

# INTRODUCTION TO MODELLING WITH TIMS

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# INTRODUCTION TO MODELLING WITH TIMS

## ***model building***

With TIMS you will be building models. These models will most often be hardware realizations of the block diagrams you see in a text book, or have designed yourself. They will also be representations of equations, which themselves can be depicted in block diagram form.

Whatever the origin of the model, it can be patched up in a very short time. The next step is to adjust the model to perform as expected. It is perfectly true that you might, on occasions, be experimenting, or just ‘doodling’, not knowing what to expect. But in most cases your goal will be quite clear, and this is where a systematic approach is recommended.

If you follow the steps detailed in the first few experiments you will find that the models are adjusted in a systematic manner, so that each desired result is obtained via a complete understanding of the purpose and aim of the intermediate steps leading up to it.

## ***why have patching diagrams ?***

Many of the analog experiments, and all of the digital experiments, display patching diagrams. These give all details of the interconnections between modules, to implement a model of the system under investigation.

*It is **not expected** that a glance at the patching diagram will reveal the nature of the system being modelled.*

The patching diagram is presented as firm evidence that a model of the system can be created with TIMS.

*The functional purpose of the system is revealed through the block diagram which precedes the patching diagram.*

It is the block diagram which you should study to gain insight into the workings of the system.

If you fully understand the block diagram you should not need the patching diagram, except perhaps to confirm which modules are required for particular operations, and particular details of functionality. This is available in the *TIMS User Manual*.

You may need an occasional glance at the patching diagram for confirmation of a particular point.

*Try to avoid patching up 'mechanically', according to the patching diagram, without thought to what you are trying to achieve.*

## **organization of experiments**

Each of the experiments in this Text is divided into three parts.

1. The first part is generally titled *PREPARATION*. This part should be studied *before* the accompanying laboratory session.
2. The second part describes the experiment proper. Its title will vary. You will find the experiment a much more satisfying experience if you arrive at the laboratory well prepared, rather than having to waste time finding out what has to be done at the last moment. Thus read this part *before* the laboratory session.
3. The third part consists of TUTORIAL QUESTIONS. Generally these questions will be answered after the experimental work is completed, but it is a good idea to read them *before* the laboratory session, in case there are special measurements to be made.

While performing an experiment you should always have access to the *TIMS* user manuals - namely the *TIMS User Manual* (fawn cover) which contains information about the modules in the TIMS Basic Set of modules, and the *TIMS Advanced Modules and TIMS Special Applications Modules User Manual* (red cover).

## **who is running this experiment ?**

These experiments and their Tasks are merely suggestions as to how you might go about carrying out certain investigations. In the final assessment it is *you* who are running the experiment, and *you* must make up your mind as to how you are going to do it. You can do this best if you read about it beforehand.

If you do not understand a particular instruction, consider what it is you have been trying to achieve up to that point, and then do it your way.

# **early experiments**

The first experiment assumes no prior knowledge of telecommunications - it is designed to introduce you to TIMS, and to illustrate the previous remarks about being systematic. The techniques learned will be applied over and over again in later work.

The next few experiments are concerned with analog modulation and demodulation.

## **modulation**

One of the many purposes of *modulation* is to convert a message into a form more suitable for transmission over a particular medium.

The analog modulation methods to be studied will generally transform the analog message signal in the audio spectrum to a higher location in the frequency spectrum.

The digital modulation methods to be studied will generally transform a binary data stream (the message), at baseband<sup>1</sup> frequencies, to a different format, and then may or may not translate the new form to a higher location in the frequency spectrum.

It is much easier to radiate a high frequency (HF) signal than it is a relatively low frequency (LF) audio signal. In the TIMS environment the particular part of the spectrum chosen for HF signals is centred at 100 kHz.

It is necessary, of course, that the reverse process, *demodulation*, can be carried out - namely, that the message may be recovered from the modulated signal upon receipt following transmission.

# **messages**

Many models will be concerned with the transmission or reception of a message, or a signal carrying a message. So TIMS needs suitable messages. These will vary, depending on the system.

## **analog messages**

The transmission of speech is often the objective in an analog system.

High-fidelity speech covers a wide frequency range, say 50 Hz to 15 kHz, but for communications purposes it is sufficient to use only those components which lie in the audio frequency range 300 to 3000 Hz - this is called 'band limited speech'. Note that frequency components have been removed from both the low and the high frequency end of the message spectrum. This is bandpass filtering. Intelligibility suffers if only the high frequencies are removed.

Speech is not a convenient message signal with which to make simple and precise measurements. So, initially, a single tone (sine wave) is used. This signal is more easily accommodated by both the analytical tools and the instrumentation and measuring facilities.

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<sup>1</sup> defined later

The frequency of this tone can be chosen to lie within the range expected in the speech, and its peak amplitude to match that of the speech. The simple tone can then be replaced by a two-tone test signal, in which case intermodulation tests can be carried out <sup>2</sup>.

When each modulation or demodulation system has been set up quantitatively using a single tone as a message (or, preferably with a two-tone test signal), a final qualitative check can be made by replacing the tone with a speech signal. The peak amplitude of the speech should be adjusted to match that of the tone. Both listening tests (in the case of demodulation) and visual examination of the waveforms can be very informative.

## **digital messages**

The transmission of binary sequences is often the objective of a digital communication system. Of considerable interest is the degree of success with which this transmission is achieved. An almost universal method of describing the quality of transmission is by quoting an error rate <sup>3</sup>.

If the sequence is one which can take one of two levels, say 0 and 1, then an error is recorded if a 0 is received when a 1 was sent, or a 1 received when a 0 was sent. The bit error rate is measured as the number of errors as a proportion of total bits sent.

To be able to make such a measurement it is necessary to know the exact nature of the original message. For this purpose a known sequence needs to be transmitted, a copy of which can be made available at the receiver for comparison purposes. The known sequence needs to be known, and useful, statistical properties - for example, a 'random' sequence. Rather simple generators can be implemented using shift registers, and these provide sequences of adjustable lengths. They are known as pseudo-random binary sequence (PRBS) generators. TIMS provides you with just such a SEQUENCE GENERATOR module. You should refer to a suitable text book for more information on these.

## **bandwidths and spectra**

Most of the signals you will be examining in the experiments to follow have well defined bandwidths. That is, in most cases it is possible to state quite clearly that all of the energy of a signal lies between frequencies  $f_1$  and  $f_2$  Hz, where  $f_1 < f_2$ .

- the *absolute bandwidth* of such a signal is defined as  $(f_2 - f_1)$  Hz.

It is useful to define the number of octaves a signal occupies. The octave measure for the above signal is defined as

$$\text{octaves} = \log_2(f_2 / f_1)$$

Note that the octave measure is a function of the *ratio* of two frequencies; it says nothing about their *absolute* values.

- a *wideband signal* is generally considered to be one which occupies one or more octaves.

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<sup>2</sup> the two-tone test signal is introduced in the experiment entitled '*Amplifier overload*'.

<sup>3</sup> the corresponding measurement in an analog system would be the signal-to-noise ratio (relatively easy to measure with instruments), or, if speech is the message, the 'intelligibility'; not so easy to define, let alone to measure.

- a *narrowband signal* is one which occupies a small fraction of an octave. Another name, used interchangeably, is a *bandpass* signal.

An important observation can be made about a narrowband signal; that is, it can contain no harmonics.

- a *baseband signal* is one which extends from DC (so  $f_1 = 0$ ) to a finite frequency  $f_2$ . It is thus a *wideband* signal.

Speech, for communications, is generally bandlimited to the range 300 to 3000 Hz. It thus has a bandwidth in excess of 3 octaves. This is considered to be a wideband signal. After modulation, to a higher part of the spectrum, it becomes a narrowband signal, but note that *its absolute bandwidth remains unchanged*.

This reduction from a wideband to a narrowband signal is a linear process; it can be reversed. In the context of communications engineering it involves *modulation*, or *frequency translation*.

You will meet all of these signals and phenomena when working with TIMS.

## **measurement**

The bandwidth of a signal can be measured with a SPECTRUM ANALYSER. Commercially available instruments typically cover a wide frequency range, are very accurate, and can perform a large number of complex measurements. They are correspondingly expensive.

TIMS has no spectrum analyser as such, but can model one (with the TIMS320 DSP module), or in the form of a simple WAVE ANALYSER with TIMS analog modules. See the experiment entitled *Spectrum analysis - the WAVE ANALYSER* (within *Volume A2 - Further & Advanced Analog Experiments*).

Without a spectrum analyser it is still possible to draw conclusions about the location of a spectrum, by noticing the results when attempting to pass it through filters of different bandwidths. There are several filters in the TIMS range of modules. See Appendix A, and also the *TIMS User Manual*.

## **graphical conventions**

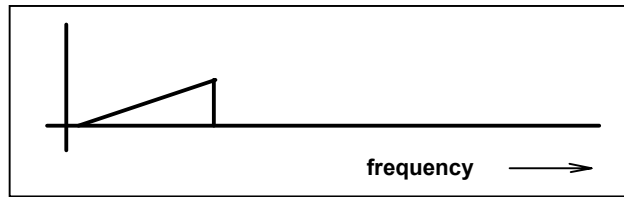
### **representation of spectra**

It is convenient to have a graphical method of depicting spectra. In this work we do not get involved with the Fourier transform, with its positive and negative frequencies and double sided spectra. Elementary trigonometrical methods are used for analysis. Such methods are more than adequate for our purposes.

When dealing with speech the mathematical analysis is dropped, and descriptive methods used. These are supported by graphical representations of the signals and their spectra.

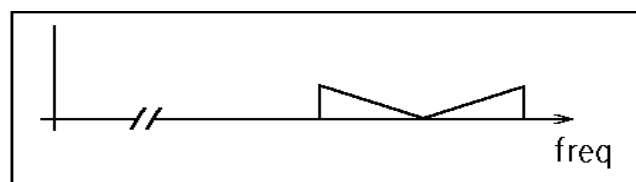
In the context of modulation we are constantly dealing with *sidebands*, generally derived from a baseband message of finite bandwidth. Such finite bandwidth signals will be represented by triangles on the spectral diagrams.

The steepness of the slope of the triangle has no special significance, although when two or more sidebands, from different messages, need to be distinguished, each can be given a different slope.



**a baseband signal (eg., a message)**

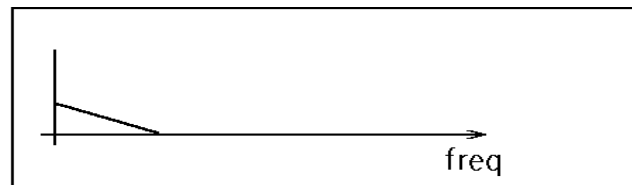
Although speech does not have a DC component, the triangle generally extends down to zero (the origin) of the frequency scale (rather than being truncated just before it). For the special case in which a baseband signal does have a DC component the triangle convention is sometimes modified slightly by adding a vertical line at the zero-frequency end of the triangle.



**a DSBSC**

The *direction* of the slope is important. Its significance becomes obvious when we wish to draw a modulated signal. The figure above shows a double sideband suppressed carrier (DSBSC) signal.

Note that there are TWO triangles, representing the individual lower and upper sidebands. They slope towards the same point; this point indicates the location of the (suppressed) carrier frequency.

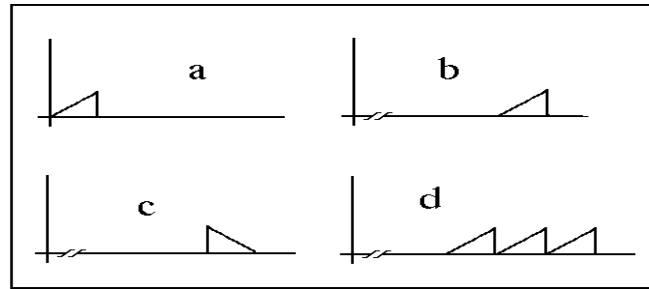


**an inverted baseband signal**

The *orientation* is important. If the same message was so modulated that it could be represented in the frequency spectrum as in the figure above, then this means:

- the signal is located in the baseband part of the spectrum
- spectral components have been transposed, or inverted; frequency components which were originally above others are now below them.
- since the signal is at baseband it would be audible (if converted with an electric to acoustic transducer - a pair of headphones, for example), but would be unintelligible. You will be able to listen to this and other such signals in TIMS experiments to come.

It is common practice to use the terms *erect* and *inverted* to describe these bands.



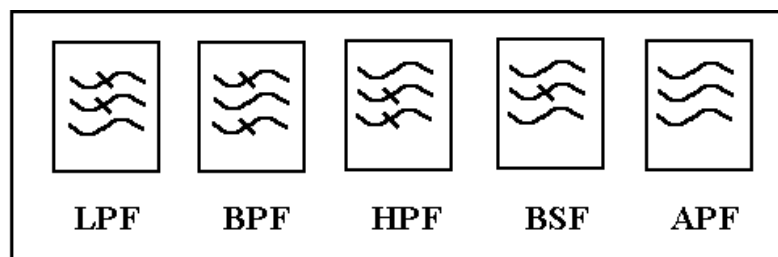
In the Figure above, a message (a) is frequency translated to become an upper single sideband (b), and a lower single sideband (c). A three-channel frequency division multiplexed (FDM) signal is also illustrated (d).

Note that these spectral diagrams do not show any phase information.

Despite all the above, be prepared to accept that these diagrams are used for purposes of *illustration*, and different authors use their own variations. For example, some slope their triangles in the opposite sense to that suggested here.

## filters

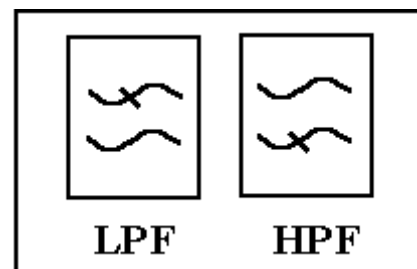
In a block diagram, there is a simple technique for representing filters. The frequency spectrum is divided into three bands - low, middle, and high - each represented by part of a sinewave. If a particular band is blocked, then this is indicated by an oblique stroke through it. The standard responses are represented as in the Figure below.



**block-diagrammatic representations of filter responses**

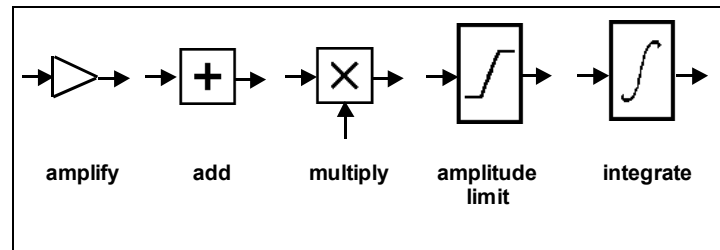
The filters are, respectively, lowpass, bandpass, highpass, bandstop, and allpass.

In the case of lowpass and highpass responses the diagrams are often further simplified by the removal of one of the cancelled sinewaves, the result being as in the figure opposite.





## other functions



some analog functions

# measuring instruments

## the oscilloscope - time domain

The most frequently used measuring facility with TIMS is the oscilloscope. In fact the vast majority of experiments can be satisfactorily completed with no other instrument.

Any general purpose oscilloscope is ideal for all TIMS experiments. It is intended for the display of signals in the time domain<sup>4</sup>. It shows their waveforms - their shapes, and amplitudes

From the display can be obtained information regarding:

- waveform shape
- waveform frequency - by calculation, using time base information
- waveform amplitude - directly from the display
- system linearity - by observing waveform distortion
- an *estimate* of the bandwidth of a complex signal; eg, from the sharpness of the corners of a square wave

When concerned with amplitude information it is customary to record either:

- the *peak-to-peak* amplitude
- the *peak* amplitude

of the waveform visible on the screen.

Unless the waveform is a simple sinewave it is always important to record the *shape* of the waveform also; this can be:

1. as a sketch (with time scale), and annotation to show clearly what amplitude has been measured.
2. as an analytic expression, in which case the parameter recorded must be clearly specified.

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<sup>4</sup> but with adaptive circuitry it can be modified to display frequency-domain information

## **the rms voltmeter**

The TIMS WIDEBAND TRUE RMS METER module is essential for measurements concerning *power*, except perhaps for the simple case when the signal is one or two sinewaves. It is particularly important when the measurement involves *noise*.

Its bandwidth is adequate for all of the signals you will meet in the TIMS environment.

An experiment which introduces the WIDEBAND TRUE RMS METER, is entitled *Power measurements*. Although it appears at the end of this Volume, it could well be attempted at almost any time.

## **the spectrum analyser - frequency domain**

The identification of the spectral composition of a signal - its components in the frequency domain - plays an important part when learning about communications.

Unfortunately, instruments for *displaying* spectra tend to be far more expensive than the general purpose oscilloscope.

It is possible to identify and measure the individual spectral components of a signal using TIMS modules.

Instruments which identify the spectral components on a component-by-component basis are generally called *wave analysers*. A model of such an instrument is examined in the experiment entitled *Spectrum analysis - the WAVE ANALYSER* in *Volume A2 - Further & Advanced Analog Experiments*.

Instruments which identify the spectral components of a signal and *display* the spectrum are generally called *spectrum analysers*. These instruments tend to be more expensive than wave analysers. Something more sophisticated is required for their modelling, but this is still possible with TIMS, using the digital signals processing (DSP) facilities - the TIMS320 module can be programmed to provide spectrum analysis facilities.

Alternatively the distributors of TIMS can recommend other affordable methods, compatible with the TIMS environment.

# **oscilloscope - triggering**

## **synchronization**

As is usually the case, to achieve 'text book like' displays, it is important to choose an appropriate signal for oscilloscope triggering. This trigger signal is almost never the signal being observed! The recognition of this point is an important step in achieving stable displays.

This chosen triggering signal should be connected *directly* to the oscilloscope sweep synchronizing circuitry. Access to this circuitry of the oscilloscope is available via an input socket other than the vertical deflection amplifier input(s). It is typically labelled '*ext. trig*' (external trigger), '*ext. synch*' (external synchronization), or similar.

## **sub-multiple frequencies**

If two or more periodic waveforms are involved, they will only remain stationary with respect to each other if the frequency of one is a sub-multiple of the other.

This is seldom the case in practice, but can be made so in the laboratory. Thus TIMS provides, at the MASTER SIGNALS module, a signal of 2.083 kHz (which is 1/48 of the 100 kHz system clock), and another at 8.333 kHz (1/12 of the system clock).

### **which channel ?**

Much time can be saved if a consistent use of the SCOPE SELECTOR is made. This enables quick changes from one display to another with the flip of a switch. In addition, *channel identification* is simplified if the habit is adopted of consistently locating the trace for CH1 *above* the trace for CH2.

Colour coded patching leads can also speed trace identification.

## **what you see, and what you don't**

Instructions such as '*adjust the phase until there is no output*', or '*remove the unwanted signal with a suitable filter*' will be met from time to time.

These instructions seldom result in the amplitude of the signal in question being reduced to zero. Instead, what is generally meant is '*reduce the amplitude of the signal until it is no longer of any significance*'.

Significance here is a relative term, made with respect to the system signal-to-noise ratio (SNR). All systems have a background noise level (noise *threshold*, noise *floor*), and signals (wanted) within these systems must over-ride this noise (unwanted).

TIMS is designed to have a 'working level', the TIMS ANALOG REFERENCE LEVEL, of about 4 volts peak-to-peak. The system noise level is claimed to be at least 100 times below this <sup>5</sup>.

When using an oscilloscope as a measuring instrument with TIMS, the vertical sensitivity is typically set to about 1 volt/cm. Signals at the reference level fit nicely on the screen. If they are too small it is wise to increase them if possible (and appropriate), to over-ride the system noise; or if larger to reduce them, to avoid system overload.

When they are attenuated by a factor of 100, and if the oscilloscope sensitivity is not changed, they *appear* to be '*reduced to zero*'; and in relative terms this is so.

If the sensitivity of the oscilloscope is increased by 100, however, the screen will no longer be empty. There will be the system noise, and perhaps the signal of interest is still visible. Engineering judgement must then be exercised to evaluate the significance of the signals remaining.

## **overload**

If wanted signal levels within a system fall 'too low' in amplitude, then the signal-to-noise ratio (SNR) will suffer, since internal circuit noise is independent of signal level.

If signal levels within a system rise 'too high', then the SNR will suffer, since the circuitry will overload, and generate extra, unwanted, distortion components; these distortion components are signal level dependent. In this case the noise is

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<sup>5</sup> TIMS claims a system signal-to-noise ratio of better than 40 dB

derived from distortion of the signal, and the degree of distortion is usually quoted as signal-to-distortion ratio (SDR).

Thus analog circuit design includes the need to maintain signal levels at a pre-defined working level, being 'not too high' and 'not too low', to avoid these two extremes.

These factors are examined in the experiment entitled *Amplifier overload* within *Volume A2 - Further & Advanced Analog Experiments*.

The TIMS working signal level, or TIMS ANALOG REFERENCE LEVEL, has been set at 4 volts peak-to-peak. Modules will generally run into non-linear operation when this level is exceeded by say a factor of two. The background noise of the TIMS system is held below about 10 mV - this is a fairly loose statement, since this level is dependent upon the bandwidth over which the noise is measured, and the model being examined at the time. A general statement would be to say that TIMS endeavours to maintain a SNR of better than 40 dB for all models.

## **overload of a narrowband system**

Suppose a channel is narrowband. This means it is deliberately bandlimited so that it passes signals in a narrow (typically much less than an octave<sup>6</sup>) frequency range only. There are many such circuits in a communications system.

If this system overloads on a single tone input, there will be unwanted harmonics generated. But these will not pass to the output, and so the overload may go unnoticed. With a more complex input - say two or more tones, or a speech-related signal - there will be, in addition, unwanted intermodulation components generated. Many of these *will* pass via the system, thus revealing the existence of overload. In fact, the two-tone test signal should always be used in a narrowband system to investigate overload.

## **the two-tone test signal**

A two-tone test signal consists of two sine waves added together ! As discussed in the previous section, it is a very useful signal for testing systems, especially those which are of narrow-bandwidth. The properties of the signal depend upon:

- the frequency ratio of the two tones.
- the amplitude ratio of the two tones.

For testing narrowband communication systems the two tones are typically of near-equal frequency, and of identical amplitude. A special property of this form of the signal is that its shape, as seen in the time domain, is very well defined and easily recognisable<sup>7</sup>.

After having completed the early experiments you will recognise this shape as that of the double sideband suppressed carrier (DSBSC) signal.

If the system through which this signal is transmitted has a non-linear transmission characteristic, then this will generate extra components. The presence of even small amounts of these components is revealed by a change of shape of the test signal.

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<sup>6</sup> defined above

<sup>7</sup> the assumption being that the oscilloscope is set to sweep across the screen over a few periods of the *difference* frequency.

# **Fourier series and bandwidth estimation**

Fourier series analysis of periodic signals reveals that:

- it is possible, by studying the symmetry of a signal, to predict the presence or absence of a DC component.
- if a signal is other than sinusoidal, it will contain *more than* one harmonic component of significance.
- if a signal has sharp discontinuities, it is likely to contain *many* harmonic components of significance
- some special symmetries result in all (or nearly all) of the ODD (or EVEN) harmonics being absent.

With these observations, and more, it is generally easy to make an engineering estimate of the bandwidth of a periodic signal.

## **multipliers and modulators**

The modulation process requires *multiplication*. But a pure MULTIPLIER is seldom found in communications equipment. Instead, a device called a MODULATOR is used.

In the TIMS system we generally use a MULTIPLIER, rather than a MODULATOR, when multiplication is called for, so as not to become diverted by the side effects and restrictions imposed by the latter.

In commercial practice, however, the purpose-designed MODULATOR is generally far superior to the unnecessarily versatile MULTIPLIER.

### **multipliers**

An ideal multiplier performs as a multiplier should ! That is, if the two time-domain functions  $x(t)$  and  $y(t)$  are multiplied together, then we expect the result to be  $x(t) \cdot y(t)$ , no more and no less, and no matter what the nature of these two functions. These devices are called *four quadrant* multipliers.

There are practical multipliers which approach this ideal, with one or two engineering qualifications. Firstly, there is always a restriction on the bandwidth of the signals  $x(t)$  and  $y(t)$ .

There will inevitably be extra (unwanted) terms in the output (noise, and particularly distortion products) due to practical imperfections.

Provided these unwanted terms can be considered 'insignificant', with respect to the magnitude of the wanted terms, then the multiplier is said to be 'ideal'. In the TIMS environment this means they are at least 40 dB below the TIMS ANALOG REFERENCE LEVEL <sup>8</sup>.

Such a multiplier is even said to be linear. That is, from an engineering point of view, it is performing as expected.

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<sup>8</sup> defined under 'what you see and what you don't'

In the mathematical sense it is not linear, since the mathematical definition of a linear circuit includes the requirement that no new frequency components are generated when it performs its normal function. But, as will be seen, multiplication always generates new frequency components.

### **DC off-sets**

One of the problems associated with analog circuit design is minimization of unwanted DC off-sets. If the signals to be processed have no DC component (such as in an audio system) then stages can be AC coupled, and the problem is overcome. In the TIMS environment module bandwidths must extend to DC, to cope with all possible conditions; although more often than not signals have no intentional DC component.

In a complex model DC offsets can accumulate - but in most cases they can be recognised as such, and accounted for appropriately. There is one situation, however, where they can cause much more serious problems by generating *new components* - and that is when *multiplication* is involved.

With a MULTIPLIER the presence of an unintentional DC component at one input will produce new components at the output. Specifically, each component at the other input will be multiplied by this DC component - a constant - and so a scaled version will appear at the output<sup>9</sup>.

To overcome this problem there is an option for AC coupling in the MULTIPLIER module. It is suggested that the DC mode be chosen only when the signals to be processed actually have DC components; otherwise use AC coupling.

## **modulators**

In communications practice the circuitry used for the purpose of performing the multiplying function is not always ideal in the *four quadrant multiplier* sense; such circuits are generally called *modulators*.

Modulators generate the *wanted* sum *or* difference products but in many cases the input signals will also be found in the output, along with other *unwanted* components at significant levels. Filters are used to remove these unwanted components from the output (alternatively there are 'balanced' modulators. These have managed to eliminate either one or both of the original signals from the output).

These modulators are restricted in other senses as well. It is allowed that one of the inputs can be complex (ie., two or more components) but the other can only be a single frequency component (or appear so to be - as in the switching modulator). This restriction is of no disadvantage, since the vast majority of modulators are required to multiply a complex signal by a single-component carrier.

Accepting restrictions in some areas generally results in superior performance in others, so that in practice it is the *switching modulator*, rather than the idealized *four quadrant multiplier*, which finds universal use in communications electronics.

Despite the above, TIMS uses the four quadrant multiplier in most applications where a modulator might be used in practice. This is made possible by the relatively low frequency of operation, and modest linearity requirements

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<sup>9</sup> this is the basis of a voltage controlled amplifier - VCA

## **envelopes**

Every narrowband signal has an envelope, and you probably have an idea of what this means.

Envelopes will be examined first in the experiment entitled *DSB generation* in this Volume.

They will be defined and further investigated in the experiments entitled *Envelopes* within this Volume, and *Envelope recovery* within *Volume A2 - Further & Advanced Analog Experiments*.

## **extremes**

Except for a possible frequency scaling effect, most experiments with TIMS will involve realistic models of the systems they are emulating. Thus message frequencies will be 'low', and carrier frequencies 'high'. But these conditions need not be maintained. TIMS is a very flexible environment.

*It is always a rewarding intellectual exercise to imagine what would happen if one or more of the 'normal' conditions was changed severely<sup>10</sup>.*

It is then even more rewarding to confirm our imaginings by actually modelling these unusual conditions. TIMS is sufficiently flexible to enable this to be done in most cases.

For example: it is frequently stated, for such-and-such a requirement to be satisfied, that it is necessary that ' $x_1 \gg x_2$ '. Quite often  $x_1$  and  $x_2$  are frequencies - say a carrier and a message frequency; or they could be amplitudes.

You are strongly encouraged to expand your horizons by questioning the reasons for specifying the conditions, or restrictions, within a model, and to consider, and then examine, the possibilities when they are ignored.

## **analog or digital ?**

What is the difference between a digital signal and an analog signal ? Sometimes this is not clear or obvious.

In TIMS digital signals are generally thought of as those being compatible with the TTL standards. Thus their amplitudes lie in the range 0 to +5 volts. They come from, and are processed by, modules having RED output and input sockets.

It is sometimes necessary, however, to use an analog filter to bandlimit these signals. But their large DC offsets would overload most analog modules. Some digital modules (eg, the SEQUENCE GENERATOR) have anticipated this, and provide an analog as well as a digital (TTL) output. This analog output comes

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<sup>10</sup> for an entertaining and enlightening look at the effects of major changes to one or more of the physical constants, see G. Gamow; *Mr Tompkins in Wonderland* published in 1940, or easier *Mr. Tompkins in Paperback*, Cambridge University Press, 1965.

from a YELLOW socket, and is a TTL signal with the DC component removed (ie, DC shifted).

## ***SIN or COS ?***

Single frequency signals are generally referred to as sinusoids, yet when manipulating them trigonometrically are often written as cosines. How do we obtain  $\cos\omega t$  from a sinusoidal oscillator !

There is no difference in the *shape* of a sinusoid and a cosinusoid, as observed with an oscilloscope. A sinusoidal oscillator can just as easily be used to provide a cosinusoid. What we call the signal (sin or cos) will depend upon the time reference chosen.

Remember that  $\cos\omega t = \sin(\omega t + \pi/2)$

Often the time reference is of little significance, and so we choose sin or cos, in any analysis, as is convenient.

## ***the ADDER - G and g***

Refer to the *TIMS User Manual* for a description of the ADDER module. Notice it has two input sockets, labelled 'A' and 'B'.

In many experiments an ADDER is used to make a linear sum of two signals  $a(t)$  and  $b(t)$ , of amplitudes **A** and **B** respectively, connected to the inputs A and B respectively. The proportions of these signals which appear at the ADDER output are controlled by the front panel gain controls **G** and **g**.

The amplitudes **A** and **B** of the two input signals are seldom measured, nor the magnitudes **G** and **g** of the adjustable gains.

Instead it is the magnitudes **GA** and **gB** which are of more interest, and these are measured directly at the ADDER output. The measurement of **GA** is made when the patch lead for input **B** is removed; and that of **gB** is measured when the patch lead for input **A** is removed.

When referring to the two inputs in this text it would be formally correct to name them as 'the input A' and 'the input B'. This is seldom done. Instead, they are generally referred to as 'the input **G**' and 'the input **g**' respectively (or sometimes just **G** and **g**). This should never cause any misunderstanding. If it does, then it is up to you, as the experimenter, to make an intelligent interpretation.



# abbreviations

This list is not exhaustive. It includes only those abbreviations used in this Text.

<i>abbreviation</i>	<i>meaning</i>
AM	amplitude modulation
ASK	amplitude shift keying (also called OOK)
BPSK	binary phase shift keying
CDMA	code division multiple access
CRO	cathode ray oscilloscope
dB	decibel
DPCM	differential pulse code modulation
DPSK	differential phase shift keying
DSB	double sideband (in this text synonymous with DSBSC)
DSBSC	double sideband suppressed carrier
DSSS	direct sequence spread spectrum
DUT	device under test
ext. synch.	external synchronization (of oscilloscope). 'ext. trig.' preferred
ext. trig.	external trigger (of an oscilloscope)
FM	frequency modulation
FSK	frequency shift keying
FSD	full scale deflection (of a meter, for example)
IP	intermodulation product
ISB	independent sideband
ISI	intersymbol interference
LSB	<i>analog</i> : lower sideband <i>digital</i> : least significant bit
MSB	most significant bit
NBFM	narrow band frequency modulation
OOK	on-off keying (also called ASK)
PAM	pulse amplitude modulation
PCM	pulse code modulation
PDM	pulse duration modulation (see PWM)
PM	phase modulation
PPM	pulse position modulation
PRK	phase reversal keying (also called PSK)
PSK	phase shift keying (also called PRK - see BPSK)
PWM	pulse width modulation (see PDM)
SDR	signal-to-distortion ratio
SNR	signal-to-noise ratio
SSB	single sideband (in this text is synonymous with SSBSC)
SSBSC	single sideband suppressed carrier
SSR	sideband suppression ratio
TDM	time division multiplex
THD	total harmonic distortion
VCA	voltage controlled amplifier
WBFM	wide band frequency modulation

## ***list of symbols***

The following symbols are used throughout the text, and have the following meanings

$a(t)$  a time varying amplitude

$\alpha, \phi, \varphi,$  phase angles

$\beta$  deviation, in context of PM and FM

$\delta f$  a small frequency increment

$\Delta\phi$  peak phase deviation

$\delta t$  a small time interval

$\phi(t)$  a time varying phase

$m$  in the context of envelope modulation, the *depth* of modulation

$\mu$  a low frequency (rad/s); typically that of a message ( $\mu \ll \omega$ ).

$\omega$  a high frequency (rad/s); typically that of a carrier ( $\omega \gg \mu$ )

$y(t)$  a time varying function