# EE 400L
# Communications

## Laboratory Exercise #2
## Analog Linear Modulation

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# PREPARATION

# 1- Envelope Recovery

## Envelope definition

When we talk of the envelopes of signals we are concerned with the appearance of signals in the time domain. Text books are full of drawings of modulated signals, and you already have an idea of what the term ‘envelope’ means. It will now be given a more formal definition.

Qualitatively, the envelope of a signal y(t) is that boundary within which the signal is contained, when viewed in the time domain. *It is an imaginary line.*

This boundary has an upper and lower part. You will see these are mirror images of each other. In practice, when speaking of the envelope, it is customary to consider only one of them as ‘the envelope’ (typically the upper boundary).

Although the envelope is imaginary in the sense described above, it is possible to generate, from y(t), a signal e(t), having the same shape as this imaginary line. The circuit which does this is commonly called an *envelope detector*.

For the purposes of this discussion a *narrowband signal* will be defined as one which has a bandwidth very much less than an octave. That is, if it lies within the frequency range $f_1$ to $f_2$ , where $f_1 < f_2$ , then:

$$\log_2 (f_1 / f_2) << 1$$

Another way of expressing this is to say that $f_1 \approx f_2$ so that

$$(f_2 - f_1)/(f_2 + f_1) << 1$$

A *wideband signal* will be defined as one which is very much wider than a narrowband signal !

Every signal has an envelope, although, with wideband signals, it is not always conceptually easy to visualize. To avoid such visualization difficulties the discussion below will assume we are dealing with narrow band signals. But in fact there need be no such restriction on the definition, as will be seen later.

Suppose the spectrum of the signal y(t) is located near $f_o$ Hz, where:

$\omega_o = 2.pi.f_o$ …. 1

We state here, without explanation, that *if y(t) can be written in the for*m:

$y(t) = a(t).cos[\omega_o t + p(t)]$ …. 2

where a(t) and j(t) contain only frequency components much lower than $f_o$ (i.e., at message, or related, frequencies), *then we define* the envelope e(t) of y(t) as the absolute value of a(t).

That is,

envelope $e(t) = | a(t) |$ …. 3

Remember that an AM signal has been defined as:

$y(t) = A.(1 + m.cos\mu t).cos\omega t$ ….4

It is common practice to think of the message as being $m.cos\mu t$. Strictly the message should include the DC component; that is $(1 + m.cos\mu t)$. But the presence of the DC component is often forgotten or ignored.

## Example 1: 100% AM

Consider first the case when y(t) is an AM signal.

From the definitions above we see:

$a(t) = A.(1 + m.cos\mu t)$ …. 5

$p(t) = 0$ …. 6

The requirement that both a(t) and j(t) contain only components at or near the message frequency are met, and so it follows that the envelope must be e(t), where:

$e(t) = | A.(1 + m.cos\mu t) |$ …. 7

For the case $m \leq 1$ the absolute sign has no effect, and so there is a linear relationship between the message and envelope, as desired for AM.

This is clearly shown in Figure 1, which is for 100% AM (m = 1). Both a(t) and its modulus is shown. They are the same.

## Example 2: 150% AM

For the case of 150% AM the envelope is still given by e(t) of eqn. 7, but this time m = 1.5, and the absolute sign does have an effect. Figure 2 shows the case for m = 1.5. As well as the message (upper trace) the absolute value of the message is also plotted (centre trace). Notice how it matches the envelope of the modulated signal (lower trace).

## Example 3: DSBSC

For a final example look at the DSBSC, where a(t) = cosμt. There is no DC component here at all. Figure 3 shows the relevant waveforms.

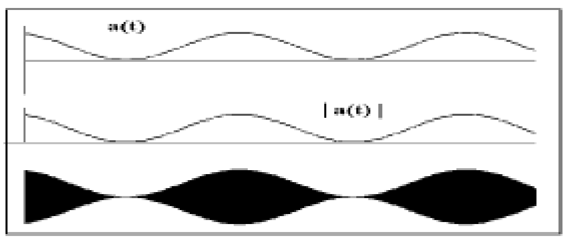


***Figure 1: AM, with m = 1***

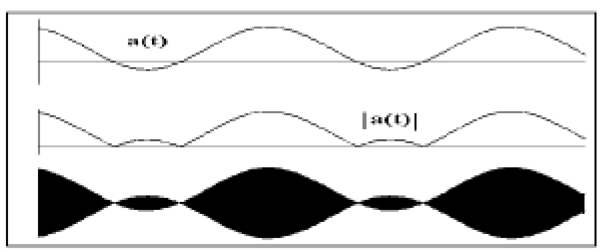


***Figure 2: 150% AM***

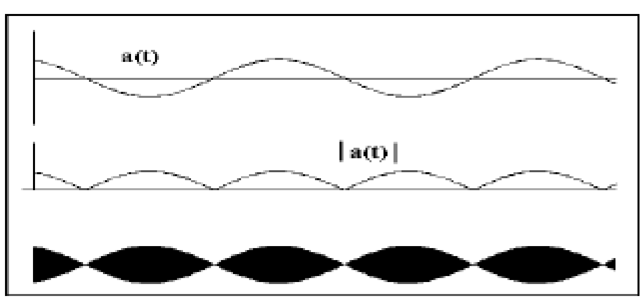

*Figure 3: DSBSC*

## Envelope detector

Although the envelope is imaginary in the sense described above, it is possible to generate, from y(t), a signal e(t), having the same shape as this imaginary line. The circuit which does this is commonly called an *envelope detecto*r. A better word for envelope detector would be *envelope generato*r, since that is what these circuits do.

In this experiment, you will model circuits which will generate these envelope signals.

## Diode detector

The ubiquitous *diode detector* is the prime example of an envelope generator. It is well documented in most textbooks on analog modulation. It is synonymous with the term 'envelope demodulator' in this context.

But remember: the diode detector is an *approximation* to the ideal. We will first examine the ideal circuit.

## Ideal envelope detector

The ideal envelope detector is a circuit which takes the *absolute value* of its input, and then passes the result through a *lowpass filte*r. The output from this lowpass filter is the required envelope signal. See Figure 4.

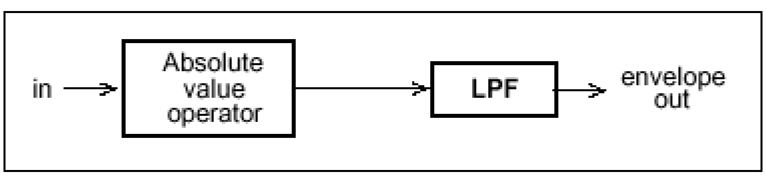


***Figure 4: Ideal envelope recovery arrangement***

The truth of the above statement will be tested for some extreme cases in the work to follow; you can then make your own conclusions as to its veracity.

The absolute value operation, being non-linear, must generate some new frequency components. Among them are those of the wanted envelope. Presumably, since the arrangement *actually work*s, the *unwanted* components lie above those *wanted* components of the envelope.

*It is the purpose of the lowpass filter to separate the wanted from the unwanted components generated by the absolute value operation.*

The analysis of the ideal envelope recovery circuit, for the case of a general input signal, is not a trivial mathematical exercise, the operation being non-linear. So it is not easy to define beforehand where the unwanted components lie.

## Ideal rectifier

A circuit which takes an absolute value is a fullwave rectifier. Note carefully that the operation of rectification is *non-linea*r. The so-called *ideal rectifier* is a precision realization of a rectifier, using an operational amplifier and a diode in a negative feedback arrangement. It is described in text books dealing with the applications of operational amplifiers to analog circuits. An extension of the principle produces an ideal fullwave rectifier.

You will find a halfwave rectifier is generally adequate for use in an envelope recovery circuit.

## Envelope bandwidth

You know what a *lowpass filter* is, but what should be its cut-off frequency in this application? The answer: 'the cut-off frequency of the lowpass filter should be high enough to pass all the wanted frequencies in the envelope, but no more'. So you need to know the envelope bandwidth.

In a particular case you can determine the expression for the envelope from the definition given before, and the bandwidth by Fourier series analysis. Alternatively, you can *estimate* the bandwidth, by inspecting its shape on an oscilloscope, and then applying rules of thumb which give quick approximations.

An envelope will *always* include a constant, or DC, term.

This is inevitable from the definition of an envelope - which includes the operation of taking the absolute value. It is inevitable also in the output of a practical circuit, by the very nature of rectification.

The presence of this DC term is often forgotten. For the case of an AM signal, modulated with music, the DC term is of little interest to the listener. But it is a direct measure of the strength of the carrier term, and so is used as an automatic gain control signal in receivers.

It is *important to note* that it is possible for the bandwidth of the envelope to be much wider than that of the signal of which it is the envelope. In fact, except for the special case of the envelope modulated signal, this is generally so. An obvious example is that of the DSBSC signal derived from a single tone message.

## DSBSC Envelope Bandwidth

The bandwidth of a DSBSC signal is twice that of the highest modulating frequency. So, for a single tone message of 1 kHz, the DSBSC bandwidth is 2 kHz. But the bandwidth of the *envelope* is many times this.

$$\text{DSBSC} = \cos\mu t . \cos\omega t \qquad \ldots 8$$

$$= a(t) . \cos[\omega t + p(t)] \qquad \ldots 9$$

because $\mu << \omega$ then

$$a(t) = \cos\mu t \qquad \ldots 10$$

$$p(t) = 0 \qquad \ldots 11$$

and envelope $e(t) = | a(t) |$ (by definition) … 12

So:

- from the mathematical definition the envelope shape is that of the absolute value of $\cos\mu t$. This has the shape of a fullwave rectified version of $\cos\mu t$.

- by looking at it, and from considerations of Fourier series analysis, the envelope must have a wide bandwidth, due to the sharp discontinuities in its shape. So the lowpass filter will need to have a bandwidth wide enough to pass at least the first few odd harmonics of the 1 kHz message; say a passband extending to *at least* 10 kHz?

# 2- PRODUCT DEMODULATION

All of the modulated signals you have seen so far may be defined as narrow band. They carry message information. Since they have the capability of being based on a radio frequency carrier (suppressed or otherwise) they are suitable for radiation to a remote location. Upon receipt, the object is to recover –*demodulate* - the message from which they were derived.

In the discussion to follow the explanations will be based on narrow band signals. But the findings are in no way restricted to narrow band signals; they happen to be more convenient for purposes of illustration.

## Frequency translation

When a narrow band signal y(t) is multiplied with a sine wave, *two* new signals are created - on the 'sum and difference' frequencies.

Figure 5 illustrates the action for a signal y(t), based on a carrier $f_c$, and a sinusoidal oscillator on frequency $f_o$.

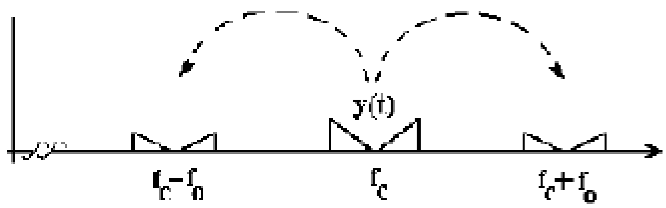


***Figure 5: Sum and difference frequencies***

Each of the components of y(t) was moved *up* an amount $f_o$ in frequency, and *down* by the same amount, and appear at the output of the multiplier.

Remember, *neither* y(t), *nor* the oscillator signal, appears at the multiplier output. This is not necessarily the case with a 'modulator'.

A filter can be used to select the new components at *either* the sum frequency (BPF preferred, or an HPF) *or* difference frequency (LPF preferred, or a BPF).

*the combination of MULTIPLIER, OSCILLATOR,*
*and FILTER is called a* ***frequency translator****.*

When the frequency translation is down to baseband the frequency translator becomes a demodulator.

## Interpretation

The method used for illustrating the process of frequency translation is just that illustrative. You should check out, using simple trigonometry, the truth of the special cases discussed below. Note that this is an amplitude versus frequency diagram; phase information is generally not shown, although annotations, or a separate diagram, can be added if this is important.

Individual spectral components are shown by directed lines (phasors), or groups of these (sidebands) as triangles. The magnitude of the slope of the triangle generally carries no meaning, but the direction does - the slope is down towards the carrier to which these are related.

When the trigonometrical analysis gives rise to negative frequency components, these are re-written as positive, and a polarity adjustment made if necessary. Thus:

$$V.\sin(-\omega t) = -V.\sin(\omega t) \quad \ldots 13$$

Amplitudes are usually shown as positive, although if important to emphasize a phase reversal, phasors can point down, or triangles can be drawn *under* the horizontal axis.

To interpret a translation result graphically, first draw the signal to be translated on the frequency/amplitude diagram in its position before translation. Then *slide* it (the graphic which represents the signal) both to the left and right by an amount $f_o$, the frequency of the signal with which it is multiplied.

If the left movement causes the graphic to cross the zero-frequency axis into the negative region, then locate this negative frequency (say $-f_x$) and place the graphic there. Since negative frequencies are not recognized in this context, the graphic is then *reflected* into the positive frequency region at $+f_x$. Note that this places components in the triangle, which were previously above others, now below them. That is, it reverses their relative positions with respect to frequency.

## Special case: fo = fc

In this case the *down* translated components straddle the origin. Those which fall in the negative frequency region are then reflected into the positive region, as explained above. They will *overlap* components already there. The resultant amplitude will depend upon relative phase; both reinforcement and cancellation are possible.

If the original signal was a DSBSC, then it is the components from the LSB which are reflected back onto those from the USB. Their relative phases are determined by the phase between the original DSBSC (on $f_c$) and the local carrier ($f_o$).

Remember that the contributions to the output by the USB and LSB are combined *linearl*y. They will both be *erec*t, and each would be perfectly intelligible if present alone. Added in-phase, or *coherentl*y, they reinforce each other, to give *twice* the amplitude of one alone, and so *four* times the power.

In this experiment the product demodulator is examined, which is based on the arrangement illustrated in Figure 6. This demodulator is capable of demodulating SSB, DSBSC, and AM. It can be used in two modes, namely synchronous and asynchronous.

## Synchronous demodulator: $\omega_0 = \omega_1$

For successful demodulation of DSBSC and AM the synchronous demodulator requires a ‘local carrier’ of exactly the same frequency as the carrier from which the modulated signal was derived, and of fixed relative phase, which can then be adjusted, as required, by the phase changer shown.

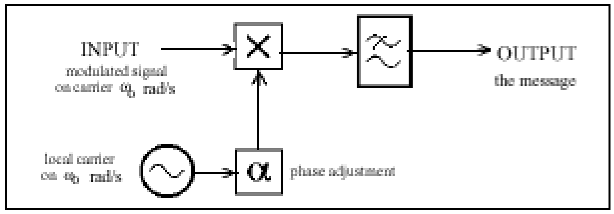


*Figure 6: Synchronous demodulator;* $\omega 1 = \omega 0$

## Carrier acquisition

In practice this local carrier must be derived from the modulated signal itself. There are different means of doing this, depending upon which of the modulated signals is being received. So as not to complicate the study of the synchronous demodulator, it will be assumed that the carrier has already been acquired. It will be 'stolen' from the same source as was used at the generator; namely, the TIMS 100 kHz clock available from the MASTER SIGNALS module.

This is known as the *stolen carrier* technique.

## Asynchronous demodulator: $\omega 0$ =/= $\omega 1$

For asynchronous operation - acceptable for SSB - a local carrier is still required, but it need not be synchronized to the same frequency as was used at the transmitter. Thus there is no need for carrier acquisition circuitry. A local signal can be generated, and held as close to the desired frequency as circumstances require and costs permit. Just how close is 'close enough' will be determined during this experiment.

## Local asynchronous carrier

For the carrier source you will use a VCO module in place of the stolen carrier from the MASTER SIGNALS module. There will be no need for the PHASE SHIFTER. It can be left in circuit if found convenient; its influence will go unnoticed.

## Demodulation of DSBSC

With DSBSC as the input to a synchronous demodulator, there will be a message at the output of the 3 kHz LPF, visible on the oscilloscope.

The magnitude of the message will be dependent upon the adjustment of the PHASE SHIFTER. Whilst watching the message on the oscilloscope, make a phase adjustment with the front panel control of the PHASE SHIFTER, and note that:

a) the message amplitude changes. It may be both maximized AND minimized.

b) the phase of the message will not change; but how can this be observed ? If you have generated your own DSBSC then you have a copy of the message, and have synchronized the oscilloscope to it.

The process of DSBSC demodulation can be examined graphically using the technique described earlier.

The upper sideband is shifted down in frequency to just above the zero frequency origin.

The lower sideband is shifted down in frequency to just below the zero frequency origin. It is then reflected about the origin, and it will lie coincident with the contribution from the upper sideband.

These contributions should be identical with respect to amplitude and frequency, since they came from a matching pair of sidebands.

Now you can see what the phase adjustment will do. The relative phase of these two contributions can be adjusted until they reinforce to give a maximum amplitude. A further 180 degrees shift would result in complete cancellation.

## Demodulation of SSB

With SSB as the input to a synchronous demodulator, there will be a message at the output of the 3 kHz LPF, visible on the oscilloscope.

Whilst watching the message on the oscilloscope, make a phase adjustment with the front panel control of the PHASE SHIFTER, and note that:

a) the message amplitude does NOT change.

b) the phase of the message will change; but how can this be observed ? If you have generated your own SSB then you have a copy of the message, and have synchronized the oscilloscope to it.

Using the graphical interpretation, as was done for the case of the DSBSC, you can see why the phase adjustment will have no effect upon the output amplitude.

*Two identical contributions are needed for a phase cancellation, but there is only one available.*

# EXPERIMENT

## 1- ENVELOPE RECOVERY

### Ideal model

The TIMS model of the ideal envelope detector is shown in block diagram form in Figure 7.

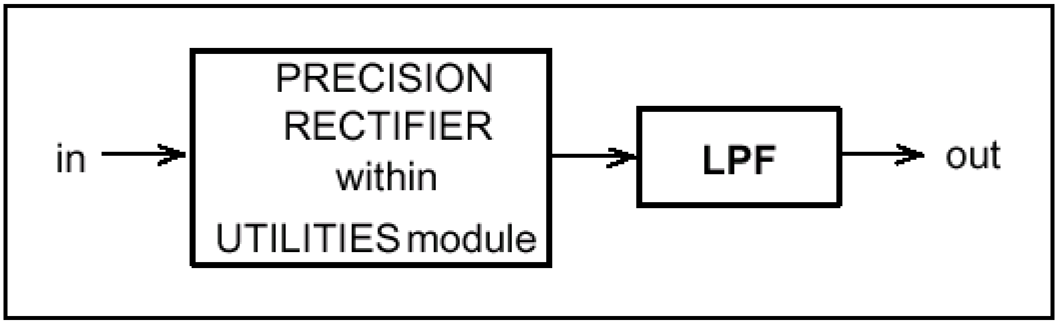


***Figure 7: Modeling the ideal envelope detector with TIMS***

The 'ideal rectifier' is easy to build, does in fact approach the ideal for our purposes, and one is available as the RECTIFIER in the TIMS UTILITIES module. For purposes of comparison, a diode detector, in the form of 'DIODE + LPF', is also available in the same module; this will be examined later.

The desirable characteristics of the lowpass filter will depend upon the frequency components in the envelope of the signal as already discussed. We can easily check the performance of the ideal envelope detector in the laboratory, by testing it on a variety of signals.

The actual envelope shape of each signal can be displayed by observing the modulated signal itself with the oscilloscope, suitably triggered.

The output of the envelope detector can be displayed, for comparison, on the other channel.

## AM envelope

For this part of the experiment we will use the generator of Figure 8, and connect its output to the envelope detector of Figure 7.

***T1*** *plug in the TUNEABLE LPF module. Set it to its widest bandwidth, which is about 12 kHz (front panel toggle switch to* WIDE, *and* TUNE *control fully clockwise). Adjust its passband gain to about unity. To do this you can use a test signal from the AUDIO OSCILLATOR, or perhaps the 2 kHz message from the MASTER SIGNALS module.*

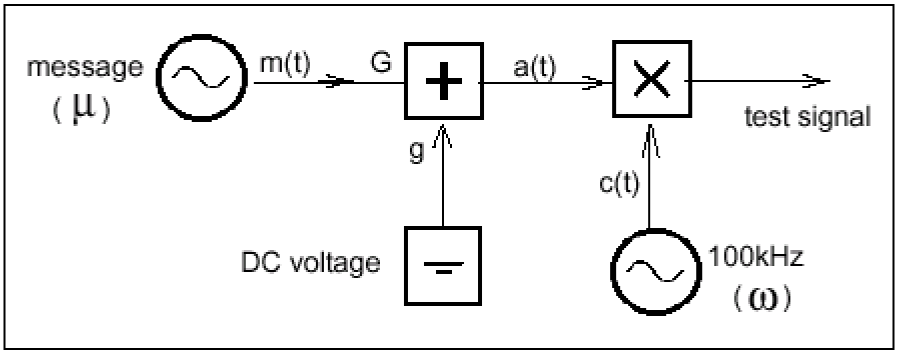


***Figure 8: Generator for AM and DSBSC***

***T2*** *model the generator of Figure8, and connect its output to an ideal envelope detector, modeled as Figure7. For the lowpass filter use the TUNEABLE LPF module. Your whole system might look like that shown modeled in Figure 9 below.*

***T3*** *set the frequency of the AUDIO OSCILLATOR to about 1 kHz. This is your message.*

***T4*** *adjust the triggering and sweep speed of the oscilloscope to display two periods of the message (CH2-A).*

***T5*** *adjust the generator to produce an AM signal, with a depth of modulation less than 100%. Don`t forget to so adjust the ADDER gains that its output (DC + AC) will not overload the MULTIPLIER; that is, keep the MULTIPLIER input within the bounds of the TIMS ANALOG REFERENCE LEVEL (4 volt peak-to-peak). This signal is not symmetrical about zero volts; neither excursion should exceed the 2 volts peak level.*

***T6*** *for the case $m < 1$ observe that the output from the filter (the ideal envelope detector output) is the same shape as the envelope of the AM signal, a sine wave.*

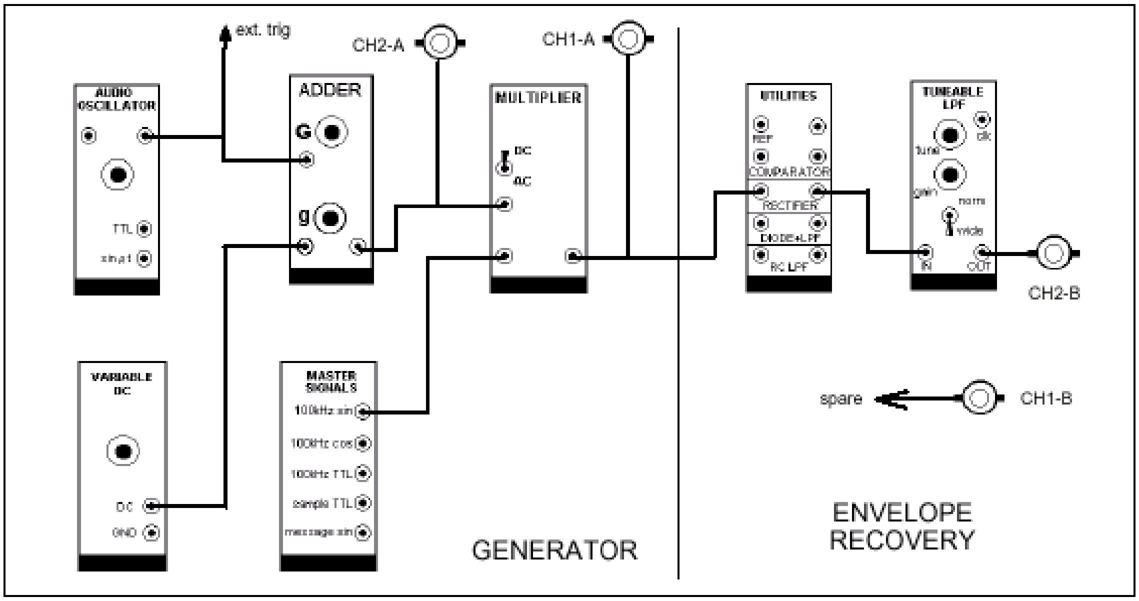


***Figure 9: Modulated signal generator and envelope recovery***

## DSBSC envelope

Now let us test the ideal envelope detector on a more complex envelope - that of a DSBSC signal.

***T7*** *remove the carrier from the AM signal, by turning* ***'g'*** *fully anti-clockwise, thus generating DSBSC. Alternatively, and to save the DC level just used, pull out the patch cord from the 'g' input of the ADDER (or switch the MULTIPLIER to AC).*

Were you expecting to see the waveforms of Figure 10? What did you see?

You may not have seen the expected waveform. Why not?

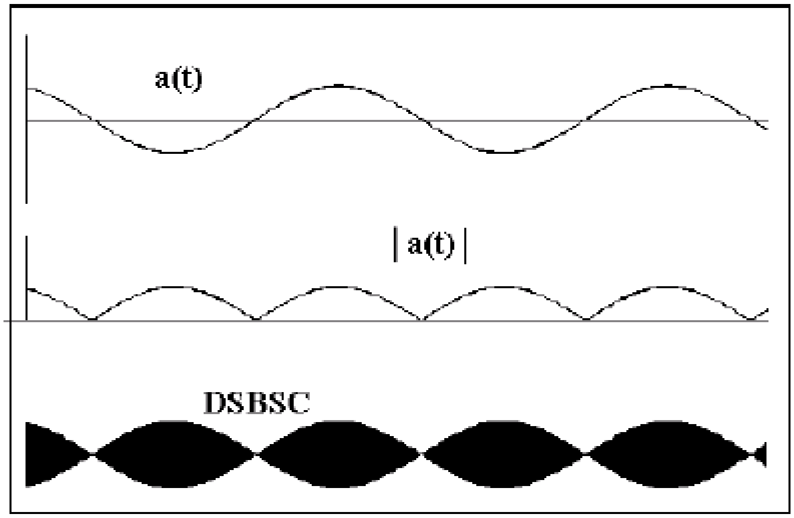


***Figure 10: DSBSC signal***

***T8*** *(a) lower the frequency of the AUDIO OSCILLATOR, and watch the shape of the recovered envelope. When you think it is a better approximation to expectations, note the message frequency, and the filter bandwidth, and compare with predictions of the bandwidth of a fullwave rectified sinewave.*

*(b) if you want to work with a 2 kHz message then replace the TUNEABLE LPF with a 60 kHz LOWPASS FILTER. Now the detector output should be a good copy of the envelope. Record the highest frequency that gives good envelope with this filter.*

## Diode detector

It is assumed you will have referred to a text book on the subject of the *diode detecto*r. This is an approximation to the ideal rectifier and lowpass filter.

There is a DIODE DETECTOR in the UTILITIES MODULE. The diode has not been linearized by an active feedback circuit, and the lowpass filter is approximated by an RC network. Your textbook should tell you that this is a good engineering compromise in practice, provided:

a) the depth of modulation does not approach 100%

b) the ratio of carrier to message frequency is ‘large’.

You can test these conditions with TIMS. The patching arrangement is simple.

***T9*** *connect an Am signal with m<1 directly to the ANALOG INPUT of the 'DIODE + LPF' in the UTILITIES MODULE, and the envelope (or its approximation) can be examined at the ANALOG OUTPUT. You should not add any additional lowpass filtering, as the true 'diode detector' uses only a single RC network for this purpose, which is already included.*

The extreme cases you could try would include:

a) an AM signal with depth of modulation say 50%, and a message of 500 Hz. What happens when the message frequency is raised? Is $\omega >> \mu$ ?

b) a DSBSC. Here the inequality $\omega >> \mu$ is meaningless. This inequality applies to the case of AM with m < 1. It would be better expressed, in the present instance, as 'the carrier frequency $\omega$ must be very much higher than the highest frequency component expected in the envelope'. This is certainly NOT so here.

***T10*** *repeat the previous Task, but with the RECTIFIER followed by a simple RC filter. This compromise arrangement will show up the shortcomings of the RC filter. There is an independent RC LPF in the UTILITIES MODULE.*

***T11*** *you can examine various combinations of diode, ideal rectifier, RC and other lowpass filters, and lower carrier frequencies (use the VCO). The 60 kHz LPF is a very useful filter for envelope work.*

***T12*** *check by observation: is the RECTIFIER in the UTILITIES MODULE a halfwave or fullwave rectifier?*

# 2- Product Demodulator

## Synchronous demodulation

The demodulator of Figure 6 is easily modeled with TIMS.

The carrier source will be the 100 kHz from the MASTER SIGNALS module. This will be a *stolen carrie*r, phase-locked to, but not necessarily in-phase with, the transmitter carrier. It will need adjustment with a PHASE SHIFTER module.

For the lowpass filter use the HEADPHONE AMPLIFIER. This has an in-built 3 kHz LPF which may be switched in or out. If this module is new to you, read about it in the ***TIMS User Manual***.

A suitable TIMS model of the block diagram of Figure 6 is shown below, in Figure 11.

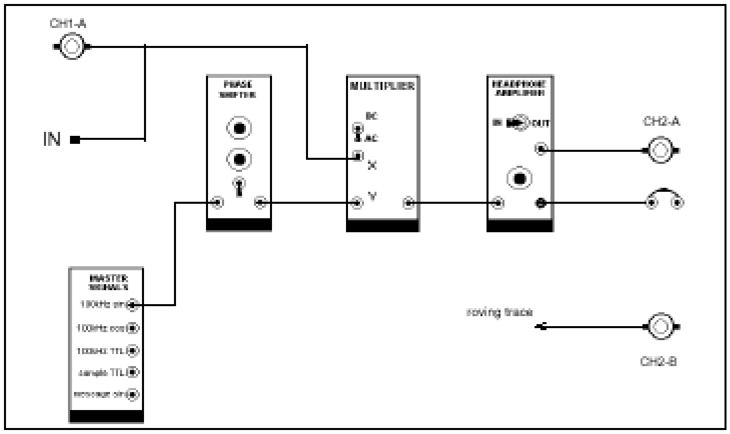


***Figure 11: TIMS model of Figure 6***

***T13*** *patch up the model of Figure 11 above. This shows* $w_0 = w_1$*. Before plugging in the PHASE SHIFTER, set the on-board switch to HI.*

***T14*** *create an Am signal with m=0.5 and connect it to input of the synchronous demodulator. Examine the output of demodulator. Record your observation.*

***T15*** *increase depth of modulation to 100% and 150%. Did this demodulator work for these signals? Remember that Envelope detector cannot use in these cases.*

***T15*** *create a DSB and connect it to input of the synchronous demodulator. Examine the output of demodulator. Record your observation.*

## Asynchronous demodulation

We now examine what happens if the local carrier is off-set from the desired frequency by an adjustable amount df, where:

$$df = |(f_c - f_o)| \qquad \text{... 14}$$

The process can be considered using the graphical approach illustrated earlier.

By monitoring the VCO frequency (the source of the local carrier) with the FREQUENCY COUNTER you will know the magnitude and direction of this offset by subtracting it from the desired 100 kHz.

## VCO fine tuning

Refer to the ***TIMS User Manual*** for details on fine tuning of the VCO. It is quite easy to make small frequency adjustments (fractions of a Hertz) by connecting a small negative DC voltage into the VCO $V_{in}$ input, and tuning with the GAIN control.

## SSB reception

Consider first the demodulation of an SSB signal.

You can show either trigonometrically or graphically that the output of the demodulator filter will be the desired message components, but each displaced in frequency by an amount df from the ideal.

If df is small - say 10 Hz - then you might guess that the speech will be quite intelligible. For larger offsets the frequency shift will eventually be objectionable. The effect upon intelligibility will be dependant upon the direction of the frequency shift, except perhaps when df is less than say 10 Hz.

***T17*** *replace the 100 kHz stolen carrier with the analog output of a VCO, set to operate in the 100 kHz range. Monitor its frequency with the FREQUENCY COUNTER.*

***T18*** *as an optional task you may consider setting up a system of modules to display the magnitude of df directly on the FREQUENCY COUNTER module. But you will find it not as convenient as it might at first appear - can you anticipate what problem might arise before trying it? **(hint**: 1 second is a long time!). A recommended method of showing the small frequency difference between the VCO and the 100 kHz reference is to display each on separate oscilloscope traces - the speed of drift between the two gives an immediate and easily recognized indication of the frequency difference.*

***T19*** *connect an SSB signal to the demodulator input. Tune the VCO slowly around the 100 kHz region. Record output. Report results.*

***Q1*** *use phasors to construct the envelope of (a) an AM signal and (b) a DSBSC signal.*

# TUTORIAL QUESTIONS

***Q1*** *use phasors to construct the envelope of (a) an AM signal and (b) a DSBSC signal.*

***Q2*** *use phasors to construct the envelope of the sum of a DSBSC and a large carrier, when the phase difference between these two is not zero (as it is for AM). The technique should quickly convince you that the envelope is no longer a sine wave, although it may be tedious to obtain an exact shape.*

***Q3*** *explain the major differences in performance between envelope detectors with half and fullwave rectifiers.*

***Q4*** *would you define the synchronous demodulator as an SSB demodulator? Explain.*

***Q5*** *if a 'DSBSC' signal had a small amount of carrier present what effect would this have as observed at the output of a synchronous demodulator?*

***Q6*** *suppose, while you were successfully demodulating a DSBSC based on 100 kHz carrier, a second DSBSC based on a 90 kHz carrier was added to it. Suppose the amplitude of this 'unwanted' DSBSC was much smaller than that of the wanted DSBSC.*

*a) would this new signal at the demodulator INPUT have any effect upon the message from the wanted signal as observed at the demodulator OUTPUT ?*

*b) what if the unwanted DSBSC was of the same amplitude as the wanted DSBSC. Would it then have any effect ?*

*c) what if the unwanted DSBSC was ten times the amplitude of the wanted DSBSC. Would it then have any effect ? Explain !*

***Q7*** *what are the differences, and similarities, between a multiplier and a modulator?*