

ENVELOPE RECOVERY

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ENVELOPE RECOVERY

ACHIEVEMENTS: *The ideal ‘envelope detector’ is defined, and then modelled. It is shown to perform well in all cases examined. The limitations of the ‘diode detector’, an approximation to the ideal, are examined. Introduction to the HEADPHONE AMPLIFIER module.*

PREREQUISITES: *completion of the experiment entitled **Envelopes** in this Volume.*

PREPARATION

the envelope

You have been introduced to the definition of an envelope in the experiment entitled *Envelopes*. There you were reminded that 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.*

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*. A better word for envelope detector would be *envelope generator*, since that is what these circuits do.

It is the purpose of this experiment for you to model circuits which will generate these envelope signals.

the 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.

the 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 filter*. The output from this lowpass filter is the required envelope signal. See Figure 1.

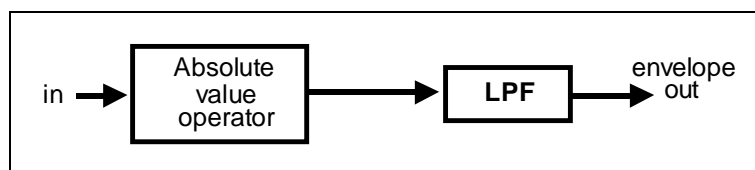


Figure 1: the ideal envelope recovery arrangement

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 works*, 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. See the Appendix to this experiment for the analysis of a special case.

the ideal rectifier

A circuit which takes an absolute value is a fullwave rectifier. Note carefully that the operation of rectification is *non-linear*. 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. Refer to the Appendix to this experiment for details.

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 in the experiment entitled *Envelopes*, 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

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.

For example, we know that, analytically:

$$\text{DSBSC} = \cos\mu t \cdot \cos\omega t \quad \text{..... 1}$$

$$= a(t) \cdot \cos[\omega_0 t + \varphi(t)] \quad \text{..... 2}$$

$$\text{because } \mu \ll \omega \text{ then } a(t) = \cos\mu t \quad \text{..... 3}$$

$$\varphi(t) = 0 \quad \text{..... 4}$$

$$\text{and envelope } e(t) = |a(t)| \quad (\text{by definition}) \quad \text{..... 5}$$

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 ¹, 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 ?

¹ see the section on *Fourier series and bandwidth estimation* in the chapter entitled ***Introduction to modelling with TIMS***, in this Volume

EXPERIMENT

the ideal model

The TIMS model of the ideal envelope detector is shown in block diagram form in Figure 2.

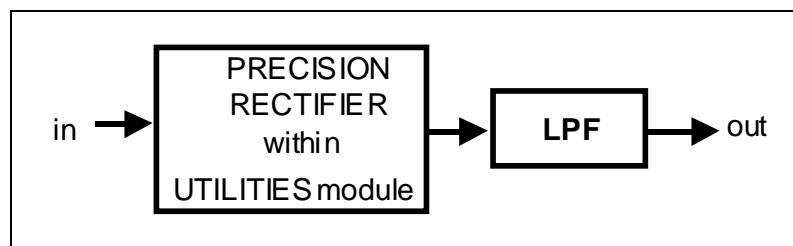


Figure 2: modelling the ideal envelope detector with TIMS

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 3, and connect its output to the envelope detector of Figure 2.

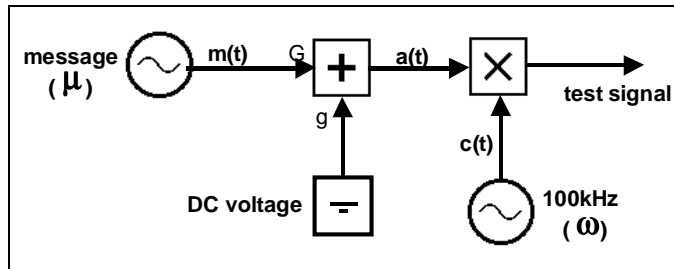


Figure 3: generator for AM and DSBSC

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.

T2 model the generator of Figure 3, and connect its output to an ideal envelope detector, modelled as per Figure 2. For the lowpass filter use the *TUNEABLE LPF* module. Your whole system might look like that shown modelled in Figure 4 below.

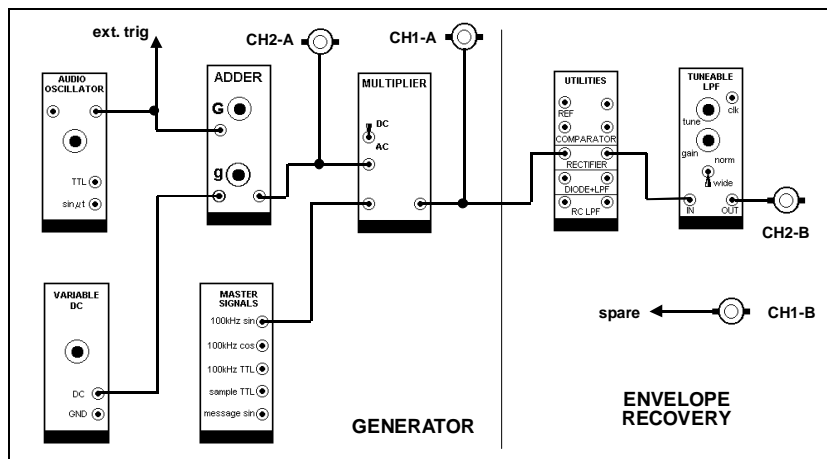


Figure 4: modulated signal generator and envelope recovery

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 TMS ANALOG REFERENCE LEVEL (4 volt peak-to-peak). This signal is not symmetrical about zero volts; neither excursion should exceed the 2 volt 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.

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 5 ? What did you see ?

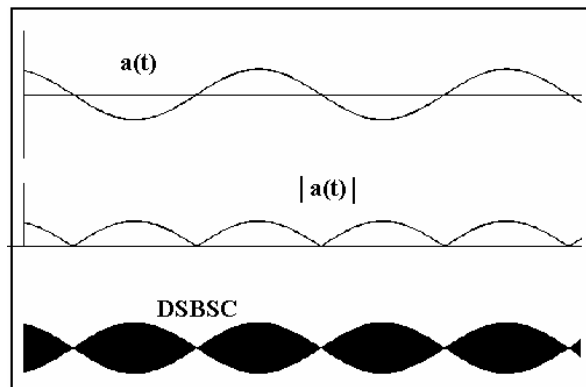


Figure 5: a DSBSC signal

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

With a message frequency of 2 kHz, a filter bandwidth of about 12 kHz is not wide enough.

You can check this assertion; for example:

- a) lower the message frequency, and note that the recovered envelope shape approaches more closely the expected shape.
- b) change the filter. Try a 60 kHz LOWPASS FILTER.

- 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 stay with the 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.*

speech as the message; $m < 1$

Now try an AM signal, with speech from a SPEECH module, as the message.

To listen to the recovered speech, use the HEADPHONE AMPLIFIER.

The HEADPHONE AMPLIFIER enables you to listen to an audio signal connected to its input. This may have come via an external lowpass filter, or via the internal 3 kHz LOWPASS FILTER. The latter is switched in and out by the front panel switch. Refer to the *TIMS User Manual* for more information.

Only for the case of envelope modulation, with the depth of modulation 100% or less, will the speech be intelligible. If you are using a separate lowpass filter, switching in the 3 kHz LPF of the HEADPHONE AMPLIFIER as well should make no difference to the quality of the speech as heard in the HEADPHONES, because the speech at TRUNKS has already been bandlimited to 3 kHz.

speech as the message; $m > 1$

Don't forget to listen to the recovered envelope when the depth of modulation is increased beyond 100%. This will be a distorted version of the speech.

Distortion is usually thought of as having been caused by some circuit imperfection.

*There is **no circuit imperfection** occurring here !*

The envelope shape, for all values of m , including $m > 1$, is as exactly as theory predicts, using ideal circuitry.

The envelope recovery circuit you are using is close to ideal; this may not be obvious when listening to speech, but was confirmed earlier when recovering the wide-band envelope of a DSBSC.

The distortion of the speech arises quite naturally from the fact that there is a non-linear relationship between the message and the envelope, attributed directly to the absolute sign in eqn. (5).

the diode detector

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

How does it perform on these signals and their envelopes ?

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 the signal, whose envelope you wish to recover, 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 $\omega \gg \mu$?
- b) a DSBSC. Here the inequality $\omega \gg \mu$ 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 ω 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. Check the ***TIMS User Manual*** regarding the time constant.

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 ?

TUTORIAL QUESTIONS

- Q1 an analysis of the ideal envelope detector is given in the Appendix to this experiment. What are the conditions for there to be no distortion components in the recovered envelope ?*
- Q2 analyse the performance of a square-law device as an envelope detector, assuming an ideal filter may be used. Are there any distortion components in the recovered envelope ?*
- Q3 explain the major difference differences in performance between envelope detectors with half and fullwave rectifiers.*
- Q4 define what is meant by 'selective fading'. If an amplitude modulated signal is undergoing selective fading, how would this affect the performance of an envelope detector as a demodulator ?*

APPENDIX A

analysis of the ideal detector

The aim of the rectifier is to take the absolute value of the signal being rectified. That is, to multiply it by +1 when it is positive, and -1 when negative.

An analysis of the ideal envelope detector is not a trivial exercise, except in special cases. Such a special case is when the input signal is an envelope modulated signal with $m < 1$.

In this case we can make the following assumption, not proved here, but verified by practical measurement and observations, namely: the zero crossings of an AM signal, for $m < 1$, are uniform, and spaced at half the period of the carrier.

If this is the case, then the action of an ideal rectifier on such a signal is equivalent to multiplying it by a square wave $s(t)$ as per Figure 1A. It is important to ensure that the phases of the AM and $s(t)$ are matched correctly in the analysis; in the practical circuit this is done automatically.

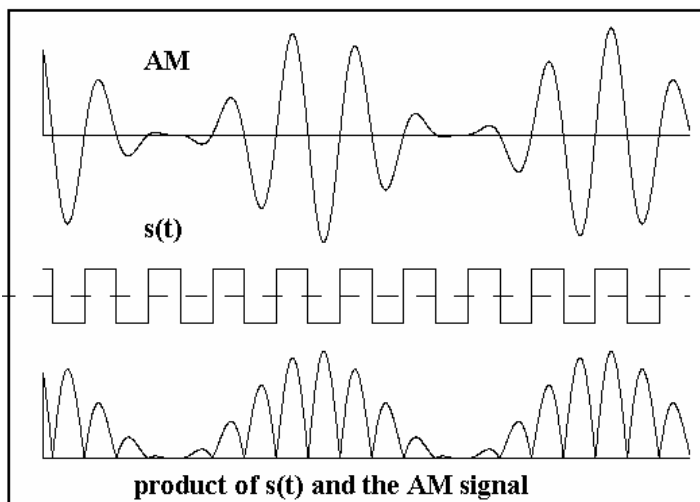


Figure 1A: the function $s(t)$ and its operation upon an AM signal

The Fourier series expansion of $s(t)$, as illustrated, is given by:

$$s(t) = \frac{4}{\pi} [1.\cos\omega t - 1/3.\cos3\omega t + 1/5.\cos5\omega t - \dots] \quad \dots\dots\dots A1$$

Thus $s(t)$ contains terms in all odd harmonics of the carrier frequency

The input to the lowpass filter will be the rectifier output, which is:

$$\text{rectifier output} = s(t) \cdot \text{AM} \quad \dots\dots\dots A2$$

Note that the AM is centred on ' ω ', and $s(t)$ is a string of terms on the ODD harmonics of ω . Remembering also that the product of two sinewaves gives 'sum and difference' terms, then we conclude that:

- the 1st harmonic in $s(t)$ gives a term near DC and another centred at 2ω

- the 3rd harmonic in $s(t)$ gives a term at 2ω and 4ω
- the 5th harmonic in $s(t)$ gives a term at 4ω and 6ω
- and so on

We define the AM signal as:

$$AM = A [1 + m(t)] \cos\omega t \quad \dots\dots\dots A3$$

where, for the depth of modulation to be less than 100%, $|m(t)| < 1$.

From the rectified output we are only interested in any term near DC; this is the one we can hear. In more detail:

$$\text{term near DC} = (1/2) \cdot (4/\pi) \cdot A \cdot m(t) \quad \dots\dots\dots A4$$

which is an *exact*, although scaled, copy of the message $m(t)$.

The other terms are copies of the original AM, but on all even multiples of the carrier, and of decreasing amplitudes. They are easily removed with a lowpass filter. The *nearest* unwanted term is a scaled version of the original AM on a carrier frequency 2ω rad/s.

For the case where the carrier frequency is very much higher than the highest message frequency, that is when $\omega \gg \mu$, an inequality which is generally satisfied, the lowpass filter can be fairly simple. Should the carrier frequency not satisfy this inequality, we can still see that the message will be recovered **UNDISTORTED** so long as the carrier frequency is at least twice the highest message frequency, and a filter with a steeper transition band is used.

practical modification

In practice it is easier to make a halfwave than a fullwave rectifier. This means that the expression for $s(t)$ will contain a DC term, and the magnitudes of the other terms will be halved. The effect of this DC term in $s(t)$ is to create an extra term in the output, namely a scaled copy of the input signal.

This is an extra unwanted term, centred on ω rad/s, and in fact the lowest frequency unwanted term. The lowest frequency unwanted term in the fullwave rectified output is centred on 2ω rad/s.

This has put an extra demand upon the lowpass filter. This is not significant when $\omega \gg \mu$, but will become so for lower carrier frequencies.

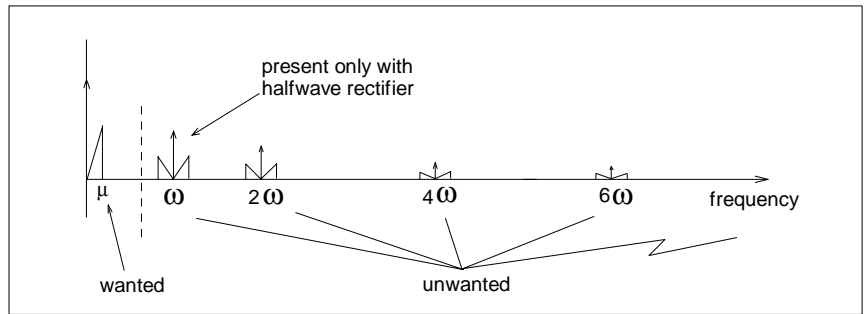


Figure 2A: rectifier output spectrum (approximate scale)