler with a sinusoidal message signal. Third, a sinusoid is easy to generate in the laboratory. After a communication system goes into the field and becomes operational, the message signal would not ordinarily be sinusoidal, of course.

If $x(t)$ is a sinusoid of frequency f_m, the modulated carrier can be written as

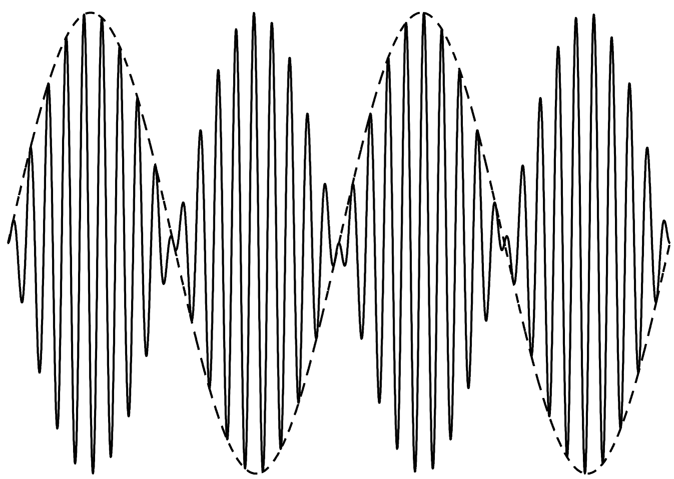
$$y(t) = \cos(2\pi f_m t) \cdot \cos(2\pi f_c t)$$

$$= \frac{1}{2}\cos[2\pi(f_c - f_m)t] + \frac{1}{2}\cos[2\pi(f_c + f_m)t] \quad (3)$$

This last expression indicates that when a carrier is DSB modulated by a message sinusoid, the modulated carrier is equivalent to the sum of two sinusoids: one having the difference frequency $f_c - f_m$ and the other with the sum frequency $f_c + f_m$. In the frequency domain there is signal content lying on both sides of the carrier frequency. When $x(t)$ is a sinusoid, each sideband is one discrete spectral line (on the positive half of the frequency axis).

In the general case where $x(t)$ is real but not a sinusoid, each sideband is more complicated than a single discrete spectral line. However, it is still true that there are two sidebands: a lower and an upper.

In the time domain, a carrier that is DSB modulated by a message sinusoid looks like this:



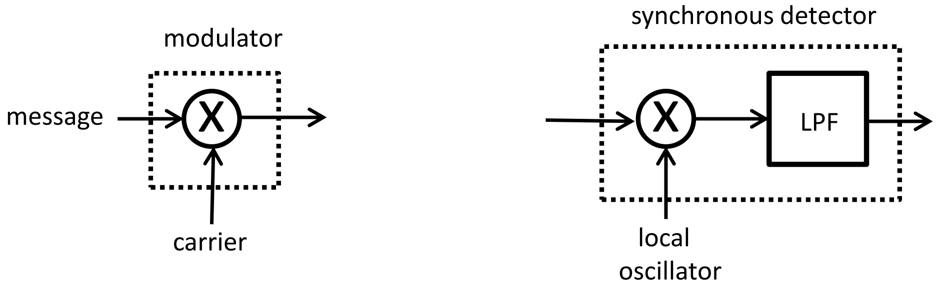
DSB modulated carrier (solid curve) and message sinusoid (dashed curve)

A DSB modulated carrier is normally demodulated with a synchronous detector. This means that the modulated carrier is multiplied by a local oscillator and the product is then sent to a low-pass filter. With synchronous detection the frequency and phase of the local oscillator are important. Its frequency must match that of the carrier. The local oscillator phase must approximately match that of the carrier.

When the modulated carrier $y(t)$ of Eq. (1) is sent to a synchronous detector, the demodulator (detector) output $z(t)$ can be written as

$$\begin{aligned} z(t) &= \mathcal{S}\{y(t)\cdot\cos(2\pi f_c t + \phi)\} \\ &= \mathcal{S}\{x(t)\cdot\cos(2\pi f_c t)\cdot\cos(2\pi f_c t + \phi)\} \\ &= \mathcal{S}\left\{\frac{1}{2}x(t)\cos(\phi) + \frac{1}{2}x(t)\cos(4\pi f_c t + \phi)\right\} \end{aligned} \quad (4)$$

The local oscillator is modeled as $\cos(2\pi f_c t + \phi)$. It has a frequency that matches that of the carrier. However, allowance is made for a phase that might not match that of the carrier. The low-pass filtering is represented by $\mathcal{S}\{\cdot\}$.



DSB modulator (left) and synchronous detector as demodulator (right)

The message signal $x(t)$ is presumably low pass. The term $x(t)\cos(4\pi f_c t + \phi)$ represents a bandpass signal centered at $2f_c$ in the frequency domain. The bandwidth of the low-pass filter should be large enough that $x(t)$ passes through the filter with little distortion. If the bandwidth of $x(t)$ is small compared with $2f_c$ (as it will be in this experiment and as it normally is in practice), then a bandwidth for the low-pass filter can easily be selected that passes $x(t)$ but blocks $x(t)\cos(4\pi f_c t + \phi)$. The demodulator output is, in this case,

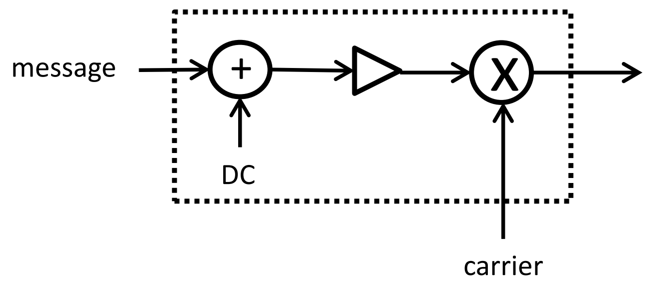
$$z(t) = \frac{K_{\text{LPF}}}{2}\cos(\phi)\, x(t) \tag{5}$$

where K_{LPF} is the lowpass gain of the filter. There will also be a time delay in going through the filter; this is not indicated in Eq. (5) because it is not important for the present discussion. The filter output is a scaled and delayed version of $x(t)$. A demodulation that produces a scaled and delayed version of the original message is considered successful.

An important consideration is the phase difference ϕ between the local oscillator and the carrier. The amplitude of the demodulator output is a function of ϕ. If $\phi = \pm 90^\circ$, $z(t)$ is zero. It is essential that ϕ remain close to 0° so that the message not suffer fading at the output of the demodulator. Keeping ϕ close to 0° is normally accomplished within the receiver by a circuit that performs carrier synchronization.

0.2 Amplitude Modulation (AM)

If a DC component is added to the message signal $x(t)$ before multiplication with a carrier, then the modulation scheme is known as amplitude modulation (AM). The purpose of the DC component is to permit the modulated carrier to be demodulated at the receiver by a means other than synchronous detection. More will be said about this later.



Modulator for AM

The output of the modulator can be written:

$$[c_1 + c_2 x(t)]\cos(2\pi f_c t)$$

Where c_1 and c_2 are constants.

For AM the maximum of the absolute value of the message signal $x(t)$ is of importance; this is denoted here by p ($p \geq 0$).

$$p = \max |x(t)| \tag{6}$$

It is useful to define a normalized message signal $x_n(t)$ as follows:

$$x_n(t) = x(t)/p \tag{7}$$

The maximum of the absolute value of $x_n(t)$ is therefore

$$\max |x_n(t)| = 1 \tag{8}$$

This is what is meant by normalization. The modulator output can be rewritten as

$$[c_1 + c_2 p x_n(t)]\cos(2\pi f_c t)$$

Factoring c_1 out of the bracket and defining

$$\mu = c_2 p / c_1 \tag{9}$$

as the modulation index for AM, the modulator output can then be written

$$c_1[1 + \mu x_n(t)]\cos(2\pi f_c t)$$

The modulation index of Eq. (9) is always non-negative (that is, positive or 0) since p is positive (by definition) and the constants c_1 and c_2 should always have the same sign.

The term $c_1\cos(2\pi f_c t)$ is a residual carrier and the term $c_1\mu x_n(t)\cos(2\pi f_c t)$ is a DSB-modulated carrier. The ratio of the peak value of the DSB term to the peak value of the residual carrier is

$$\left|\frac{c_1\mu}{c_1}\right| = \mu$$

In the above, we recognize that μ is positive. The ratio of the peak value of the DSB term to the peak value of the residual carrier is the modulation index.

There are some AM calculations that involve the ratio of the DSB term to the residual carrier. For this class of problem, it is acceptable to forget about the common factor c_1. The important problem of selecting an appropriate value for the modulation index μ is an example of this class of problem. Therefore, in the analysis that appears below, we sometimes forget about c_1 and use the following definition:

$$y(t) = [1 + \mu x_n(t)]\cos(2\pi f_c t) \tag{10}$$

There are, of course, problems for which the constant c_1 must be considered. One problem of this type is when we must determine if any stage in the transmitter is overloaded (the voltage is so large that the stage is unintentionally operated in a nonlinear regime).

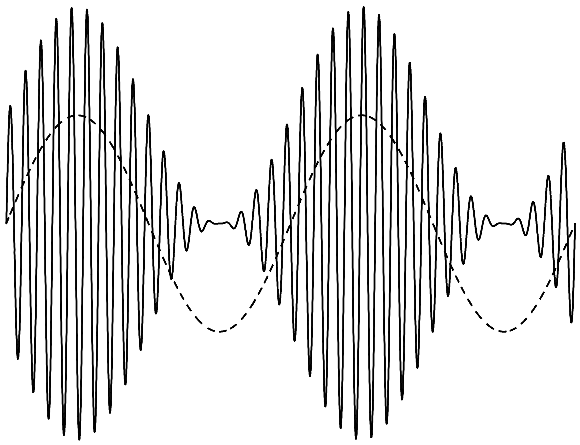
As was done with DSB, the next step in this description of AM is to consider a sinusoidal message signal.

$$x_n(t) = \cos(2\pi f_m t) \tag{11}$$

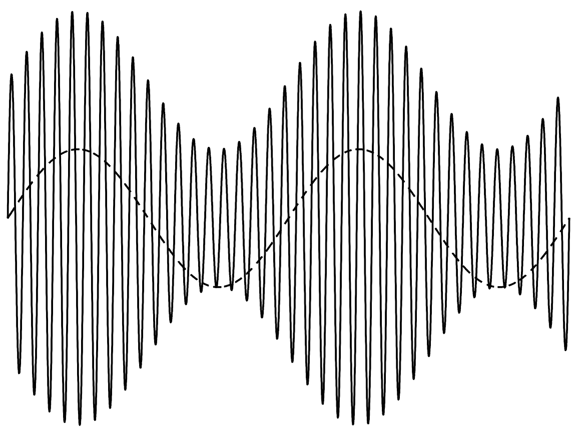
Eq. (11) satisfies Eq. (8). Three cases are of interest:

$\mu = 1$	100% modulation
$\mu < 1$	undermodulation
$\mu > 1$	overmodulation

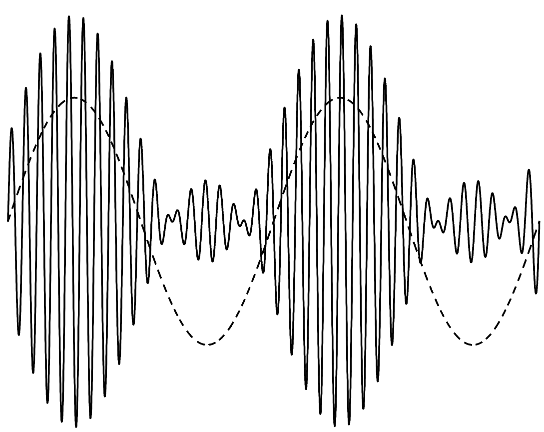
The time-domain view of the modulated carrier is shown below for these three cases. It helps when interpreting these figures to consider the quantity $1 + \mu x_n(t)$. This quantity has a minimum value of 0 for $\mu = 1$. The minimum value is positive and non-zero for $\mu < 1$. The minimum value is negative for $\mu > 1$. (It is assumed here that $\min x(t) = -p$ and therefore that $\min x_n(t) = -1$.)



$\mu = 1$ (100% modulation): AM carrier (solid curve) and message sinusoid (dashed curve)



$\mu < 1$ (undermodulation): AM carrier (solid curve) and message sinusoid (dashed curve)



$\mu > 1$ (overmodulation): AM carrier (solid curve) and message sinusoid (dashed curve)

In the laboratory, the modulation index can be set with good accuracy when the message signal is a sinusoid. The rms value of $c_1 \mu x_n(t)$ and the rms value of c_1 are separately measured. For the $x_n(t)$ of Eq. (11), these rms values are $|c_1|\,\mu/\sqrt{2}$ and $|c_1|$, respectively. The ratio of these two rms values is:

$$\text{ratio of rms message component to rms of DC component} = \mu/\sqrt{2} \;\; \text{(sinusoid)} \qquad (12)$$

A desired modulation index μ can be set by adjusting the gain on the (sinusoidal) message signal until the ratio equals $\mu/\sqrt{2}$. It must be emphasized that Eq. (12) is valid only for a sinusoidal message signal.

There is a technique for estimating the modulation index of an AM carrier that is quite general; it does not require the message signal to be a sinusoid. This technique uses the XY view of an oscilloscope. The multiplier input that is not the carrier is applied to Channel A. In the case of DSB, this is $x(t)$. In the case of AM, this is $1 + \mu x_n(t)$. To Channel B is applied the output of the multiplier (that is, the modulated carrier). When this is done, one of the following figures is displayed:

DSB | AM: $\mu = 1$ 100% modulation | AM: $\mu < 1$ undermodulation | AM: $\mu > 1$ overmodulation

Consider first the DSB figure. The modulated carrier $x(t)\cos(2\pi f_c t)$ is plotted against $x(t)$. At any instant when the message signal is 0, the modulated carrier is also zero. The center of the "bowtie" represents this point; it is the origin. When $x(t)$ takes on the value x', the modulated carrier can take on any value between $-|x'|$ and $|x'|$. The larger $|x'|$, the wider the range of values that the modulated carrier can assume.

For AM, the modulated carrier $[1 + \mu x_n(t)]\cos(2\pi f_c t)$ is plotted against $1 + \mu x_n(t)$. If $\mu = 1$, then $1 + \mu x_n(t)$ takes on only non-negative values, so only the right half of the "bowtie" is present. If $\mu < 1$, then the minimum value of $1 + \mu x_n(t)$ is a positive and non-zero value. If $\mu > 1$, then the minimum value of $1 + \mu x_n(t)$ is negative.

The nice thing about using the XY view to estimate the modulation index is that this works for any message signal, even an audio signal (from a microphone, for example). The problem of calculating the modulation index for a general audio signal is that it can be difficult even to identify p and therefore $x_n(t)$ for a signal that appears random. The XY view provides a quick but approximate characterization of the AM.

It is worthwhile to consider how an AM carrier looks in the frequency domain. This is done here for the case of a sinusoidal message signal. Eqs. (10) and (11) together can be rewritten as

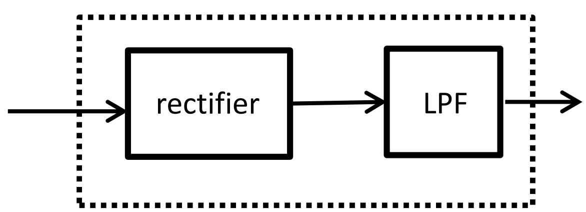
$$\begin{aligned} y(t) &= \cos(2\pi f_c t) + \mu\cos(2\pi f_m t)\cos(2\pi f_c t) \\ &= \cos(2\pi f_c t) + \frac{\mu}{2}\cos[2\pi(f_c - f_m)t] \\ &\quad + \frac{\mu}{2}\cos[2\pi(f_c + f_m)t] \end{aligned} \tag{13}$$

It is clear from Eq. (13) that the spectrum of an AM carrier with sinusoidal message signal contains three discrete spectral lines (on the positive half of the frequency axis). The frequencies are: f_c, $f_c - f_m$, and $f_c + f_m$. This is similar to the case of DSB, except for the additional discrete spectral line at f_c, which is known as a *residual carrier*. Each of the sideband lines has a line height that is different from that of the residual carrier by

$$\text{sideband line height relative to residual carrier} = 20\log(\mu/2) \quad \text{dBc} \tag{14}$$

The unit dBc is the number of decibels relative to the residual carrier. For 100% modulation ($\mu = 1$), each sideband is -6 dBc.

Demodulation of an AM carrier is normally done with an envelope detector. An envelope detector, implemented as a rectifier followed by a low-pass filter, extracts the envelope of the modulated carrier. For the modulated carrier of Eq. (10), the envelope is $|1 + \mu x_n(t)|$.



Demodulator for AM

If $\mu \le 1$, $1 + \mu x_n(t)$ is always non-negative, so the envelope is just $1 + \mu x_n(t)$. With AC coupling (to remove the DC component) a scaled (and delayed) version of the original message is recovered.

With overmodulation ($\mu > 1$), the envelope $|1 + \mu x_n(t)|$ does not equal $1 + \mu x_n(t)$. In this case, only a distorted version of the message can be recovered. However, if the overmodulation is slight, then the distortion is slight.

Ideally, AM should be used with μ close to 1. If μ is significantly greater than 1, then distortion occurs in envelope detection at the receiver. If μ is significantly less than 1, then the system is inefficient because an unnecessarily large fraction of the transmitted power is devoted to the residual carrier, which contains no information about the message. It is the sidebands that contain the information about the message; the residual carrier is only present to make distortion-free envelope detection possible at the receiver.

Envelope detection, unlike synchronous detection, requires no carrier synchronization circuit in the receiver. This is why AM was selected over DSB for broadcast radio. The goal was to permit the many receivers of broadcast radio to be simple (and therefore cheap) and reliable.

1 DSB Modulation and Synchronous Detection

You will build a DSB modulator. This will be paired with a synchronous detector.

Initially, you will use a 5-kHz sinusoid as the message signal. Adjust the frequency of the Audio Oscillator to approximately 5 kHz and place this message signal at one of the inputs of the Multiplier. It is necessary that the Multiplier be set for DC coupling. Connect a 100-kHz sinusoid (Master Signals) to the second input of the Multiplier. The output of the Multiplier is the modulated carrier.

Simultaneously observe the message signal on Channel A and the modulated carrier on Channel B of the oscilloscope. Use the TTL output from the Audio Oscillator as an external trigger source. This will stabilize the envelope, but you will see that the fast carrier oscillation inside the envelope is not stabilized. This is a consequence of the fact that the message sinusoid and the 100-kHz sinusoid are not coherently related. (The message sinusoid has a frequency that is only *approximately* 5 kHz; it is not *exactly* 100 kHz divided by 20.)

Channel A: 5-kHz message sinusoid
Channel B: DSB carrier

Switch to an XY View (**`Views > X-Axis > A`**). Verify that this view gives the correct picture for DSB. (In order to obtain the desired XY display, the message signal must be on Channel A and the modulated carrier on Channel B.)

Channel A: 5-kHz message sinusoid
Channel B: DSB carrier

Observe the DSB carrier on the spectrum analyzer. Note the frequencies of the two tall spectral lines.

Channel B: DSB carrier

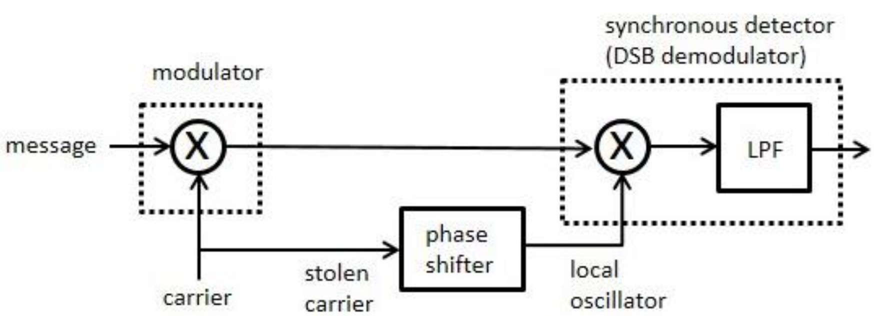
We want to generate a local oscillator for the demodulator that has a variable phase relative to the 100-kHz carrier. Place the 100-kHz sinusoid that you used for the DSB modulator on the input of a Phase Shifter. Using the oscilloscope to view the Phase Shifter input and output, adjust the Phase Shifter for a phase difference between input and output of 0°. (In other words, the output will be in phase with the input.) This phase difference will be changed later.

Now build a synchronous detector. This will serve as the DSB demodulator. (A synchronous detector is a type of demodulator. When discussing DSB, we will use the terms demodulator and detector to mean the same thing.) You will need a multiplier for this demodulator; we will call this the detector's multiplier. You should use one of the multipliers in the Quadrature Utilities module for this purpose. Connect the modulated carrier to one input of the detector's multiplier. Use the output of the Phase Shifter as the local oscillator, so this phase-shifted 100-kHz sinusoid should be connected to the second input of the detector's multiplier.

You will use a Tuneable LPF in the demodulator. Before placing it in the demodulator, adjust its bandwidth to approximately 6 kHz. (The Tuneable LPF's clock output has a frequency equal to 100 times the bandwidth.) Use the Noise Generator module to get a quick display of $|H(f)|$.

Channel A: Tuneable LPF output, showing $|H(f)|$

Connect the output of the detector's multiplier to the Tuneable LPF.



Simultaneously observe the original 5-kHz message sinusoid (coming out of the Audio Oscillator) and the Tuneable LPF output on the oscilloscope. Use the TTL output from the Audio Oscillator as an external trigger source. It should be possible to stabilize simultaneously both sinusoids in the display since they share a common frequency. You may want to increase the gain of the Tuneable LPF in order to get a nice, strong signal out of the detector.

Channel A: 5-kHz message sinusoid from Audio Oscillator
Channel B: demodulator output (0° phase offset)

Notice that the detector has reproduced the original message, with some scaling and a delay. (Of course there is delay, there is always delay in passing through a filter.) In judging whether a demodulation is successful, we ask whether the shape of the message signal is reproduced on the demodulator's output. The amplitude of the demodulator's output is not important here (as long as it is not *too* small); in a practical receiver, the demodulator's output can always be rescaled with an amplifier.

You will now record the amplitude of the detector's output for a 0° phase offset of the 100-kHz local oscillator from the 100-kHz sinewave that is used in the modulator.

DSB

Now adjust the Phase Shifter so that the output lags the input by 90°. To do this, you will simultaneously observe both the input and the output of the Phase Shifter on the oscilloscope, adjusting the phase of the Phase Shifter until the phase difference is 90°. The XY view is helpful for this. (When the phase offset is 90°, the ellipse axes will be horizontal and vertical.) You should note, however, that the XY view does not tell you which sinusoid is leading and which is lagging.

You should *not* change the gain of the Tuneable LPF before making this measurement. We want the LPF gain to be the same for the two different phase offsets: 0° and 90°.

Simultaneously observe the original 5-kHz message sinusoid (coming out of the Audio Oscillator) and the Tuneable LPF output on the oscilloscope. Use the TTL output from the Audio Oscillator as an external trigger source.

Channel A: 5-kHz message sinusoid from Audio Oscillator
Channel B: demodulator output (90° phase offset)

Now with a 90° phase offset between local oscillator and carrier, record the amplitude of the detector's output.

As you can see, synchronous detection requires not only that the local oscillator match the carrier in frequency but also (at least approximately) in phase. This is an important issue in receiver design. For the purpose of this laboratory, the receiver has "stolen" a copy of the (unmodulated) carrier from the transmitter. This is feasible as long as both transmitter and receiver are sitting in the same laboratory. In the field, it is generally impossible for the receiver to steal a copy of the (unmodulated) carrier from the transmitter. Instead, the receiver must perform carrier synchronization.

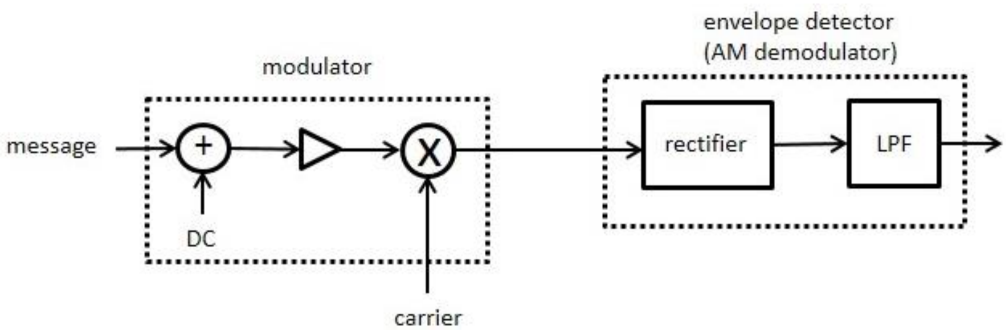
2 AM and Envelope Detection

You will build an AM modulator. This will be paired with an envelope detector.

2.1 Sinusoidal Message

Build a modulator for AM. Use a 5-kHz sinusoid from the Audio Oscillator as the message signal. Use a weighted adder to combine the message signal and a DC component. The DC component will come from the Variable DC panel, and the knob on this panel should be set clockwise from the vertical position, so that a positive DC value is supplied. Connect the output of the Adder to a Buffer Amplifier and the output of this amplifier to a multiplier (set for DC coupling). The other input to the multiplier will be a 100-kHz sinusoid (Master Signals).

In the Adder, the gain on each input is negative. The purpose of the Buffer Amplifier, with its negative gain, is to cancel minus signs so that the net gain through Adder and amplifier is positive for both the DC component and the message signal. Care should be taken that the absolute value of the gain is not set too high. If it is too high, saturation (overloading) can result.



Set the modulation index μ to 1. According to Equation (12), a modulation index of 1 is achieved by having the rms voltage of the message component equal to 0.707 times the rms voltage of the DC component. You will measure rms voltages by connecting the Buffer Amplifier output to the Wideband True RMS Meter. You will adjust the rms voltages using the following procedure. First, disconnect the message signal from the Adder, so that only the DC component is present. Second, adjust the gain applied by the Adder to the DC component until the output of the Buffer Amplifier is 1.00 V rms. Third, reconnect the message signal to the Adder input and disconnect the DC component from the Adder, so that only the message component is present. Fourth, adjust the gain applied by the Adder to the message component until the output of the Buffer Amplifier is 0.707 V rms. Fifth, reconnect the DC component to the Adder. The ratio of the rms voltage of the message component to that of the DC component should now be approximately $1/\sqrt{2}$, so that $\mu = 1$.

Connect the 5-kHz message sinusoid to Channel A and the AM modulator output to Channel B. Use the TTL output from the Audio Oscillator as the trigger source. This will stabilize the display of the envelope but not the fast carrier fluctuations inside the envelope.

Channel A: 5-kHz message sinusoid
Channel B: AM carrier ($\mu = 1$)

Place the Buffer Amplifier output on Channel A and the AM modulator output on Channel B. Switch to an XY view and observe the resulting figure. You should see the characteristic shape for 100% modulation ($\mu = 1$).

Channel A: Buffer Amplifier output
Channel B: AM carrier ($\mu = 1$)

Observe the spectrum of the modulated carrier. You should see three strong spectral lines: the residual carrier at 100 kHz, a lower sideband, and an upper sideband. The lower and upper sidebands should have approximately the same level. You should use a logarithmic vertical scale, having units of dBu. Make a note of the upper sideband line height in dBu and the residual-carrier line height in dBu. Subtract the residual-carrier line height (dBu) from the upper sideband line height (dBu). The difference is the upper sideband line height in dBc (decibels relative to the carrier). This should be a negative number, since in this case the upper sideband has a smaller level than the residual carrier.

Channel B: AM carrier ($\mu = 1$)

In addition to recording the measured upper sideband line height in dBc, also record the line height as predicted by Eq. (14).

AM sinusoidal message

You will use a Tuneable LPF in the AM demodulator. Before placing it in the demodulator, adjust its bandwidth to approximately 6 kHz. (If you use the same Tuneable LPF that you used for DSB demodulation, it already has a bandwidth of 6 kHz.)

You will now build an envelope detector. This will serve as the demodulator for the AM signal. Connect the modulated carrier to a rectifier (on the Utilities module) and the rectifier output to a Tuneable LPF (with bandwidth 6 kHz). The combination of the rectifier and the low-pass filter makes an envelope detector.

Connect the 5-kHz message sinusoid to Channel A and the rectifier output to Channel B. Use the TTL output from the Audio Oscillator as the trigger source.

Channel A: 5-kHz message sinusoid
Channel B: rectifier output ($\mu = 1$)

On Channel B replace the rectifier output with the envelope detector output (that is, the detector's Tuneable LPF output). For the envelope detector output, set the coupling for AC. (Now that the envelope detection is done, the DC component is no longer needed.)

Channel A: 5-kHz message sinusoid
Channel B: envelope detector output ($\mu = 1$)

You will have noticed that no local oscillator (and therefore no stolen carrier) is required for envelope detection. For an AM receiver, carrier synchronization is not required; and this simplifies the receiver design.

Set the modulation index to the new value of 0.5. You will use the same procedure that you earlier used for setting the modulation index to 1, except that now the ratio of the rms voltage of the message component to that of the DC component will be 0.35, in accord with Equation (12). The table below summarizes one way (but not the only way) of achieving this ratio.

μ	message component (V rms) measured at Buffer Amplifier output	DC component (V rms) measured at Buffer Amplifier output	ratio
1.0	0.71	1.00	0.71
0.5	0.35	1.00	0.35
1.5	1.06	1.00	1.06

With $\mu = 0.5$, observe the AM carrier on the oscilloscope. Use the TTL output from the Audio Oscillator as the trigger source.

Channel A: 5-kHz message sinusoid
Channel B: AM carrier ($\mu = 0.5$)

Place the Buffer Amplifier output on Channel A and the AM modulator output on Channel B. Switch to an XY view and observe the resulting figure. This should look like undermodulation, $\mu < 1$.

Channel A: Buffer Amplifier output
Channel B: AM carrier ($\mu = 0.5$)

Observe the spectrum of the modulated carrier and record the line height of the upper sideband relative to that of the residual carrier, characterizing the difference in decibels as some number of dBc. Make sure that you get the algebraic sign correct. (If the upper sideband is smaller than the residual carrier, then the upper sideband line height is a negative number of dBc.)

Channel B: AM carrier ($\mu = 0.5$)

Place the 5-kHz message sinusoid from the Audio Oscillator on Channel A and the envelop detector output on Channel B. For the envelope detector output, set the coupling for AC.

Channel A: 5-kHz message sinusoid
Channel B: envelope detector output ($\mu = 0.5$)

Set the modulation index to the new value of 1.5 with the help of Equation (12). In this case, the 5-kHz message signal will be distorted by the envelope detector. You should increase the bandwidth of the Tuneable LPF (beyond the original bandwidth of 6 kHz) in order to get a clear picture of this distortion.

With $\mu = 1.5$, observe the AM carrier on the oscilloscope. Use the TTL output from the Audio Oscillator as the trigger source.

Channel A: 5-kHz message sinusoid
Channel B: AM carrier ($\mu = 1.5$)

Place the Buffer Amplifier output on Channel A and the AM modulator output on Channel B. Switch to an XY view and observe the resulting figure. This should look like overmodulation, $\mu > 1$.

Channel A: Buffer Amplifier output
Channel B: AM carrier ($\mu = 1.5$)

Observe the spectrum of the modulated carrier and record the line height of the upper sideband relative to that of the residual carrier, characterizing the difference in decibels as some number of dBc. Make sure that you get the algebraic sign correct.

Channel B: AM carrier ($\mu = 1.5$)

Place the 5-kHz message sinusoid from the Audio Oscillator on Channel A and the envelop detector output on Channel B. For the envelope detector output, set the coupling for AC.

Channel A: 5-kHz message sinusoid
Channel B: envelope detector output ($\mu = 1.5$)

2.2 Audio Message

Use an audio signal from the Speech module as the message signal. Otherwise, your configuration will remain the same.

You will use the XY view to adjust the modulation index (instead of using rms voltage measurements). Place the Buffer Amplifier output on Channel A and the AM modulator output on Channel B. Switch to an XY view. You will notice that, given the random nature of the audio message, the XY display changes with each frame. Adjust the relative gains of the weighted adder in the modulator in order that the modulation index appears as approximately 1 (the right triangle is complete, and the left triangle is absent) in those frames where the message signal is at its largest value. (For some frames, the display will look like undermodulation.) We consider this to be an overall modulation index of 1.

Channel A: Buffer Amplifier output
Channel B: AM carrier ($\mu \cong 1$)

Connect the AM modulator output to the envelope detector. Connect the detector output to the headphones. Listen to the recovered audio message.

Repeat the above steps, but setting μ to a value less than 1 (that is, undermodulation). In this case, the display should look like undermodulation in *all* frames.

Repeat the above steps, but setting μ to a value greater than 1 (that is, overmodulation). In this case, the display should look like overmodulation in *all* frames.

Record your (subjective) judgment of audio quality, using adjectives like “good”, “okay” and “poor”.

AM audio message