Electronic Signals and Systems

7761L
Student Workbook

National Module No. EA190
Electrical Engineering
St George TAFE
Sydney Institute

MODULE SECTIONS

Section 1: Electrical and electronic systems

SUGGESTED DURATION	PREAMBLE
4 hrs	To enable you to identify the similarities and important differences between an electrical power distribution system and a communications system, and to introduce students to the concept of a control system and the difference between open and closed loop operation.

This section covers learning outcomes 1 and 2 of the Module Descriptor.

Objectives

At the end of this section you should be able to:

☐ draw the block diagram of an information transfer system showing:
- input transducer
- transmission medium
- amplifier
- output transducer
- power supply

☐ describe the basic purpose of each block in an information transfer system and describe the general operation of the system

☐ indicate typical power levels in an information transfer system

☐ draw a simplified block diagram of a power distribution system showing power input and losses

☐ describe the general characteristics of an electrical power transfer system

☐ identify where energy is lost in conversion and transmission processes in electrical and electronic systems

☐ identify the possible need for feedback in electrical and electronic systems

☐ state the different characteristics of open and closed loop systems

☐ given examples of systems, determine whether they are open or closed loop

☐ given examples of closed loop systems, determine the action of the feedback function.

Electrical and electronic systems

Overview of systems

An engineering definition of a system is ' a collection of things which receives certain inputs and acts on them to produce certain outputs, with the objective of maximising some function of the inputs and outputs'. This broad definition includes physical and non-physical systems. Electrical, mechanical, hydraulic, acoustical and thermal systems are examples of physical systems. Economic, political, and industrial-planning systems are examples of socio-economic systems.

A system is characterised by its inputs, its outputs (or responses) and the laws of operation. In electrical systems, the laws of operation are the current-voltage relationships for the various components and the laws of interconnection, i.e. Kirchhoff's laws. These laws can be used to derive mathematical equations relating the outputs to the inputs. These equations form a mathematical model of the system. Thus, a system is characterised by its inputs, its outputs and its mathematical model.

In this section we will be taking a broad look at electrical and electronic systems. We will compare electrical power systems with electronic communication systems and we will look at open and closed loop control used in these systems.

Let's start with a block diagram of a simple information transfer system.

Information transfer system
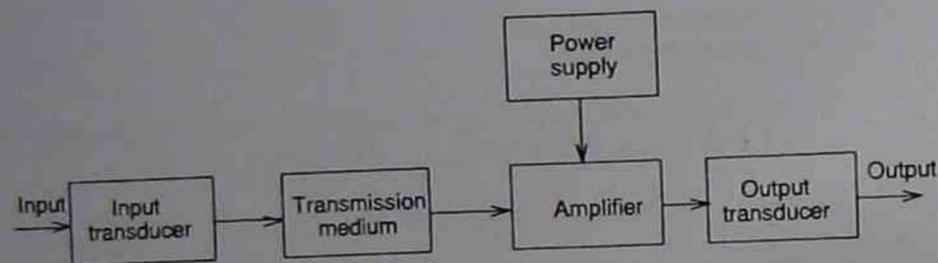
- The purpose of the input transducer is to convert the information from the source into an electrical signal.
- The transmission medium conveys the signal to the amplifier, where the form of the signal is not altered, but its power level is increased. The added power is supplied by the power supply, which is a DC source (most likely derived from an AC source).

Now let's look at a practical information transfer system - a public address system.
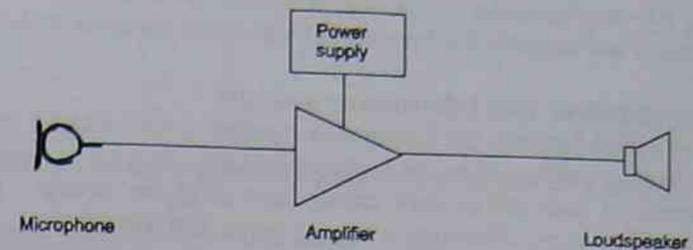

- The microphone is the input transducer.
- The microphone cable and the loudspeaker cable are the transmission media.
- The loudspeaker is the output transducer.

The power output from the microphone would be of the order of nW, and the power delivered to the loudspeaker would be of the order of a few Watts, depending on the application. In a telephone system, we would find power levels of the order of a mW. (1mW is a convenient reference level and is defined as 0dBm.)

Now let us look at a power distribution system and compare it with the information transfer system.

Power distribution system
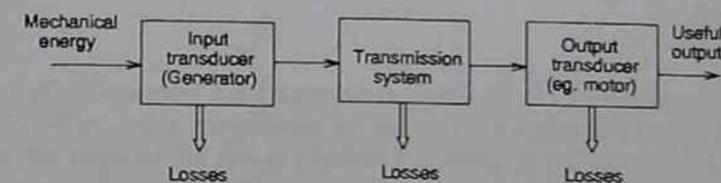
- The mechanical input power may come from a steam turbine, falling water, a diesel generator or some other source. The generator converts most of this power to electrical energy and the remainder is lost as heat.

- The transmission medium consists of transformers and transmission lines. There will be considerable i^2R losses in the transmission lines. Transformers are used to allow the use of high voltage and low current to minimise i^2R losses.

TAFE

8

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

9

TAFE

- The output transducer converts some of the received energy to mechanical energy and the remainder is lost as heat. (Some of the energy is converted to mechanical and acoustical vibrations but these also dissipate as heat.)

## Comparison of power and information systems

There are similarities between the information transfer system and the power distribution system. For example, the loudspeaker cable in our P.A. system may have transformers at each end to allow transmission at higher voltage. Though this results in lower i^2R losses, efficiency is not the prime consideration.

- The prime consideration in a power distribution system is efficiency, ie. minimising power (or energy) loss.

- The prime consideration in an information transfer system is waveform integrity. Waveforms are degraded by noise and various forms of distortion.

Open and closed loop control systems

Closed loop control systems differ from open loop control systems in that they use negative feedback. Feedback simply means a transfer of information between the output and input of a system.

Here is an example of a negative feedback amplifier.
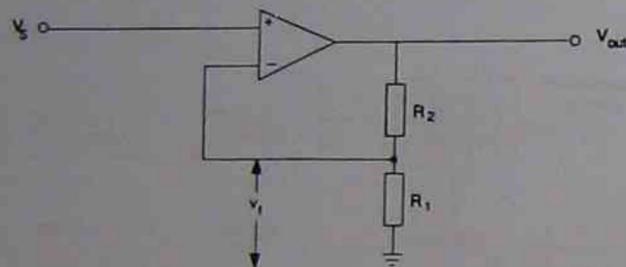
$$v_f = \frac{R_1}{R_1 + R_2} v_{out}$$

The output voltage is sampled by the resistive voltage divider to produce a feedback voltage, v_f. This voltage is compared, ie. subtracted from the original input signal V_s, and the difference is amplified.

It is quite easy to show that if the amplifier gain is A and the fraction of the output which is fed back (V_f / V_{out}) is B, then

$$if \; |AB| \; > \; >1, \qquad \frac{v_{out}}{v_s} \approx \frac{1}{B}.$$

Our mathematical model characterising the system is now very simple.

In our example above, if $R_1 = 1k\Omega$ and $R_2 = 9k\Omega$

$$then \; B = \frac{R_1}{R_1 + R_2} = \frac{1}{10} \quad and \quad \frac{v_{out}}{v_s} = 10$$

Here is another example. The position of a shaft is controlled by a potentiometer. The amplifier adjusts the position of the shaft until the control potentiometer and the position sensing potentiometer, are equal.
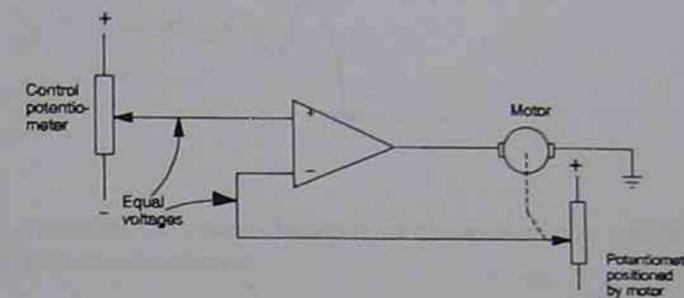

Negative feedback control systems occur in many areas of engineering, as well as in biological systems and socio-economic systems. An example of a biological system is the way the pupil of the eye is adjusted according to the intensity of received light to maintain a fairly constant illumination of the retina. An example of a soico-economic system - prices go up, sales go down, so that revenue from sales stays roughly constant.

An example of an open loop system is the control of a car's speed by the position of the accelerator. Ideally the speed might depend only on the accelerator position but there are other controlling factors - wind speed, road gradient, load, engine condition, gear selected. If cruise control is used (a closed loop system) then these factors are almost eliminated. There is one factor, however, which cannot be eliminated. This is the effect of mass (or inertia) when we wish to change speed.

All feedback systems are affected by inertia of some kind. If the price of chocolate goes up, it may be some time before your chocolate consumption settles to a lower level! Feedback amplifiers always have capacitance (even if it is only stray wiring capacitance) which provide 'inertia' and affect the response to a change in input. Where a motor is involved, there is clearly a large amount of 'inertia'.

10

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

11

Our simplified mathematical model does not allow for inertia but the following diagrams show how a closed loop control system might respond to a change in input.

Input

Possible output responses

time
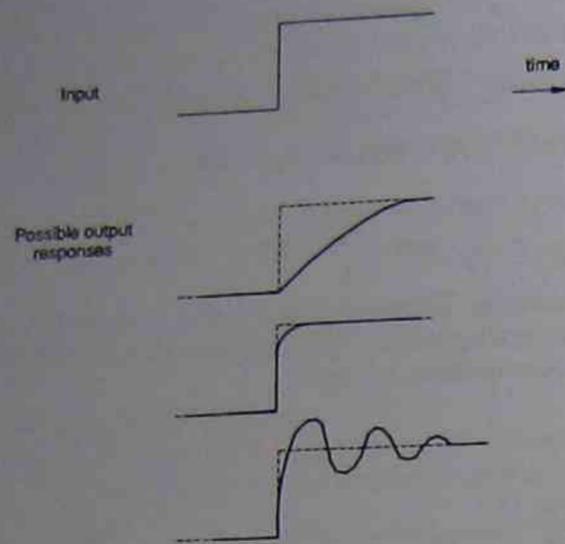
Review questions

These questions will help you revise what you have learnt in Section 1.

1. Draw a block diagram of an energy transfer system, showing power input and losses.

2. State two important differences between an electrical power transfer system and an information transfer system.

_____

_____

_____

_____

_____

3. State which of the following electrical/electronic systems would use closed loop control.

    - The speed control on a kitchen food processor.
    - The laser tracking system in a compact disk player.
    - A remote controller for a model aeroplane.
    - An AGC system in a radio receiver.
    - An oven for the crystal in a high stability oscillator.
    - The pressure controller in a steel rolling mill.

    _____

    _____

    _____

    _____

    _____

    _____

4. Briefly describe the action of the feedback in a voltage amplifier.

    _____

    _____

    _____

    _____

5. Two motor speed controllers are controlled by potentiometers. One uses open loop control and the other uses closed loop control. Briefly compare the likely control characteristics.

    _____

    _____

    _____

    _____

Section 2: Signals, spectra and non-linearity

SUGGESTED DURATION	PREAMBLE
7 hrs	To introduce you to Fourier concepts and the spectra of some common signals, and to extend these concepts in explaining the effect of non-linearity.

This section covers learning outcomes 3 and 4 of the Module Descriptor.

Objectives

At the end of this section you should be able to:

- ☐ predict the spectral frequencies of a given periodic wave

- ☐ recognise that a non-periodic signal such as speech contains a continuous band of frequencies

- ☐ sketch typical time and frequency domain diagrams for white noise, speech, music, video and random binary data

- ☐ relate one line of video to a grey scale

- ☐ define non-linearity

- ☐ calculate harmonic and intermodulation distortion frequencies.

Time and frequency domains

Time domain
The traditional method of observing electrical signals is to view them in the time domain, using an oscilloscope.
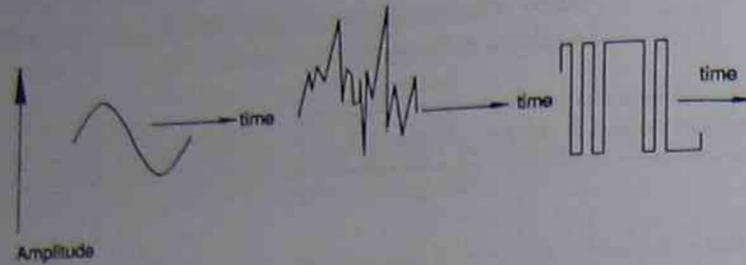
Time domain displays

The information displayed is amplitude (voltage) versus time, which is adequate for most low frequency audio and digital waveform measurements involving timing and phase.

However, time domain measurements are not usually adequate when studying RF devices such as amplifiers, oscillators, filters, mixers, modulators and antennas. The reasons for this are given below.

1. Oscilloscopes cannot normally view waveforms above several hundred MHz. Their internal amplifiers are not capable of amplifying many of the high frequency signals found in communications equipment. For example, AUSSAT signals return to earth at 12GHz (12000 MHz!!!). An oscilloscope can not be used to view these signals.

2. Oscilloscopes are often not sensitive enough to display the tiny signals found in communications equipment. For example, most oscilloscopes have 1mV/cm as the most sensitive range, and could not display signals with uV levels. It would be even more difficult to display a 1uV signal at the same time as a 10V signal.

3. An oscilloscope cannot break a complex signal down into its constituent parts; it displays them all added together. Many signals are complex; that is, they are composed of more than one frequency component. It is impossible with an oscilloscope to examine individual components of a complex wave.

Are you ready to throw your oscilloscope away? Don't! Even though it can't do the things mentioned above, it is probably the most versatile general purpose laboratory instrument. Besides, the instrument that can do everything mentioned above may cost between $10,000 and $100,000. This expensive instrument is called a spectrum analyser.

Frequency domain
In general, the term frequency domain refers to any graph or measurement which is taken as a function of frequency. The most commonly encountered measurement is amplitude versus frequency. The resulting display is known as a spectrum or a spectral diagram.

The spectrum analyser displays amplitude versus frequency on the screen. A signal having only one frequency component appears as a single vertical line. The height of this line represents amplitude measured either in volts or milliwatts or dBm (another way of expressing power measurements). The position of this vertical line along the horizontal axis tells us its frequency.
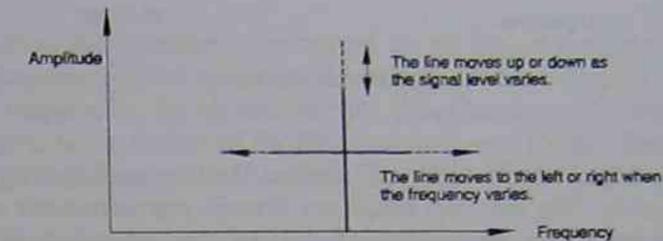
Frequency domain display

One feature of the spectrum analyser is that it allows either linear or logarithmic (dB) scales. The logarithmic scale permits both large and small signals to be displayed simultaneously. For example a signal which is 60dB below another is $\frac{1}{1,000,000}$ of that signal's power. On a linear scale only the larger signal would be seen. Viewed on an oscilloscope, the effect of the smaller signal would not be noticeable.

TAFE

16

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

17

TAFE

Amplitude
10dB/div

(a) Logarithmic (b) Linear
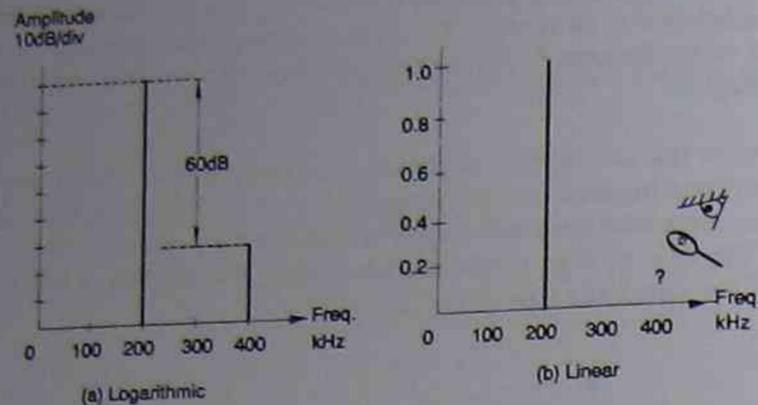
Spectrum analyser display

In many cases a signal can be observed in either the time or frequency domains. The choice is yours. The time and frequency domain representations of a signal are complementary, and if one representation is known, then the other can be derived from it.

This field of mathematics is known as Fourier analysis, after Jean Baptiste Joseph, Baron de Fourier (1768-1830). He accompanied Napoleon on the Egyptian campaign in 1798, becoming Governor of Lower Egypt before returning to France where he produced his classic paper 'Theories Analytique de la Chaleur' (Analysis of the Flow of Heat). In it he evolved the mathematical series which bears his name today, and has found application in most branches of applied science.

Summary
- Time domain refers to signals and quantities viewed as a function of time.
- The oscilloscope displays signals in the time domain.
- Frequency domain refers to signals and quantities viewed as a function of frequency.
- The spectrum analyser displays signals in the frequency domain.

Fundamentals of Fourier Analysis

Stated in the simplest of terms, Fourier's theorem says:

> A complex periodic waveform may be analysed as a number of harmonically related sinusoidal waves.

This means that we can synthesise (make) any complex periodic waveform by adding together pure sine waves in the right amounts. Electronic music can be created in exactly this way: certain combinations of sine waves may sound like a flute, while another combination may sound like a fog-horn. The term periodic simply means that the waveform repeats itself after a given time period T.

The frequencies of the constituent sine waves are all integer multiples of the fundamental frequency of the waveform concerned. These multiples are known as the harmonic frequencies, the second multiple being known as the second harmonic, the third multiple being known as the third harmonic and so on. The first harmonic is just the original frequency, and is referred to simply as the fundamental.

This applies to all complex periodic waveforms, such as square, triangle, pulsed and sawtooth signals. An ideal sine wave however, has only a fundamental component and no harmonics.

Generally speaking, the higher harmonics are weaker than the lower ones, although the individual amplitudes may vary in a complex manner. (Note that in the figure on the next page, the ninth harmonic is larger than the eighth harmonic). Also note that the fundamental or any harmonic(s) may have zero amplitude.

An example of this is the square wave which has only odd harmonics. Another property of the square wave is that the third harmonic has an amplitude $\frac{1}{3}$ that of the fundamental, the fifth harmonic has an amplitude $\frac{1}{5}$ that of the fundamental and so on.

TAFE

18

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

19

TAFE

Amplitude

No even harmonics!

Freq.

0 1 2 3 4 5 6 7 8 9 10 11

(a) Square wave

Amplitude

Freq.

0 1 2 3 4 5 6 7 8 9 10 11

(b) Rectangular wave
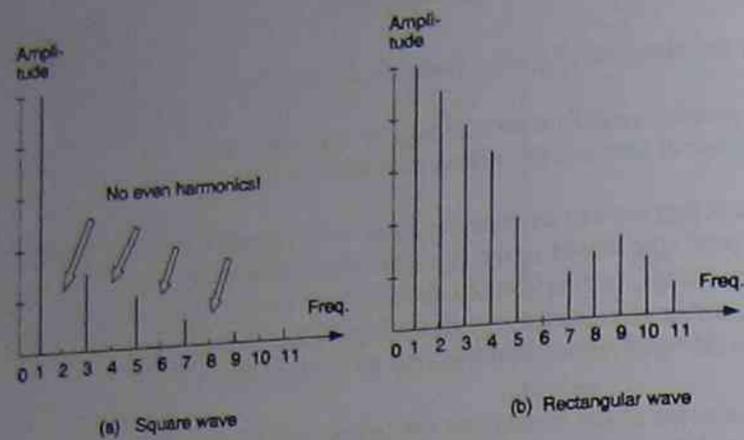
Distribution of harmonics

Example

A waveform has a period T = 40mS. Calculate the frequency of the fundamental, and the second, third and fourth harmonics.

$$\text{Fundamental frequency} = \frac{1}{T}$$

$$= \frac{1}{40 \times 10^{-3}}$$

$$= 25Hz$$

Second harmonic   = 2 x 25Hz
= 50Hz

Third harmonic   = 3 x 25Hz
= 75Hz

Fourth harmonic   = 4 x 25Hz
= 100Hz

Let us now see how a square wave can be made by adding a fundamental frequency and a number of harmonics. You will recall that only odd harmonics are required, and that the third harmonic has $\frac{1}{3}$ the amplitude of the fundamental and so on.

Figure (a) below shows the fundamental, the third and fifth harmonics and their phase relationships, and Figure (b) shows the resultant. Note that all odd harmonics to infinity must be considered to construct a perfect square wave, although in practice the higher order harmonics become insignificant.

Final square wave

A

Fundamental

3rd harmonic

5th harmonic

Amplitude

Phase
relationship
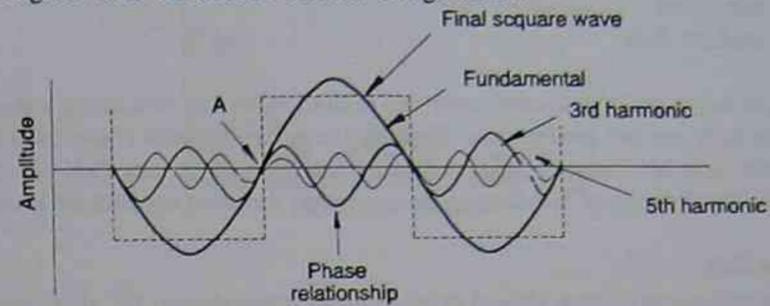
(a) Components of a square wave

Frequncies to the
5th harmonic
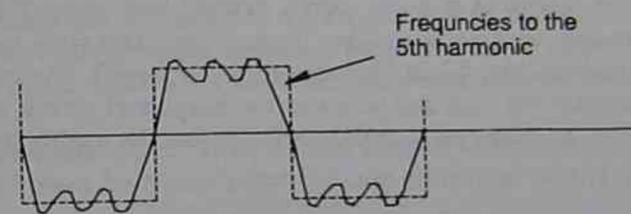
(b) Resultant of frequencies to fifth harmonic

20

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

21

TAFE

TAFE

Waveforms and spectral diagrams for common signals

You will now be introduced to some of the waveforms commonly found in communications equipment and systems. The waveforms are:

- the sine wave
- the square wave
- white noise
- speech
- music
- television
- random data.

The sine wave and the square wave are periodic, but the remaining signals are not, because they are not predictable. Each of these has its own characteristic frequency spectrum, and we know in general terms what each one looks like, but the precise detail of the frequency distribution at any given moment can not be predicted.

Bandwidth

A communication system should provide good transmission for all frequencies where the signal power spectrum is significant.

Speech

For speech, the entire collection of vocal sounds extends from about 80Hz to 12kHz, with strongly decreasing energy at the higher frequencies. This wide range of frequencies is required for high fidelity broadcast quality speech. For most communications purposes (eg. taxi and police radio, telephone) such a wide range is unnecessary and it can be restricted to 300-3400Hz before the intelligibility suffers.

Music

For high fidelity music, the band from 30Hz to 15kHz is required.

Video

For video, frequencies from 0 to 5MHz are required.

Random data

For data, the required bandwidth is related to the bit rate. The higher the bit rate, the more bandwidth is required. For most applications we can say that the required bandwidth is equal to the reciprocal of duration of the narrowest pulse.
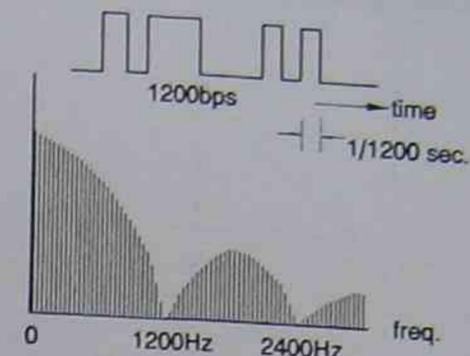
Spectrum of random data

The term bandwidth in relation to the various signals above, refers to the numerical difference between the upper and lower frequency limits of the signal. For example, the speech signal above has a minimum acceptable bandwidth of 3100Hz.

But what does this mean? Does it mean that the component frequencies in the speech that comes from your telephone stop sharply at 3400Hz?

No! It means that outside the limits of 300Hz and 3400Hz, the spectral components are weaker by 3 decibels (dB) or more below the strongest spectral components within the 300 to 3400Hz band.

Note: The decibel is a logarithmic power ratio which is used throughout all fields of electronics. In particular, 3 dB means that the power has fallen by half.

In the case of speech which has been transmitted through a telephone network, the bandwidth will have been reduced so that the 3dB points are 300Hz and 3400Hz.

TAFE

22

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

23

TAFE

Symbolic representation of the baseband

Since baseband signals have varying spectral diagrams, it is necessary to adopt a standard representation for all of them, regardless of their actual spectra. The standard shape is show in the figure below, the hypotenuse of the triangle increasing in the direction of increasing frequency.
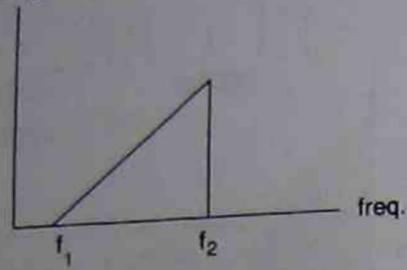

Symbolic representation of the baseband

Note that this symbolic shape usually bears no resemblance to the actual baseband signal in the system. Speech signals in fact have the opposite characteristic.

The significance of this representation will become apparent when we consider the topic Modulation in Section 4.

The spectrum of a pulse train

For a rectangular pulse train with period T and pulse width τ as shown below, the relative amplitudes of the spectral components can be found from the $\frac{sinx}{x}$ curve.
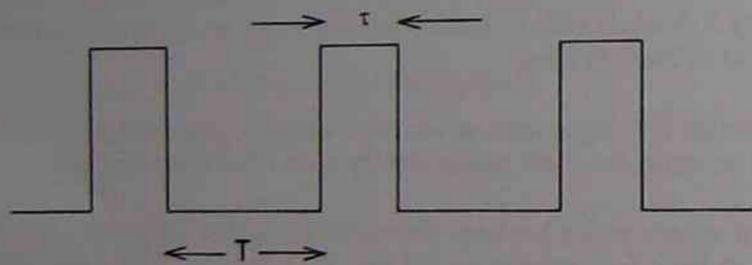
The first zero point on the $\frac{sinx}{x}$ curve corresponds to the frequency $\frac{1}{\tau}$.

The fundamental frequency is, of course, $\frac{1}{T}$.
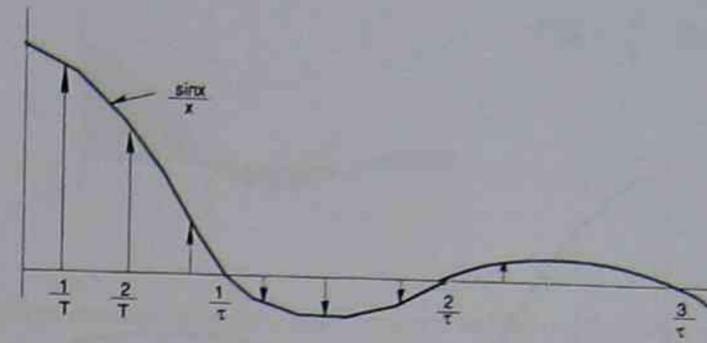
Example 1

$T = 1ms, \quad \tau = 0.2ms$

$\therefore \quad \frac{1}{T} = 1kHz, \quad \frac{1}{\tau} = 5kHz$

Locate 5kHz at the first zero of the curve.
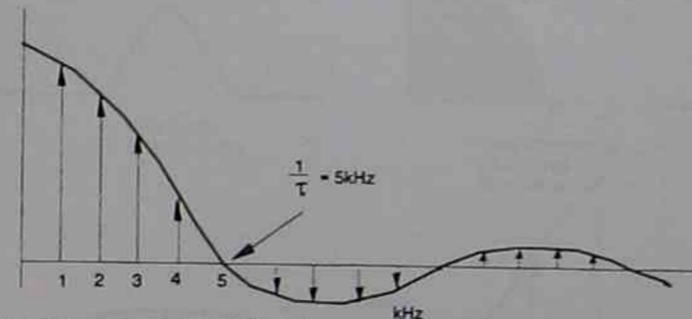
In this case the 5th harmonic (5kHz) has zero amplitude.

Note that when the above spectra are displayed on a spectrum analyzer, all the spectral lines are shown above the horizontal axis.

24

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

25

Example 2

T = 1ms,   τ = 0.5ms (square wave)

$$\therefore \frac{1}{T} = 1kHz, \quad \frac{1}{\tau} = 2kHz$$
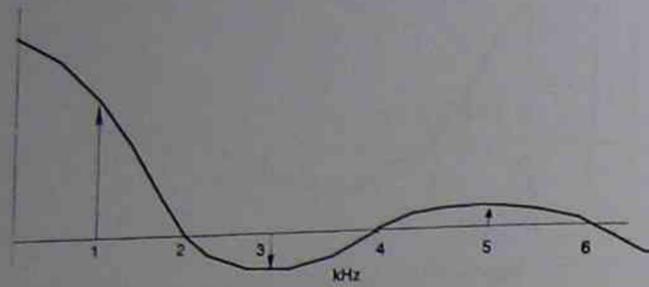

Example 3

T = 1ms,   τ = 0.4ms

$$\therefore \frac{1}{T} = 1kHz, \quad \frac{1}{\tau} = 2.5kHz$$
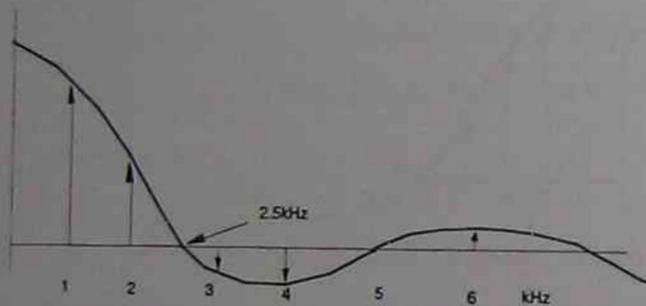
Note that in this case $\frac{1}{\tau}$ is not a harmonic, and that the first component to have zero amplitude is at 5kHz (the fifth harmonic).

As T increases, the spectral lines get closer together.

If T → ∞ (ie. we have only one pulse), the spectral lines form a continuum.
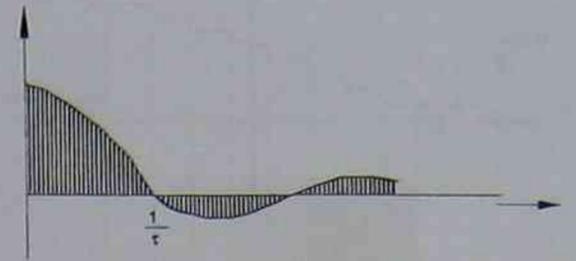
Note that a large proportion of the pulse's energy lies below the frequency $\frac{1}{\tau}$.

Even one cycle of a sinewave has a continuous spectrum. This is used as a test signal in Television.

TAFE

26

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

27

TAFE
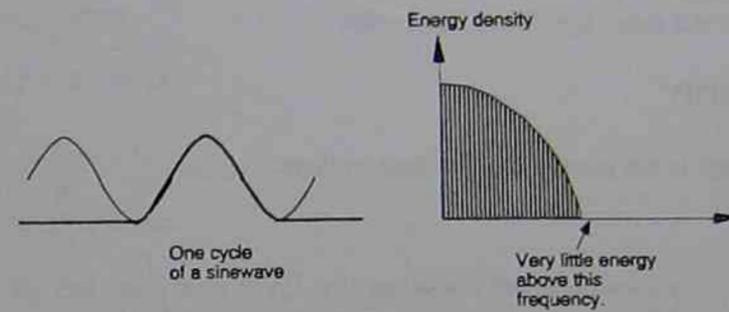
The video signal
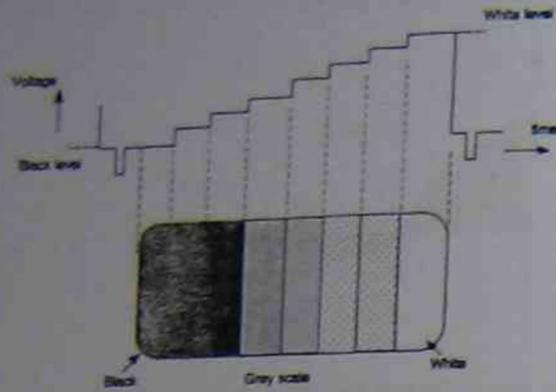
Spectrum with varying signal:
- Luminance: 0 - 5MHz
- Chrominance: Centred on 4.43MHz
- Horizontal sync: 15625Hz + harmonics.

Non-linearity

Output voltage is not proportional to input voltage.
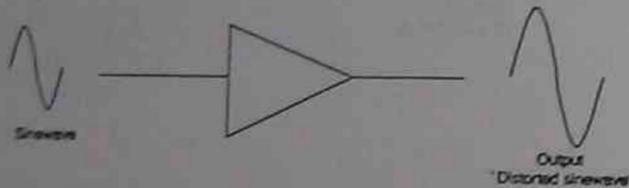
Output is not a sinewave but is still periodic. It therefore contains harmonics and has the same fundamental frequency as the input.

A linear device has $v_0 \propto v_{in}$

$$v_0 = A v_{in}$$
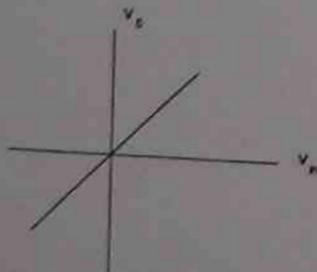
Non-linear device
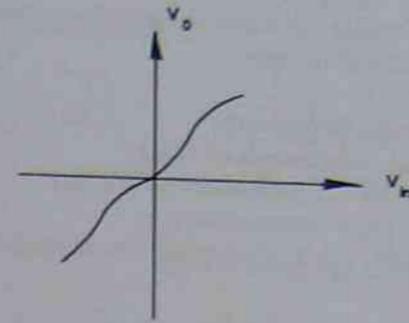
The curve can be represented by as power series:

$$v_0 = A_1 v_{in} + A_2 v_{in}^2 + A_3 v_{in}^3 + ...$$

Harmonic distortion

If we let $v_{in} = \sin\omega t$

then $A_2 v_{in}^2 = A_2 \sin^2\omega t$

$$= \frac{A_2}{2} - \frac{A_2}{2} \cos2\omega t$$

$\cos2\omega t = $ 2nd harmonic.

Likewise the 3rd order term $A_3 v_{in}^3$ will produce a 3rd harmonic.

Intermodualation distortion
Let the input consist of two sinewaves:

$$v_{in} = \sin\omega_1 t + \sin\omega_2 t$$

The second term in the power series above becomes
$A_2 V_{in}^2 = A_2 (\sin\omega_1 t + \sin\omega_2 t)^2$

A little trigonometry will show that this term produces not only the harmonics $2\omega_1$ and $2\omega_2$ but also sum and difference frequencies $\omega_1 \pm \omega_2$

The 3rd order term will produce $\omega_1 \pm 2\omega_2$
and $2\omega_1 \pm \omega_2$.

TAFE

28   EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook   29
June 1995

TAFE

Example

A 1kHz sinewave and a 10kHz sinewave are added together and applied to a non-linear device. The output will include:

- 1kHz, 10kHz (original frequencies)
- 2kHz, 3kHz, 4kHz ......(harmonics of 1kHz)
- 20kHz, 30kHz, 40kHz ......(harmonics of 10kHz)
- 9kHz, 11kHz (2nd order intermodulation)
- 8kHz, 12kHz, 19kHz, 21kHz (3rd order intermodulation).

TABLE 3-1 Some Periodic Waveforms and Their Fourier Series Mathematical Expressions

a. $v(t) = A \cos 2\pi f_0 t$, $f_0 = \dfrac{1}{T}$

b. $v(t) = V_0 + A \cos 2\pi f_0 t$

c. $v(t) = \dfrac{A}{\pi} + \dfrac{A}{2}\sin 2\pi f_0 t - \dfrac{2A}{3\pi}\cos 2\pi(2f_0)t - \dfrac{2A}{15\pi}\cos 2\pi(4f_0)t + \ldots$

$= \dfrac{A}{\pi} + \dfrac{A}{2}\sin 2\pi f_0 t + \displaystyle\sum_{n=2}^{\infty}\dfrac{A[1+(-1)^n]}{\pi(1-n^2)}\cos 2\pi(nf_0)t$

d. $v(t) = \dfrac{2A}{\pi} + \dfrac{4A}{3\pi}\cos 2\pi f_0 t - \dfrac{4A}{15\pi}\cos 2\pi(2f_0)t + \ldots$

$= \dfrac{2A}{\pi} + \displaystyle\sum_{n=1}^{\infty}\dfrac{4A(-1)^n}{\pi[1-(2n)^2]}\cos 2\pi(nf_0)t$

* (the rectifier input signal will have a period of 2 *T*)

e. $v(t) = \dfrac{2A}{\pi}\sin 2\pi f_0 t + \dfrac{2A}{3\pi}\sin 2\pi(3f_0)t + \ldots$

$= \displaystyle\sum_{n,\,\text{odd only}}^{\infty}\dfrac{2A}{n\pi}\sin 2\pi(nf_0)t$

f. $v(t) = \dfrac{2A}{\pi}\cos 2\pi f_0 t - \dfrac{2A}{3\pi}\cos 2\pi(3f_0)t + \dfrac{2A}{5\pi}\cos 2\pi(5f_0)t + \ldots$

$= \displaystyle\sum_{n=1}^{\infty}\left(A\dfrac{\sin n\pi/2}{n\pi/2}\right)\cos 2\pi(nf_0)t$

g. $v(t) = \dfrac{A\tau}{T} + \displaystyle\sum_{n=1}^{\infty}\left(2A\dfrac{\tau}{T}\right)\left(\dfrac{\sin n\pi\tau/T}{n\pi\tau/T}\right)\cos 2\pi(nf_0)t$

h. $v(t) = \displaystyle\sum_{n,\,\text{odd only}}^{\infty}\left(A\dfrac{\sin n\pi/4}{n\pi/4}\right)\cos 2\pi(nf_0)t$

(special case of 50% "alternate inversion")

i. $v(t) = \dfrac{8A}{\pi^2}\cos 2\pi f_0 t + \dfrac{8A}{9\pi^2}\cos 2\pi(3f_0)t + \dfrac{8A}{25\pi^2}\cos 2\pi(5f_0)t + \ldots$
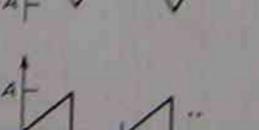
$= \displaystyle\sum_{n,\,\text{odd}}^{\infty}\dfrac{8A}{(n\pi)^2}\cos 2\pi(nf_0)t$

j. $v(t) = \dfrac{2A}{\pi}\left[\sin 2\pi f_0 t - \dfrac{1}{2}\sin 2\pi(2f_0)t + \dfrac{1}{3}\sin 2\pi(3f_0)t + \ldots\right]$

$= \displaystyle\sum_{n=1}^{\infty}[(-1)^{n+1}]\left(\dfrac{2A}{n\pi}\right)\sin 2\pi(nf_0)t$

Paul Young   Electronic Communication Technique

These questions will help you revise what you have learnt in Section 2.

1. For each of the following waveforms, state the frequencies of the first three components present.

(a)

1ms

Output of a common emitter amplifier

(b)

1ms

Output of an amplifier with crossover distortion

(c)

Square wave

10μs

where $X = \frac{n\pi}{T}$

$y = \frac{\sin \pi X}{\pi X}$

(d)

Sawtooth wave

1μs

(e)

Triangular
wave

1μs

(f)

Output of a
half wave
rectifier

20ms

(g)

Output of a full
wave rectifier

10ms

(Discuss this with your teacher!)
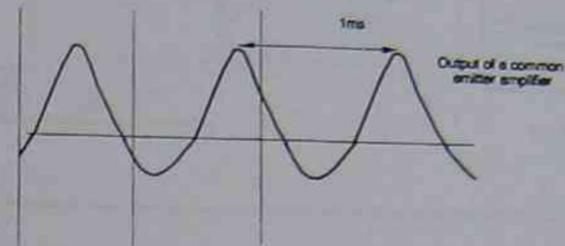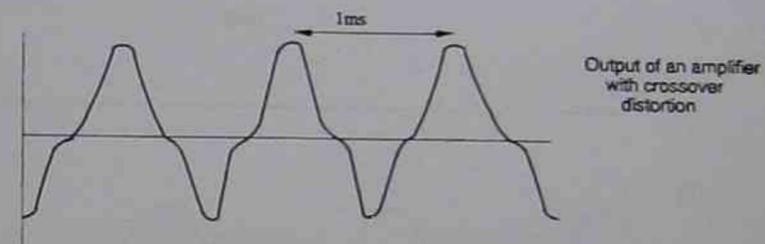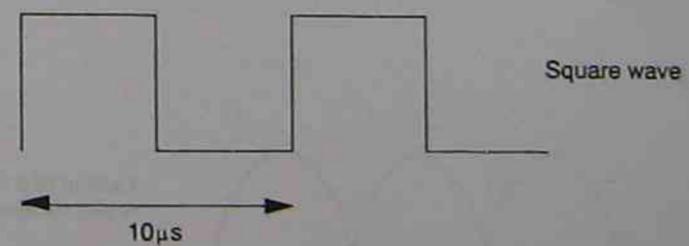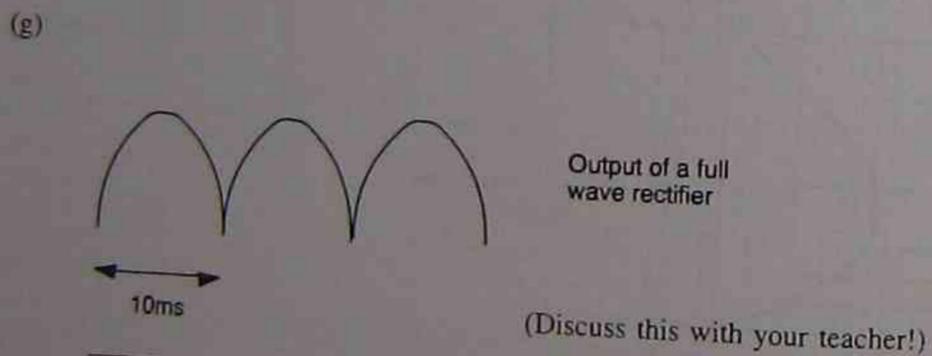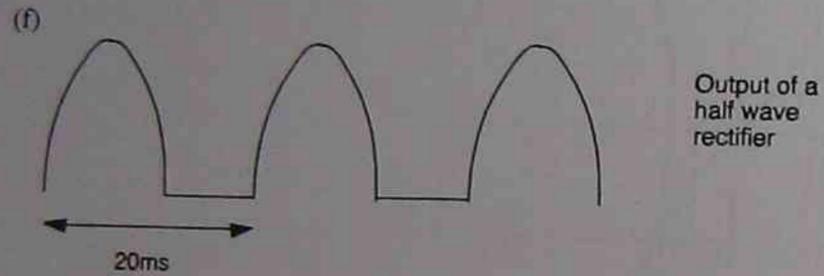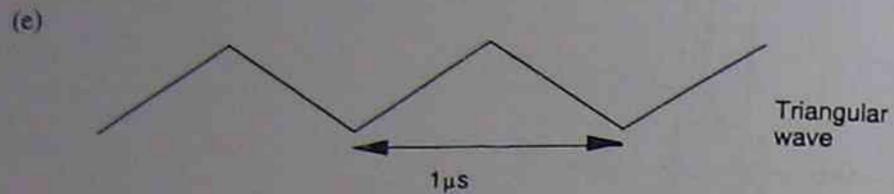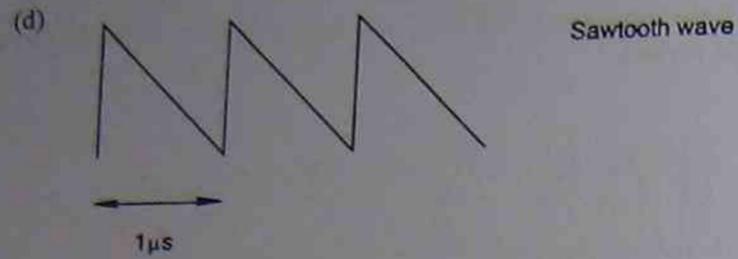
2. For the waveform shown below, will the 5th harmonic be present?

Rectangular
wave

2μs

10μs
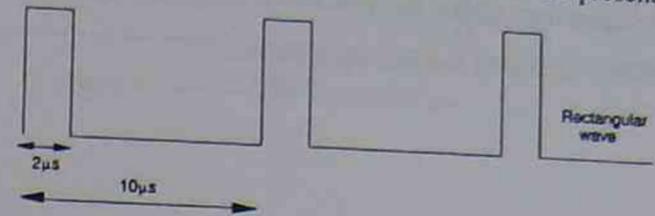
3. A rectangular pulse train has a peak amplitude of 5V, a pulse width of 2ms and a period of 5ms.

(a) Calculate the DC ('average') value of this waveform.

(b) Calculate the mark-space ratio.

(c) On a $\frac{1}{x}$ sinx curve, sketch the five lowest frequency components in the waveform, excluding the DC component.

(d) Which harmonics have zero amplitude?

_____

4. (a) Using a $\frac{1}{x}$ sinx curve, sketch the spectra of each of the pulse trains below, marking the frequency order of the harmonic at the first zero of the curve.

(i)

(ii)
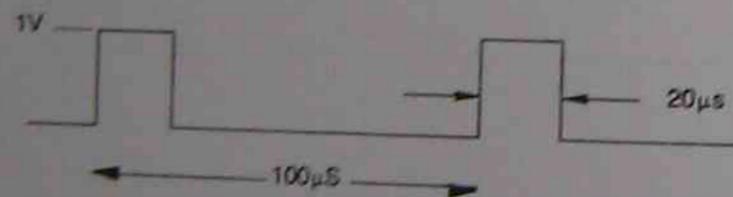
(iii)
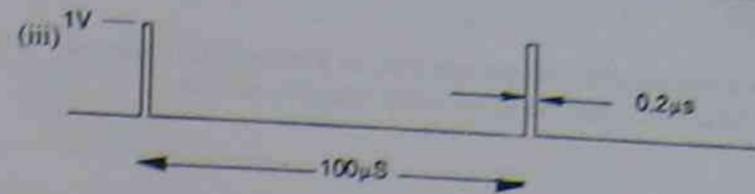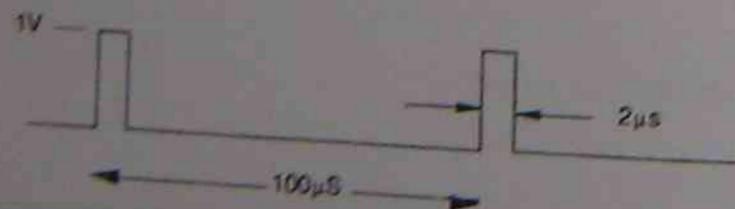
(iv)
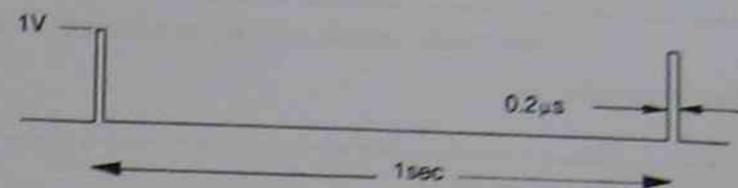
(v)
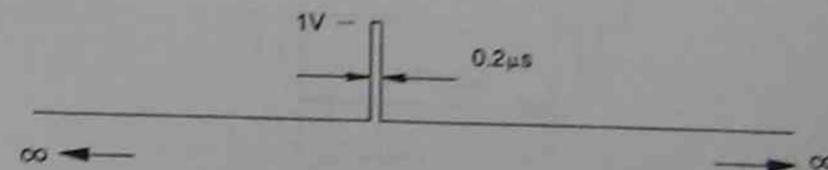
(b) Describe the effect on the spectrum of a rectangular pulse train if the pulse width is decreased, while the pulse period remains constant.

_____

_____

_____

_____

Insertion Loss

The loss introduced into the passband when a filter is inserted into a system. (The same as transmission loss for an ideal filter with equal input and load impedances)
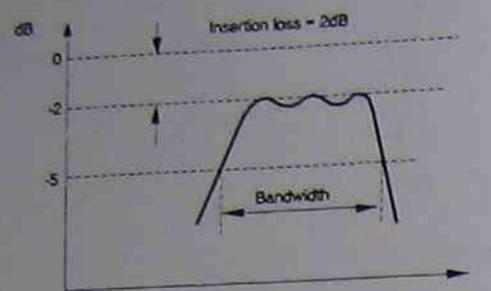
Low Pass Filtering (reduction of high frequency response) - results in a loss of clarity and intelligibility of speech.

High Pass filtering (reduction of low frequency response) - results in a loss of fullness of tone.

Low pass filtering of square waves

Examples

(i) An ideal LPF removing everything above the 5th harmonic.
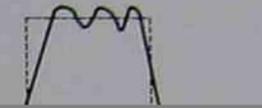
4. How is the bandwidth of a filter normally defined?

_____

Question 5 refers to the figure below which shows the amplitude/frequency response of a filter.
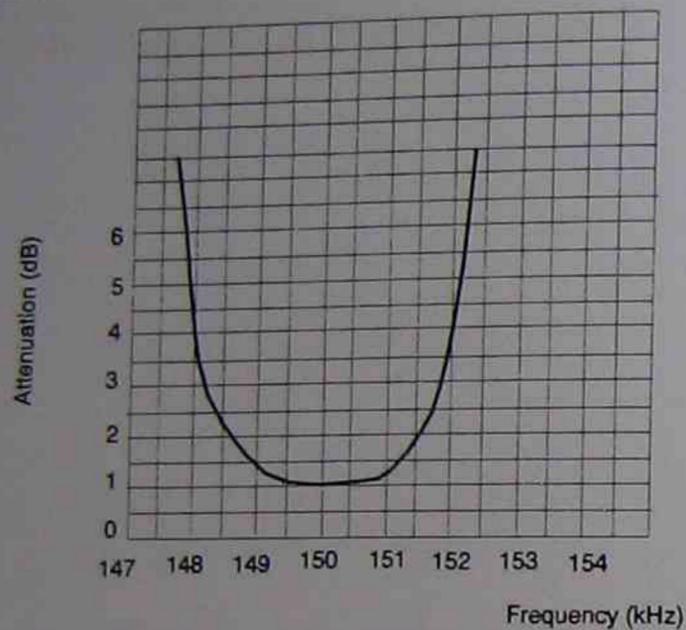
5. (a) What kind of filter is it?

    _____

    (b) State is the insertion loss.

    _____

    (c) Determine the 3dB bandwidth.

    _____

6. State is the ultimate roll-off slope of the simplest R-C filter.

    _____

7. A 2MHz LPF is inserted between the composite video output of a colour video source and the input of a video monitor. List two effects on the picture.

    _____

    _____

8. Sketch the effect of a 2400Hz LPF on the 2400 bps data signal below.
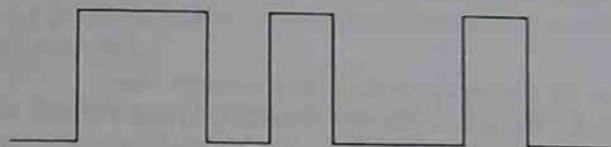
9. Why is the phase/frequency response of the filter referred to in Question 8 important?

    _____
    _____
    _____
    _____

10. Give two reasons for using filters in communications equipment.

    _____
    _____

Skill practice 2
Laboratory report 2

Measurement of ceramic filter response
Suggested duration
Skill practice: 2 hours
Laboratory report: 25 minutes

Assessment
- This skill practice forms part for the assessment for this module. It assesses part of learning outcomes 3 and 4 of the Module Descriptor.
- You must work on your own in this exercise when completing the assessable part of this exercise.
- The assessable part of this exercise is Step 8.
- You must complete the assessable part of this exercise in 25 minutes.
- The total marks available is 5.

Tasks
- To measure the frequency response of a ceramic filter.
- To observe the effect of a ceramic filter on a square wave.

Equipment
- communications trainer
- ceramic filter module
- frequency counter module
- AC voltmeter
- oscilloscope
- audio oscillator (10Hz to 1MHz) sine/square

Procedure
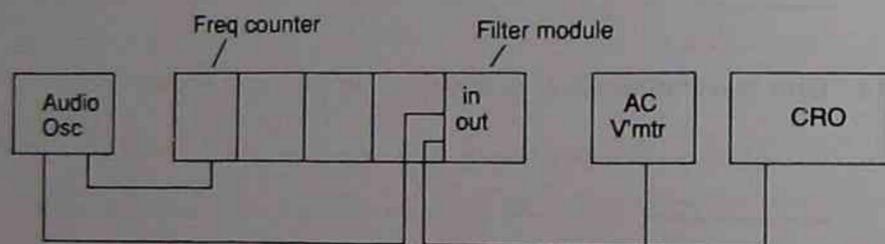
1. Before installing the Ceramic Filter Module into the Communications Trainer frame, examine the module and locate the actual ceramic filter (colour red). Note that there are transistor amplifier stages before and after the filter. These must not be overdriven.

2. Connect the frequency counter module to the square wave output of the audio oscillator. This provides a constant level signal for reliable counting.

Section 4: Communication systems

SUGGESTED DURATION	PREAMBLE
6 hrs	To introduce you to the principles of a communications system, the need for modulation and the effects of noise on both analog and digital systems.

This section covers learning outcome 6 of the Module Descriptor.

Objectives

At the end of this section you should be able to:

☐ draw the block diagram of a general communication system, with or without modulation

☐ explain the difference between analog and digital signals and list three sources of baseband analog and three sources of baseband digital signals

☐ state two reasons for modulation

☐ state three reasons for the differences between received and transmitted information

☐ explain the term 'multiplexing'

☐ with respect to radio communications, state the decade frequency ranges from VLF and SHF to state which bands are used for:
  - AM broadcasting
  - FM broadcasting
  - television
  - satellite communications

☐ list and explain the main differences between analog channels and digital channels

☐ explain how digital signals may be sent over analog channels which were not designed for such signals (eg. data over telephone lines)

☐ draw a block diagram with waveforms, of the communication system arrangements listed below and give an example of where each combination could be found in practice
  - analog signal (speech) over an analog channel (telephone line)
  - digital signal (data) over an analog channel (telephone line) using FSK or PSK modulation
  - analog signal over a digital channel
  - digital signal over a digital channel.

Communication systems

Block diagram of a communications system
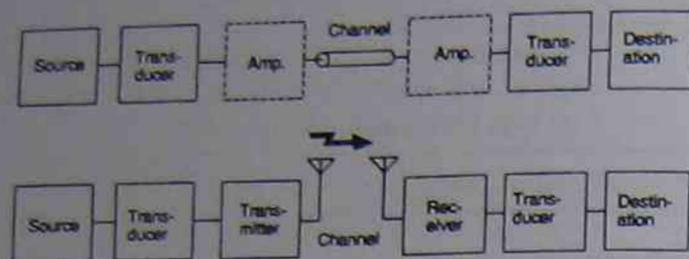
Two possible block diagrams of a communications system are shown above. The first diagram shows a baseband system where the channel is a cable. The second diagram shows a system employing modulation and radio transmission.

Example of information sources are:
- someone speaking into a microphone
- the picture 'seen' by a video camera
- temperature or pressure being measured by a transducer
- information on a computer disk.

Following the source is a transducer which is a device that converts the information into an electrical signal. Examples include a microphone, video camera or temperature sensor. The transducer at the destination end converts the electrical signal to another form, such as sound or light. Possible transducers at the receiving end include a loudspeaker, a video monitor and a digital display.

The transmitter will include a modulator to convert the baseband signal (ie. the transducer output) to a form more suitable for the channel. For example, when an audio signal is transmitted by radio, the radio signal is of a much higher frequency. This allows the use of a more convenient size antenna and allows different radio stations to use different frequencies. Using modulation allows a large number of telephone channels to use the same pair of wires, optical fibre or microwave radio link. A computer modem (modulator/demodulator) allows digital data to be transmitted over a channel designed for speech.

The channel on our block diagrams should strictly be defined as a means of one way communication. In a radio system, the channel occupies a certain frequency band. In a telephone system the frequency band is normally 4kHz wide. Broadcast television Channel 2 occupies the band 63-70MHz in the radio spectrum. A single pair of wires may carry one or two baseband channels (as in the local telephone network) or, with the use of modulation, a large number of channels. A cable or radio system carrying many channels is often referred to as a bearer. The term channel may refer to one link in the system or a set of links.

Multiplexing

The term multiplexing refers to the use of one bearer to carry more than one channel. Frequency division multiplexing (FDM) is where each baseband signal is modulated to a different frequency range. Time division multiplexing (TDM) is where different digital signals or samples of analog signals are sent in different time slots. For example, a 12 channel FDM telephone system may allocate 4kHz to each channel and occupy the range 60-108kHz. Radio broadcasting (including television) uses FDM as each station or channel occupies a different frequency range.

The diagrams below illustrate a 12 channel FDM system. Each of the 12 baseband signals occupies the range 0-4kHz. After modulation, each channel occupies a 4kHz slot.

0   4kHz

Before modulation

60kHz   108kHz

62   EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995   63
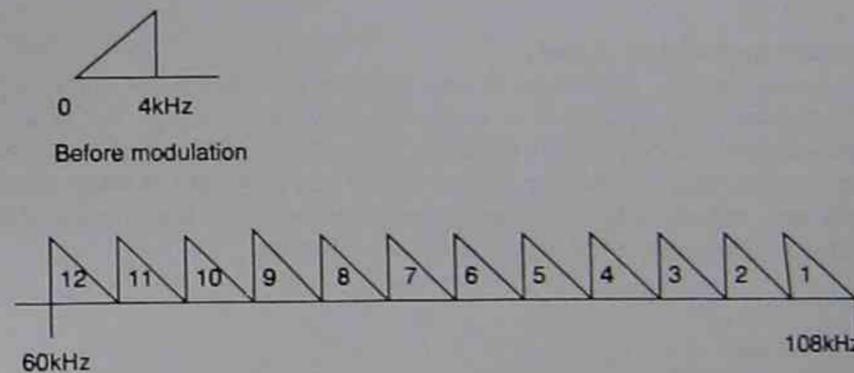
The radio spectrum

In radio communications, different parts of the spectrum are referred to using the following terms:

3kHz - 30kHz	VLF	Very low frequency
30kHz - 300kHz	LF	Low frequency
300kHz - 3MHz	MF	Medium frequency
3Mhz - 30MHz	HF	High frequency
30MHz - 300MHz	VHF	Very high frequency
300MHz - 3GHz	UHF	Ultra high frequency
3Ghz - 30GHz	SHF	Super high frequency

In Australia, AM broadcasting uses the MF band (also referred to as medium wave). Short wave broadcasting uses both MF and HF.

The standard FM broadcast band is 88-108MHz, which is in the VHF band.

Television broadcasting uses both VHF and UHF.

Communications via satellites generally use SHF, but VHF and UHF are used in some applications.

Analog and digital signals

An analog signal is continuous in time and may have any value within a given range. (A baseband analog signal is an analog of a physical quantity. For example, the voltage from a microphone is an analog of the sound pressure. Sound pressure can vary continuously with time). Digital signals can only have certain values (usually only two) and can only change at certain points in time. The terms analog and digital roughly correspond to continuous and discrete.
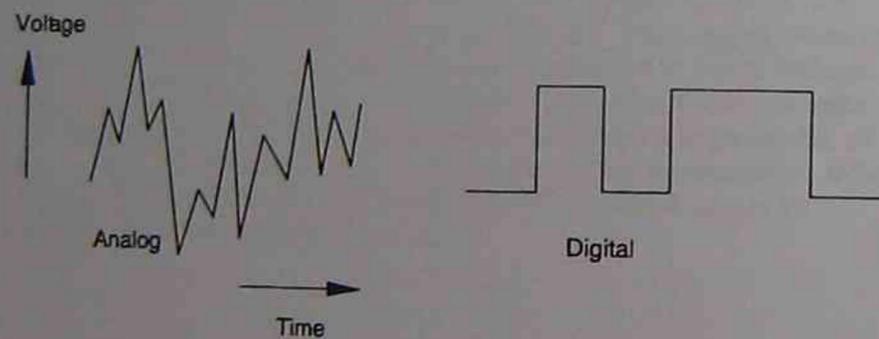
Signals and channels

Both analog and digital signals may be carried on copper wires, radio or optical fibres. However, an analog channel will have amplifiers which cannot handle digital signals. Likewise a digital channel will have repeaters which discriminate between two voltage levels and cannot handle analog signals.

It is possible, however, for analog channels to carry digital information and for digital channels to carry analog information. Thus we have four combinations of signal type and transmission channel type.

(i)   *Analog signal, analog channel*

Example: Speech on a telephone line
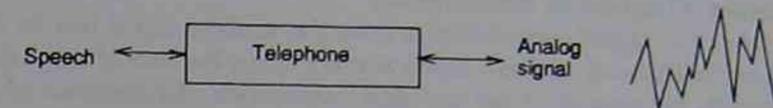
(ii) *Digital signal, analog channel*

Example: Computer data on a telephone line.
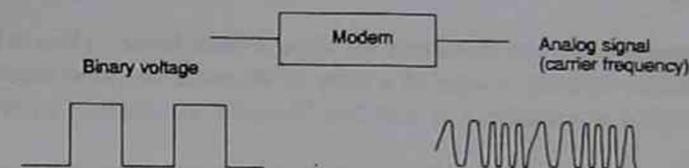
(iii) *Analog signal, digital channel*

Example: Speech or music on a digital channel.
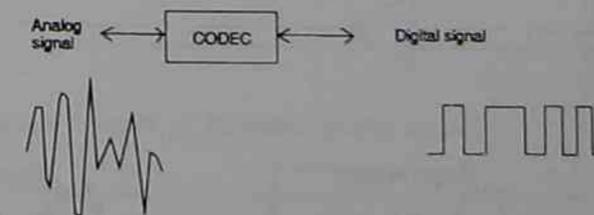
A codec (coder-decoder) takes an analog signal, such as one directly representing speech, and approximates the signal by a bit stream. At the receiving end the bit stream is used to reconstruct the analog signal. You will learn more about this in the next section.

(iv) Digital signal, digital channel

Example: Computer data on a local area network (LAN)
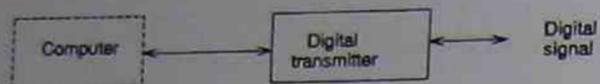
Degradation of signals in transmission

Whether analog or digital transmission is used, the received signal will be different from the original because of noise added at every stage, because of crosstalk from other channels, and because of the non-ideal transmission characteristics of the channel (distortion, limited frequency response non-uniform time delays).

In the case of digital transmission, if repeaters (regenerators) are used before the signal is badly degraded, the error rate can be kept very very small. With digital signals, errors can be detected and even corrected. (This is very important if you are transferring large sums of money electronically!)

In an analog system, noise and distortion are always cumulative. (You will be aware of this if you have ever made a copy of a copy of an audio or video cassette tape recording).

The diagram below illustrates the difference between an amplifier in an analog channel and a regenerator in a digital channel. It also shows how a regenerator can remove noise.
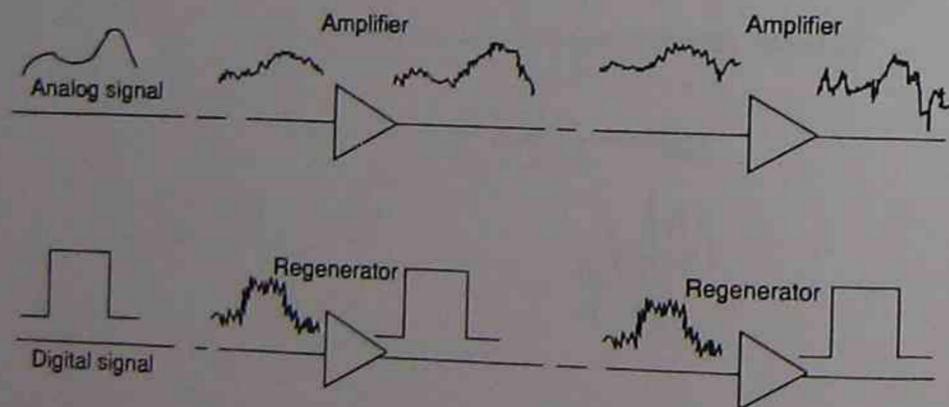
Review questions

These questions will help you revise what you have learnt in Section 4.

1. (a) Draw a block diagram of a communications system employing radio.

    (b) What is meant by 'channel' and how is it different to a 'bearer'?

    _____

    _____

    (c) List three possible source transducers and two possible destination transducers.

    _____

    _____

2. Is a microphone a source of baseband analog signal?

    _____

    _____

3. State which of the following are reasons for using modulation.

    *   To improve the signal to noise ratio
    *   To allow multiplexing
    *   To increase the speed of transmission
    *   To allow use of a more suitable frequency range.

    _____

    _____

4. Give three reasons why a received signal may differ from the transmitted signal.

    _____

    _____

    _____

5. Name the two types of multiplexing.

    _____

6. State the frequency bands for the following:

    (a) TV channel 0 _____

    (b) TV channel 28 _____

    (c) 548MHz _____

    (d) CB Radio channel 40 on 27.405MHz _____

7. Briefly explain what is meant by SHF, including what frequencies it covers and state a major application.

    _____

    _____

    _____

8. What would happen to an analog signal if an attempt was made to transmit it through a digital channel?

    _____

    _____

9. Briefly describe what a modem is and what it does.

    _____

    _____

10. What is the name given to a device which allows analog signals to be transmitted over a digital channel?

    _____

Section 5: Pulse code modulation

SUGGESTED DURATION	PREAMBLE
6 hrs	To introduce you to the sampling theorem and PCM.

This section covers learning outcome 7 of the Module Descriptor.

Objectives

At the end of this section you should be able to:

- [ ] sketch the spectrum of a pulse amplitude modulated signal for each of the following cases:
  - sampling rate greater than 2f(max)
  - sampling rate equal to 2f(max)
  - sampling rate less than 2f(max)

- [ ] use the spectrum of a PCM signal to:
  - deduce the sampling theorem
  - show that the original signal can be recovered with a low pass filter

- [ ] explain why in practice, signals are sampled at a rate higher than that required by the sampling theorem

- [ ] explain why an anti-aliasing filter is necessary

- [ ] use a block diagram to describe the operation of a PCM communication link comprising the following processes or components:
  - filter
  - sampling gate
  - A/D conversion/quantising
  - parallel/serial conversion
  - D/A conversion
  - low pass filter signal recovery.

Introduction

What is PCM?

PCM is a means of communication in which an analog signal is encoded into a digital signal, then transmitted over a transmission medium in a digital format, and finally decoded back into its original analog form.
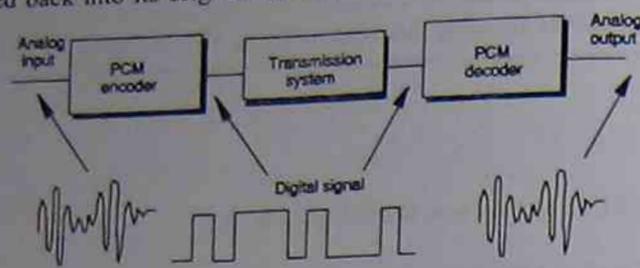
Single channel PCM communication system

The concept of PCM is not new, and is credited to Alec Reeves who conceived the idea whilst working in the Paris laboratory of ITT in 1937. Partial implementation of PCM however, had to wait for the invention of the transistor many years later. The first commercial PCM systems were installed in the early 1960s.

What are the advantages of PCM?

- Being a digital signal, PCM has high immunity to noise and interference.

- PCM signals can be regenerated 'as good as new' at regular intervals along the transmission medium. This ensures that the transmission quality remains high over the length of the path, regardless of the length.

- The present worldwide trend is towards integrated digital networks. PCM is compatible with this trend. Analog signals, once 'digitised', are indistinguishable from data signals, and both such signals may be treated as one.

- PCM allows the capacity of existing transmission systems such as telephone cables to be greatly increased. For example, 30 users can be multiplexed on just 2 twisted pairs of wires.

Are there any disadvantages with PCM?

The primary disadvantage is the large bandwidth required. However, most existing types of transmission media are under-utilised when carrying analog signals, and actually have sufficient bandwidth to carry PCM signals.

The PCM encoding process

The basic PCM encoding process involves the following steps:

- Low-pass filtering
- Sampling
- Analog to Digital Conversion (Quantisation & Encoding)
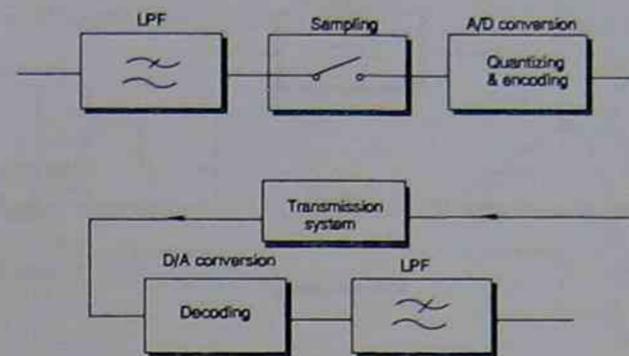
PCM system components

Low-pass filtering

The analog input signal could contain frequencies higher than the capabilities of the PCM system. If such frequencies are not removed prior to sampling, a form of distortion known as 'aliasing' occurs. To avoid this aliasing distortion, a low-pass filter is provided as the first block in most PCM systems.

Low-pass filtering
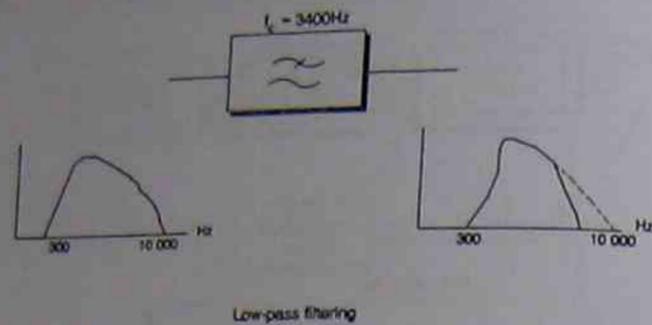
Sampling

When an analog message is sent over an analog communications system, the full message is carried at all times. In a digital communications system, only samples of the message are transmitted at regular intervals. It may seem astonishing that regular samples of a message and not the entire waveform can adequately describe all the information contained within the message.

Sampling - known as
Pulse amplitude modulation (PAM)
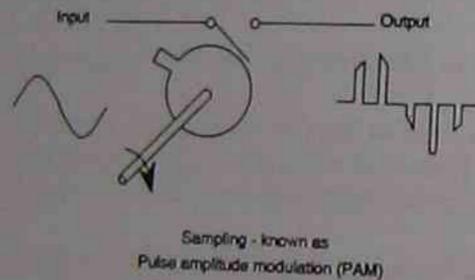
Shannon's Sampling Theorem (1949) states that any band-limited signal with maximum frequency f_{max} Hertz, is uniquely determined by equally spaced samples which occur at least $2f_{max}$ times per second.

Therefore a speech signal which has been band-limited to 300-3400 Hz can theoretically be sampled at 6800 times per second. In practice however, a slightly faster sampling rate (8000Hz) is used, which greatly assists in the demodulation process.

One of the great benefits associated with sampling is that it is possible to utilise the gaps between samples with samples from other message sources. This technique is known as Time Division Multiplexing (TDM).

Time division multiplexing
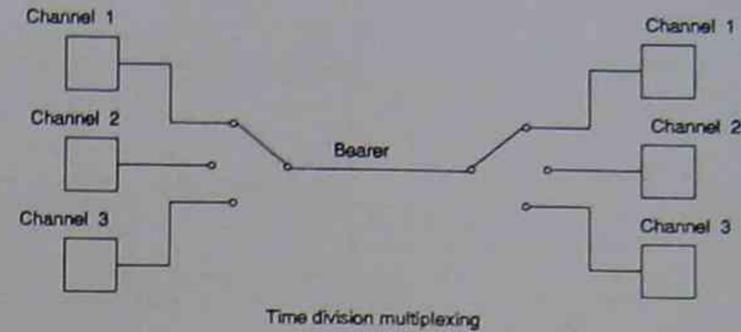
Spectrum of sampled signals

Let's assume that our message signal is speech, and has a frequency spectrum which extends from 0Hz to 3400Hz.

Spectrum of sampled signal
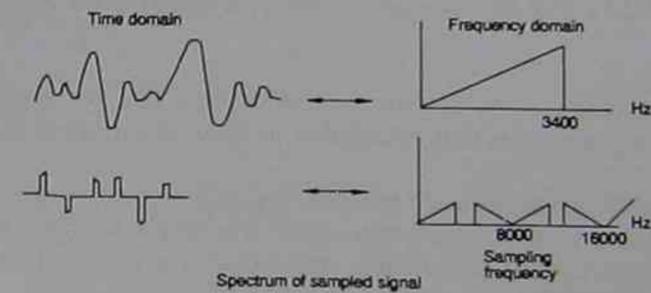
If this message signal is sampled at a rate greater than $2f_{max}$, for example 8000Hz, the resulting sampled signal has a frequency spectrum which consists of upper and lower sidebands around the sampling frequency and its multiples.

Note that there is no overlapping of spectra. At the low frequency end of the spectrum is the original message spectrum, which may be recovered with the aid of a low-pass filter.

74
EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995
75

If the message signal is sampled at exactly $2f_{max}$, the sidebands just touch one another. Recovery of the message without encroaching onto the adjacent sideband is not possible using a practical low-pass filter.

If the message signal is sampled at less the $2f_{max}$, the sidebands overlap, and if the message is recovered with a low-pass filter, 'aliasing distortion' will be present.

$f_s > 2f_{max}$ ... Hz

$f_s = 2f_{max}$ ... Hz

$f_s < 2f_{max}$ ... Hz
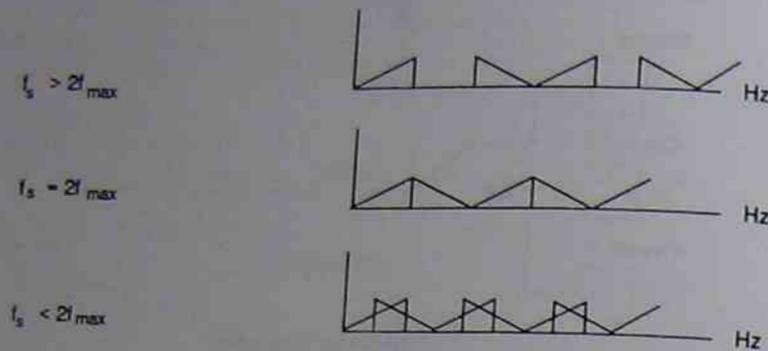
Effect of sampling frequency on spectrum

Analog to digital conversion

The signal that emerges from the sampling gate is known as Pulse Amplitude Modulation (PAM), with the information being contained in the amplitude of the sample pulses.

Little would be gained in transmitting this PAM signal directly, since it is basically still an analog signal, and is thus susceptible to noise and interference.

In PCM, the PAM signal is passed to an analog to digital converter, in which each sample value is represented by a binary code. The amplitude range of the analog to digital convertor is divided into discrete steps, known as quantisation steps, each step having its own unique binary code.

Quantisation is a process of 'rounding off' so that a limited number of levels represents the complete signal.
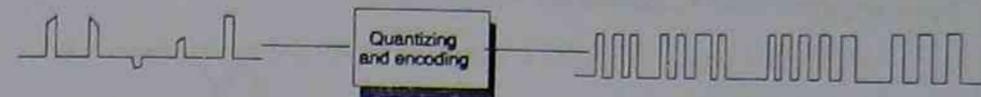
Analog/digital conversion

The difference between the actual sample height and the quantised approximation is called quantisation error. An accumulation of these random errors is called quantisation noise.

The largest error that occurs due to quantising is half of the step size, and the greater the number of steps, the lower the quantisation noise. However, a greater number of steps requires more bits, which in turn increases the data rate, hence bandwidth of the system.

In an n-bit system, there are 2^n quantising steps. Typically, 7 or 8 bits are required for reasonable quality of speech, giving 128 and 256 steps respectively.

The quantising and encoding functions are performed simultaneously by the analog to digital converter. The output is generally in a parallel form, so a stage of parallel to serial conversion completes the basic PCM encoding process.

The PCM decoding process

The serial PCM signal is first converted to a parallel signal. The D to A converter then reconstructs quantised PAM. Finally the message is recovered with a low pass filter.

76   EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995   77
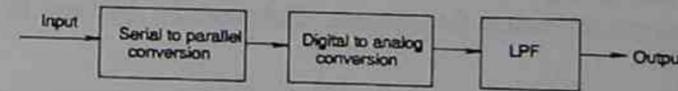
Derivation of PAM spectrum

Pulse amplitude modulation (PAM)
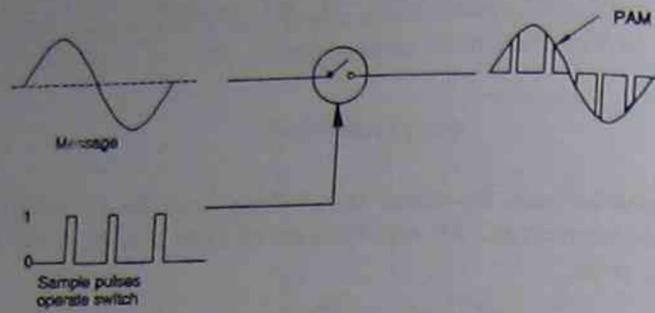
Switch open: Output = Input x 0
Switch closed: Output = Input x 1

∴ PAM = Message x Pulse train

$$= \sin 2\pi f_m t \times [DC + A_1 \sin 2\pi f_s t + ...]$$

Multiplying the message by the DC component leaves the message frequencies intact.

Multiplying the message by each of the other spectral components of the pulse train produces sum and difference frequencies. The resulting spectrum is:
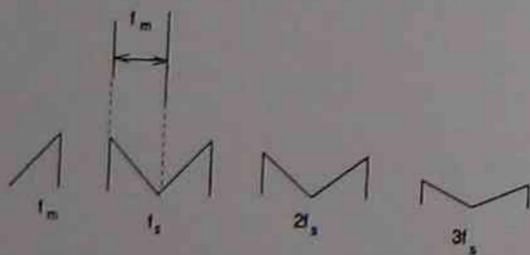
From the above spectrum we see that the message can be recovered with a LPF, provided that $f_s > 2f_m$.

Review questions

These questions will help you revise what you have learnt in Section 5.

1. An audio signal with a frequency range 0-15kHz is sampled at 32kHz.

    (a) Sketch the spectrum of the resulting PAM signal.

    (b) How could the message be recovered from the PAM signal?
    _____

    (c) Is the sampling rate high enough?
    _____

    (d) Suppose the message had a 19kHz whistle superimposed on it (not removed by an anti-aliasing filter). What effect would this have on the recovered message?

    _____

2. State the steps required to convert PAM to PCM.
    _____
    _____

3. A PCM modulator uses an 8 bit A to D converter.

   (a) How many quantisation levels are possible?

   _____

   (b) What is the maximum possible signal to quantisation noise ratio?

   _____

4. List the steps involved in demodulating PCM.

   _____

   _____

   _____

5. A 12 channel TDM PCM telephone system uses 8kHz sampling and 8 bits per sample.

   (a) What is the bit rate of the system?

   _____

   (b) What is the minimum bandwidth required in theory?

   _____

Section 6: Noise

SUGGESTED DURATION	PREAMBLE
4 hrs	To enable you to explain the various classifications of noise and the effects of noise on communication systems.

This section covers learning outcome 8 of the Module Descriptor.

Objectives

At the end of this section you should be able to:

- define and give a typical source for each of the following types of noise:
  - external
  - man-made
  - natural
  - internal
  - thermal
  - white
  - impulse
  - random

- explain why the SNR decreases as an analog signal passes through a communication system

- explain why impulse noise may cause errors in a digital signal yet cause little impairment to an analog signal

- convert SNR to dB and vice versa.

TAFE

80

EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995

85

TAFE

Noise

Introduction

Noise can be defined as interference which degrades the useful information in a signal. Its ultimate effect may be audible (the conventional meaning of noise) from a loudspeaker, visible as 'snow' or other interference on a television screen, or it may result in errors in computer data. When the interference comes from another channel in the system, it is generally called crosstalk.

Noise is usually the most important limiting factor in a communication system.

There are many different sources of noise and we will start by classifying them as external and internal.

External noise

External noise may come from man-made sources or natural sources.

Sources of man-made noise include car ignition systems, switches (eg. in a refrigerator), and brushes in electric motors. Switching noise is worse where the current is large and the load is inductive. In fact all electrical equipment will radiate some noise. In principle, the noise from these sources can be suppressed at the source.

Natural external noise sources include electrical storms and discharges in the atmosphere and ionosphere, and cosmic and solar noise. These noise sources have a greater effect on radio communication systems than on other systems. The effects can be minimised by suitable choices of frequencies and antennas, and by careful positioning of antennas.

Internal noise

This kind of noise is generated naturally in electronic components such as resistors, diodes and transistors. Noise generated in resistors is called thermal noise. It is due to the random movement of electrical charge. Because this movement is thermally generated, the noise power is proportional to absolute temperature. Also, because the movement is random, the noise is uniform across the spectrum and the available noise power is proportional to bandwidth.

The available noise power from a resistor is kTB (watts) and the available noise voltage into an open circuit is $\sqrt{4kTBR}$.

Where $k = 1.38 \times 10^{-23}$
         T = absolute temperature in Kelvin
         B = the bandwidth of measurement in Hz.

Shot noise is generated in semiconductors when electrons cross a potential barrier. The power produced by shot noise is directly proportional to the bias current. Like thermal noise, it is also purely random and its power spectrum is flat with frequency.

Flicker noise is another type of random noise occurring in resistors and semiconductors. Above a few kHz its power spectrum is essentially flat, but at lower frequencies it increases. For this reason it is also called 1/f noise. Metal film resistors produce less flicker noise than carbon resistors.

Internal noise may also be produced from dirty switch contacts, dry solder joints, other poor connections and faulty components. Ideally these sources of noise can be eliminated.

White noise, as we have already said, is noise which has uniform power across the spectrum, measured in W/Hz.

Impulse noise is noise which occurs in short sharp bursts, such as noise produced by lightning or from switching a motor off. Measures can be taken to reduce the effect of impulse noise in some equipment (eg. TV receivers). Because impulse noise may have a large amplitude, it can easily corrupt digital signals, causing a '1' to be interpreted as a '0' or vice versa. Impulse noise may be annoying when you are listening on the telephone (though it is usually too short) but it is not likely to reduce intelligibility.
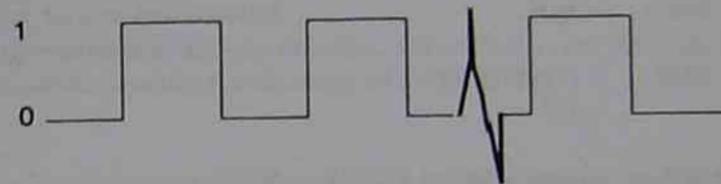
This impulse may cause the zero to be interpreted as a one.

Random noise is noise which is unpredictable. Examples of non-random noise are 50Hz hum induced into the system from the mains, and hum produced by power supply ripple.

Effect of noise on an analog signal

Generally when noise is added to an analog signal it becomes indistinguishable from the signal. It is therefore not possible to remove it. (Two exceptions here are certain types of impulse noise and noise which can be removed by a filter because it is outside the frequency range of interest.) Because noise is added at every stage of an analog system, the further the signal travels, the worse the signal-to-noise ratio becomes.

86   EA190 Electronic Signals and Systems
Student Workbook
June 1995

EA190 Electronic Signals and Systems
Student Workbook
June 1995   87

(We showed in Section 4 how digital regenerators can remove noise from a digital signal).

Signal-to-noise ratio (SNR) is the ratio of signal power/noise power.

To convert this dB we take $10\log_{10}$(SNR).

Example: Signal power = 1mW
          Noise power = 1nW

          SNR = 1mW/1nW
               = 1,000,000 or 10^6

          $10\log 10^6$ = 60dB

If we know voltages rather than power, SNR = voltage ratio squared (this is because power is proportional to voltage squared).

Example:
          Signal = 1mV
          Noise = 1μV

          SNR = $(1mV/1μV)^2$
               = 10^6

To convert to dB we still take 10 log (SNR) → 60dB

or we can take 20 log (1mV/1μV) and get the same result.

ESSnoise.doc

Electrical Noise and Distortion

Noise - unwanted electrical energy present in the passband of a communication system.

Correlated noise - produced as a result of the signal and related to the signal. Generally called distortion, and further divided into these categories: harmonic distortion, intermodulation distortion, transient intermodulation distortion and quantisation noise (although this is generally thought of as being uncorrelated).

Uncorrelated noise - independent of the signal and present during the absence of the signal (unless noise gates are used). Examples are: phone line noise, amplifier hum and tape hiss.

A.   External Noise - produced outside the communications system and entering into it in the passband.

(i)   Man made.
- Electric field produced from voltage which is in close proximity. May be shielded with grounded conductive screen.
- Magnetic field produced from current flow in close proximity. May be shielded with low reluctance screen, ie. iron or mu-metal.
- Spark producing mechanisms such as switches, commutators, car ignitions and fluorescent lamp starters. Impulses produce a wide range of frequencies.

(ii)   Natural
- Atmospheric such as lightening and air ionisation (aurora borealis), called "static".
  Frequency response:

f(lin)
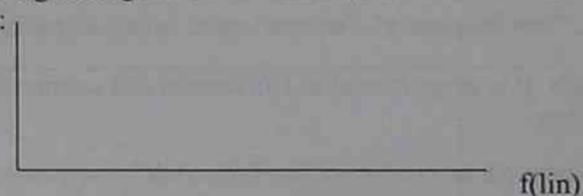
- Extraterrestrial, "deep space noise" such as caused by solar noise from the sun's reactions (low intensity) or solar flares and sunspots (high intensity).
  The sun has an 11year cycle within a 99 year cycle.
- Cosmic noise, "galactic" or "black body noise", emanates from the stars and dark matter between the stars. It is evenly distributed across the sky.

B.   Internal Noise - produced within components in the communications system.

- Thermal noise, "Brownian", "random", "Johnson", "resistance", "white" noise.
  Caused by the random motion of electrons in conductors moving between different energy levels. Proportional to temperature and frequency.
    Frequency response:

f(lin)   f(log)
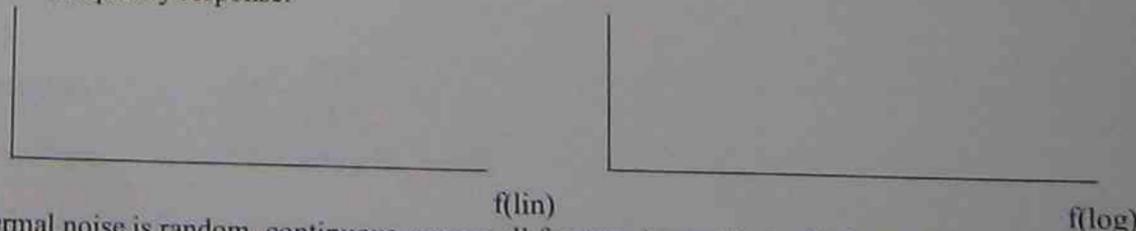
Thermal noise is random, continuous, covers all frequencies and is in all electronic devices. It is the most important source of noise.

1

Noise power: $N = kTB$, k = Boltzmann's constant (1.38×10^{-23})
T = absolute temperature ($0^0K = -273^0C$)
B = bandwidth of system

V_N is noise voltage produced by
equivalent resistance of the system.

$$N = \frac{V_N^2}{R_N}$$

$R_L = R_N$ for maximum transfer of power
ie. power to load is $N = kTB$

Output voltage $V_O = \frac{V_N}{2} = \sqrt{kTBR_L}$

$$V_N = \sqrt{4kTBR_N}$$

- Shot noise, "transistor noise", appears in semiconductors and valves and is caused by the different path lengths of the charge carriers due to their random motion. It is proportional to bias current through the device and to bandwidth.
  It has a "white "noise spectrum.
  Frequency response:

  f(lin)

- Flicker noise, "$\frac{1}{f}$", "low frequency", "excess" noise is caused by surface defects in crystal structures of materials. It is proportional to DC current and temperature.
  Frequency response:

  f(lin)
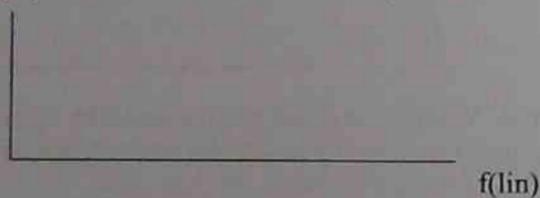
- Other less significant noises:
  - Transit noise,   low frequency
  - Burst noise (popcorn),   low frequency ($1/f^2$)
  - Resistance noise in semiconductors
  - Switching noise from temporary open circuits

Note: correlated noise (distortion) can be reduced by negative feedback, as used in opamp circuits
      Uncorrelated noise cannot be reduced by NFB due to its randomness.

Noise spectrum.

White noise: has equal power across the frequency spectrum.
               sounds like hiss, eg. an FM receiver tuned off station.
Pink noise: has equal power per frequency octave or decade.
               used for audio testing.

f(lin)

f(log)
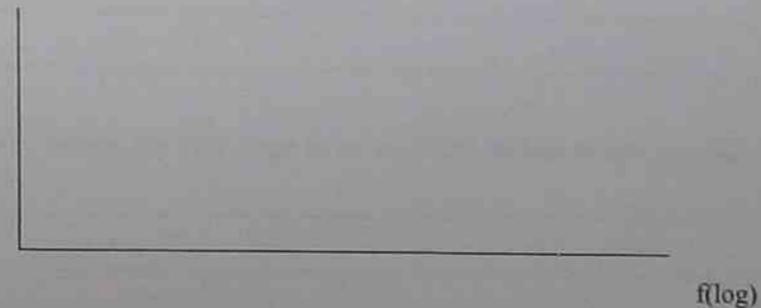
Effects of noise.

Analogue systems: becomes part of the signal and therefore usually impossible to remove or reduce without affecting the signal.
Digital systems: causes bit errors, but may be detected and corrected, although if too major there can be catastrophic loss of signal.

Signal to Noise Ratio

Power ratio: $SNR(dB) = 10\log_{10}\dfrac{signal\,power}{noise\,power}$

Voltage ratio: $SNR(dB) = 20\log_{10}\dfrac{signal\,voltage}{noise\,voltage}$

This is probably the most important parameter in evaluating a communications system or an audio system.

Noise Ratio $NR = \dfrac{SNR_i}{SNR_o} > 1$

Noise Figure $NF = 10\log NR > 0dB$

The lower these figures are the better especially at low level input stages.

Review questions

These questions will help you revise what you have learnt in Section 6.

1. State two sources of man-made external noise.

    _____

    _____

2. State two sources of natural external noise.

    _____

    _____

3. Name three types of internal noise generated in components.

    _____

    _____

    _____

4. Why should the first stage of an amplifier operate at low current?

    _____

    _____

5. Why are metal film resistors preferred over carbon resistors in the first stage of
    an amplifier?

    _____

6. Is 1/f noise an example of white noise? Explain your answer.

    _____

    _____

    _____

7. Which will have greater effect on an analog system - impulse noise or white
    noise?

    _____

8. Which will have the greater effect on a digital system - impulse noise or white noise?

_____

9. A microphone has 100 Ohms internal resistance.

    (a) Calculate the open circuit thermal noise voltage generated over a 20kHz bandwidth at 27°C.

    (b) If the microphone delivers 2mV of signal, what is the signal-to-thermal noise ratio in dB?

10. An amplifier is rated to deliver 100W to a loudspeaker with a signal-to-noise ratio of 80dB. What noise power does it produce?

ELECTRONIC SIGNALS AND SYSTEMS

Practical 1

This consists of three demonstration labs set up for you. There will be limited hands-on opportunity.
A report of these labs will be written and handed in.
Included will be:
    -description of equipment.
    -aim of lab.
    -diagrams.
    -discussion of findings and observations.

Lab A.   Spectrum of signals.
    (i)   sine
    (ii)   square
    (iii)   white noise
    (iv)   pink noise
-sketch the waveforms.
-describe the aural properties.
-describe the difference in sound between sine and square waves.
-why do sine and square waves above about 6kHz sound the same?

Lab B   Spectrum of radio transmissions
-sketch the spectrum of a number of radio stations.
-sketch the spectrum of a single radio station.
-record relevant frequencies.
-list frequency ranges of AM and FM radio broadcast stations and of the Sydney TV channels.

Lab C   Fourier synthesis of waveforms
-observe the construction of a square wave from its harmonics.
-sketch all waveforms.
-is the output a "good" square wave?
-how can the output wave shape be improved?

The Electromagnetic Spectrum

(labels: MAINS POWER · OMEGA NAVIGATION · RADIO NAVIGATION BEACON · MF-AM BROADCAST BAND · 27MHz CB/MARINE RADIO · VHF TV · VHF FM BROADCAST BAND · UHF MARINE RADIO · UHF TV · MOBILE PHONES · GPS NAVIGATION · DOMESTIC MICROWAVE OVENS)

	ELF	VOICE	VLF	LF	MF	HF	VHF	UHF	SHF	EHF		INFRARED	VISABLE	UV
FREQUENCY	30Hz	300Hz	3kHz	30kHz	300kHz	3MHz	30MHz	300MHz	3GHz	30GHz	300GHz	3THz	30THz	300THz
WAVELENGTH	10Mm	1Mm	100Km	10Km	1Km	100m	10m	1m	10cm	1cm	1mm	100µm	10µm	1µm

PREFIXES

- μ = micro 10^{-6}
- m = milli 10^{-3}
- c = centi 10^{-2}
- k = kilo 10^{3}
- M = mega 10^{6}
- G = giga 10^{9}
- T = tera 10^{12}

Australian MF-AM Broadcast Standards

Frequency Range:	526.5-1606.5kHz
Channel Frequencies:	531, 540, 549 ... 1593, 1602 (Note: Some special services are located slightly above this band. Receivers may need modifications.)
Stereo Signal:	Compatible quadrature amplitude modulation (C-Quam) 25Hz +/-0.1 Hz with a dev. of 3% to 5% of the quadrature ch
Frequency Response:	within 2dB (50Hz-7.5kHz)
THD:	Not exceding 4% at 80% modulation (400Hz to 5kHz)
Stereo Separation:	18dB min.

Australian VHF-FM Broadcast Standards

Frequency Range:	88-108MHz
Channel Frequencies:	88.1, 88.3, ... 107.7, 107.9
Deviation:	75kHz
Pre-emphasis/De-emphasis:	50 microseconds
Stereo Channel Subcarrier Frequency:	
Pilot Frequency:	19kHz

Notes:
1. Ancillary Communications Services may be included in the main program channel. Subcarrier Frequency: 67kHz; Maximum Deviation: 7.5kHz; Modulation: FSK or FM; Pre-emphasis: up to 150us
2. The frequencies 87.6, 87.7, 87.8, 87.9 & 88MHz have been released for use by certain Low Power Open Narrowcasting Services, such as Tourist Radio.

Australian Colour TV Broadcast Standards

Channel Width:	7MHz vestigial sideband trans.
Vision Carrier:	1.25MHz above lower frequency edge of channel
Primary Sound Carrier:	5.5MHz above vision carrier
Secondary Sound Carrier:	242.1875kHz above primary sound carrier
Vision Modulation:	Negative amplitude modulation
Lines per Picture:	625 lines, interlaced 2:1
Line Frequency:	15625Hz
Field Frequency:	50Hz
Colour Subcarrier Frequency:	4.43361875MHz
Aspect Ratio:	4:3
Horizontal Sync Pulse Amplitude:	100% carrier amplitude
Front Porch:	1.3 - 1.8us
Back Porch:	
Line Blanking Interval:	Line blanking interval - (front porch + sync pulse)
Black and Blanking Level:	11.8-12.3us
White Level:	76% peak carrier level
Pre-equalisation Pulse Interval:	20% peak carrier level
Post-equalisation Pulse Interval:	2.5H (H-line period, 64 us)
Field Sync Pulse Interval:	2.5H
Field Blanking Interval:	2.5 +12us
Ratio peak vision to peak sound:	Single sound carrier systems 10dB; Dual sound carrier systems Ch1 10dB, Ch2 20dB
Sound Modulation:	FM
Sound Deviation:	+-50kHz
Audio Pre-emphasis:	50u

Australian TV Channels

Ch	Vision Carrier (MHz)	Primary Sound Carrier (MHz)	Ch	Vision Carrier (MHz)	Primary Sound Carrier (MHz)
0	46.25	51.75	44	639.25	644.75
1	57.25	62.75	45	646.25	651.75
2	64.25	69.75	46	653.25	658.75
3	86.25	91.75	47	660.25	665.75
4	95.25	100.75	48	667.25	672.75
5	102.25	107.75	49	674.25	679.75
5A	138.25	143.75	50	681.25	686.75
6	175.25	180.75	51	688.25	693.75
7	182.25	187.75	52	695.25	700.75
8	189.25	194.75	53	702.25	707.75
9	196.25	201.75	54	709.25	714.75
10	209.25	214.75	55	716.25	721.75
11	216.25	221.75	56	723.25	728.75
28	527.25	532.75	57	730.25	735.75
29	534.25	539.75	58	737.25	742.75
30	541.25	546.75	59	744.25	749.75
31	548.25	553.75	60	751.25	756.75
32	555.25	560.75	61	758.25	763.75
33	562.25	567.75	62	765.25	770.75
34	569.25	574.75	63	772.25	777.75
35	576.25	581.75	64	779.25	784.75
39	604.25	609.75	65	786.25	791.75
40	611.25	616.75	66	793.25	798.75
41	618.25	623.75	67	800.25	805.75
42	625.25	630.75	68	807.25	812.75
43	632.25	637.75	69	814.25	819.75

Ch 0-2 VHF Band I · Ch 3-5 VHF Band II · Ch 5A-11 VHF Band III · Ch 28-35 UHF Band IV · Ch 39-69 UHF Band V

VHF Marine Frequencies

Ch	Freq Ship (MHz)	Shore (MHz)	Ch	Freq Ship (MHz)	Shore (MHz)
1	156.050	160.650	60	156.025	160.625
2	156.100	160.700	61	156.075	160.675
3	156.150	160.750	62	156.125	160.725
4	156.200	160.800	63	156.175	160.775
5	156.250	160.850	64	156.225	160.825
6	156.300	156.300	65	156.275	160.875
7	156.350	160.950	66	156.325	160.925
8	156.400	156.400	67	156.375	156.375
9	156.450	156.450	68	156.425	156.425
10	156.500	156.500	69	156.475	156.475
11	156.550	156.550	70	156.525	156.525
12	156.600	156.600	71	156.575	156.575
13	156.650	156.650	72	156.625	156.625
14	156.700	156.700	73	156.675	156.675
15	156.750	156.750	74	156.725	156.725
16	156.800	156.800	77	156.875	156.875
17	156.850	156.850	78	156.925	161.525
18	156.900	161.500	79	156.975	161.575
19	156.950	161.550	80	157.025	161.625
20	157.000	161.600	81	157.075	161.675
21	157.050	161.650	82	157.125	161.725
22	157.100	161.700	83	157.175	161.775
23	157.150	161.750	84	157.225	161.825
24	157.200	161.800	85	157.275	161.875
25	157.250	161.850	86	157.325	161.925
26	157.300	161.900	87	157.375	161.975
27	157.350	161.950	87A	157.375	157.375
28	157.5000	162.000	88	157.425	162.025

Ch	Use		Ch	Use
6	SAR, Port Operations, Commercial		72	Yachts, Commercial, Professional Fishing, Port Operations
8	Port Operations, Commercial		73	Non-Commercial, Yachts, etc.
9,14	Port Operations, Commercial		74	Commercial
16	Distress and Safety		77	Yachts, Professional Fishing
20	Port Operations		78	Commercial
23-28	Public Correspondence (Seaphone)		67	Public Correspondence (Seaphone)
67	Distress and Safety		87A	Withdrawn 21-3-92
68	Port Operations			
80	Professional Fishing			

NZ VHF TV Channels

Ch	Vision (MHz)	Sound (MHz)	Ch	Vision (MHz)
1	45.25	50.75	7	196.25
2	55.25	60.75	8	203.25
3	62.25	67.75	9	210.25
4	175.25	180.75	10	217.25
5	182.25	187.75	11	224.25
6	189.25	194.75	12	231.25

Australian & NZ CB Radio Channels

Ch	Aust HF (AM) (MHz)	Aust UHF (MHz)	NZ HF (MHz)
1	26.965	476.425	26.330
2	26.975	476.450	26.340
3	26.985	476.475	26.350
4	27.005	476.500	26.360
5	27.015	476.525	26.370
6	27.025	476.550	26.380
7	27.035	476.575	26.390
8	27.055	476.600	26.400
9	27.065	476.625	26.410
10	27.075	476.650	26.420
11	27.085	476.675	26.430
12	27.105	476.700	26.440
13	27.115	476.725	26.450
14	27.125	476.750	26.460
15	27.135	476.775	26.470
16	27.155	476.800	26.480
17	27.165	476.825	26.490
18	27.175	476.850	26.500
19	27.185	476.875	26.510
20	27.205	476.900	26.520
21	27.215	476.925	26.530
22	27.225	476.950	26.540
23	27.255	476.975	26.570
24	27.235	477.000	26.550
25	27.245	477.025	26.560
26	27.265	477.050	26.610
27	27.275	477.075	26.620
28	27.285	477.100	26.640
29	27.295	477.125	26.660
30	27.305	477.150	26.670
31	27.315	477.175	26.680
32	27.325	477.200	26.690
33	27.335	477.225	26.700
34	27.345	477.250	26.710
35	27.355	477.275	26.720
36	27.365	477.300	26.730
37	27.375	477.325	26.740
38	27.385	477.350	26.750
39	27.395	477.375	26.760
40	27.405	477.400	26.770

Emergency Channels		Calling Channels	
Aust HF (AM)	Ch. 9	Aust HF (AM)	Ch.11
Aust UHF	Ch. 5/35	Aust HF (SSB)	Ch.16
		NZ HF (AM)	Ch.11
NZ	Ch.15	NZ HF (AM)	Ch.15

Notes: NZ Channels 1-24 are for AM use only. NZ Channels 25-40 are for SSB use only.

Australian 27MHz Marine Frequencies

Freq (MHz)	Use
27.680	Commercial organisations. Calling and working ship-ship and ship-shore.
27.720	Professional fishing. Calling and working ship-ship and ship-shore.
27.820	Professional fishing. Calling and working ship-ship and ship-shore.
27.860	Non-Commercial. Yachts. etc.
27.880	Distress, safety and calling, supplementary to 27.88
27.900	Distress, safety and calling
27.910	Non Commercial organisations. Calling and working ship-shore.
27.940	Non Commercial organisations. Calling and working for club events, ship-shore and ship-ship
27.960	Non-Commercial organisations. Calling and working (Seaphone)
27.980	Recognised rescue organisations, e.g. Surf rescue. Calling and working. Ship-ship and ship-shore.


(Right page — handwritten lab notes)

Equipment: Comms Trainer
Filter CM10
Function Gen. TG550
AC mV meter
4 x BNC/BNC
2 x BNC T

**ESS Lab No2**

AMLITUDE AND PHASE RESPONSE OF A FILTER

AIM:
1) To plot the AMPLITUDE and PHASE RESPONSE of a BANDPASS FILTER.
2) To observe the effect of a filter on square and triangular wave inputs.

EQUIPMENT: Comms Trainer, Bandpass Filter CM10, CRO, Signal Generator, ac Millivoltmeter, Adaptors and Leads, Frequency Counter. log/lin paper

PROCEDURE:

(block diagram: Signal Gen → Counter, CRO①, CM10 → CRO②, mV meter)
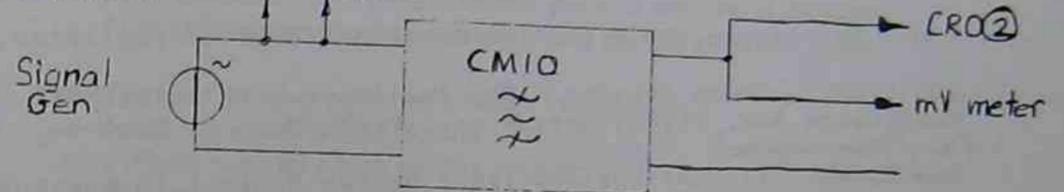
1) Connect up the equipment as above.
2) Set the Bandpass Filter for a lower cut-off frequency of 300Hz and an upper cut-off frequency of 3000Hz.
3) Switch on and vary the input frequency to achieve maximum filter output. (watch for clipping)
4) Adjust the signal level from the generator to give an output of 0dBm at the frequency found in 3) above.
5) Vary the generator input frequency in steps from 50Hz to 15kHz, recording the amplitude of the filter output (mV meter) and phase difference between input and output (use CRO and note lead/lag).
6) Plot the above on log/linear graph paper as shown below.

(graph: Output dB / Phase Angle Lead / 0° / Lag / f(Hz) log scale)
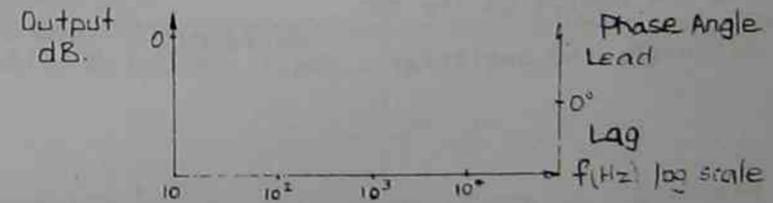
7) Show your graph to the Teacher.
8) Change input to 500Hz Triangular wave and sketch the filter output for filter frequency settings of
(a) 300/3000Hz and (b) 100/10,000Hz.
9) REPEAT 8) above for a Squarewave input.
10) Explain the effect of the filter on the waveforms of 8) and 9).

CONCLUSIONS:

Draw bode plot on your waveform.
Specify filter: f -3dB 's
tc's
roll-off rates
filter type and order

MEASURING AMPLIFIER SNR

AIM: To measure the Signal to Noise Ratio of an amplifier.

EQUIPMENT: Comms Trainer, CM2 Amplifier, mV meter, CRO, Decade Resistance Box, Signal Generator. 2 x BNc/BNc, 2 x BNc/4mm, BNc T piece (BNc/4mm adaptor)

PROCEDURE: 1) Set the amplifier volume control to maximum and leave it set in this position for the remainder of the lab. Switch the amplifier output to the internal load.

2) Set Vs to produce maximum undistorted output from the amplifier, using the CRO to monitor the output. Measure output level with the mV meter.

Max.O/P = ... V.rms = ... dBV

3) Place a decade box in series with the amplifier's input and adjust it to halve the amplifier's output. The setting on the decade box will equal the amplifier's input resistance, Ri.

Ri = ...

4) Connect a resistance equal to Ri across the amplifier's input terminals and measure the amplifier's output with the mV meter. Observe the output on the CRO.

Noise O/P = ... V.rms = ... dBV

5) Calculate the amplifier's SNR.

SNR = ... dB.

6) Repeat 2), 3) and 5) above, using the dB scale on the mV meter. Ask Teacher if not familar with this method.

SNR = ... dB.

CONCLUSIONS: What errors are involved in the SNR measurement as carried out above? Are any of these significant? Explain.

ANSWERS TO REVIEW QUESTIONS

Section 1

1.

2. ■ In an electrical power transfer system, efficiency is the prime consideration. In an information transfer system, waveform integrity and signal-to-noise ratio are the prime considerations.

   ■ In an electrical power transfer system, power is provided at the source. In an information transfer system, power is provide at various points.

3. • The laser tracking system in a compact disk player
   • an agc system in a radio receiver
   • a crystal oven for a high stability oscillator
   • The pressure controller in a steel rolling mill

   Note that a remote controller for a model aeroplane transmits but does not receive. Therefore there can be no feedback to the controller itself, though there would be feedback in the plane.

4. The output voltage is sampled by a resistive divider. The sample is compared with the source and the difference (error signal) is amplified.

5. ■ Open loop control: The motor speed will greatly depend on the motor load and supply voltage, as well as on the controller setting.

   ■ Closed loop control: The effect of load and supply voltage will be greatly reduced by the feedback.

ESS Lab No 3

MEASURING AMPLIFIER SNR

AIM: To measure the Signal to Noise Ratio of an amplifier.

EQUIPMENT: Comms Trainer, CM2 Amplifier, mV meter, CRO, Decade Resistance Box, Signal Generator. 2x BNc/BNc, 2x Bnc/4mm, BNc T piece (BNc/4mm adaptor)

PROCEDURE: 1) Set the amplifier volume control to maximum and leave it set in this position for the remainder of the lab. Switch the amplifier output to the internal load.

2) Set Vs to produce maximum undistorted output from the amplifier, using the CRO to monitor the output. Measure output level with the mV meter.

Max.O/P = ... V.rms = ... dBV

3) Place a decade box in series with the amplifier's input and adjust it to halve the amplifier's output. The setting on the decade box will equal the amplifier's input resistance, Ri.

Ri = ...

4) Connect a resistance equal to Ri across the amplifier's input terminals and measure the amplifier's output with the mV meter. Observe the output on the CRO.

Noise O/P = ... V.rms = ... dBV

5) Calculate the amplifier's SNR.

SNR = ...dB.

6) Repeat 2), 3) and 5) above, using the dB scale on the mV meter. Ask Teacher if not familar with this method.

SNR = ...dB.

CONCLUSIONS: What errors are involved in the SNR measurement as carried out above? Are any of these significant? Explain.

ANSWERS TO REVIEW QUESTIONS

Section 1

1.

2. ■ In an electrical power transfer system, efficiency is the prime consideration. In an information transfer system, waveform integrity and signal-to-noise ratio are the prime considerations.

   ■ In an electrical power transfer system, power is provided at the source. In an information transfer system, power is provide at various points.

3. • The laser tracking system in a compact disk player
   • an agc system in a radio receiver
   • a crystal oven for a high stability oscillator
   • The pressure controller in a steel rolling mill

   Note that a remote controller for a model aeroplane transmits but does not receive. Therefore there can be no feedback to the controller itself, though there would be feedback in the plane.

4. The output voltage is sampled by a resistive divider. The sample is compared with the source and the difference (error signal) is amplified.

5. ■ Open loop control: The motor speed will greatly depend on the motor load and supply voltage, as well as on the controller setting.

   ■ Closed loop control: The effect of load and supply voltage will be greatly reduced by the feedback.

Section 2

1. (a) 1kHz, 2kHz, 3kHz
   (b) 1kHz, 3kHz, 5kHz
   (c) 100kHz, 300kHz, 500kHz
   (d) 1MHz, 2MHz, 3MHz
   (e) 1MHz, 3MHz, 5MHz
   (f) 50Hz, 100Hz, 200Hz
   (g) 100Hz, 200Hz, 300Hz.

2. No

3. (a) 2V
   (b) 2:3 or 0.667

   (c)
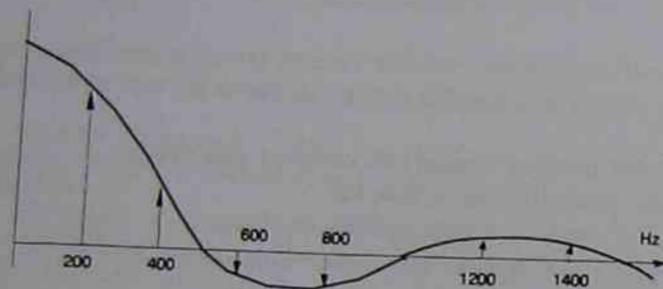
   (d) 5th, 10th 15th etc.

4. (a) (i) The 5th harmonic falls at the first zero of the curve.
       (ii) The 50th harmonic falls at the first zero of the curve.
       (iii) The 500th harmonic falls at the first zero of the curve.
       (iv) The 5 millionth harmonic falls at the first zero of the curve.
       (v) The spectrum is a continuum.

   (b) The spectral frequencies remain the same but more components fall in the first lobe of the sinx/x curve. That is, there are more harmonics with significant amplitude. Therefore, the signal requires more bandwidth.

   (c) The frequency corresponding to the first zero of the sinx/x curve remains unchanged. Therefore, the required bandwidth remains unchanged, even though there are more harmonics in the first lobe.

5. 30Hz - 15kHz (or perhaps 20kHz for young people with excellent hearing).

6. Noise with equal power per unit of bandwidth.

7.
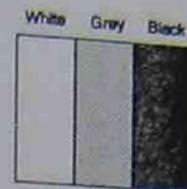
8.
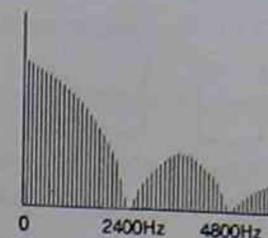
9. 2nd order: ▪ 100Hz, 14kHz (Harmonics)
              ▪ 6.95kHz, 7.05kHz (Intermod products)

   3rd order: ▪ 150hz, 21kHz (Harmonics)
              ▪ 6.9kHz, 7.1khz, 13.95kHz, 14.05kHz

10. 98, 99, 100, 102, 103 and 104MHz

Section 3

1.
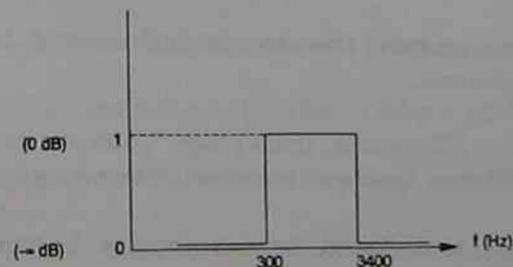


2. 20°

3. Band stop filter. (Stop band = 100 - 200 kHz).

4. As the 3dB bandwidth.

5. (a) Band pass
   (b) 1dB
   (c) 4kHz (148 - 152kHz)

6. 6dB/octave.

7. ▪ loss of colour
   ▪ loss of definition

8.

Rounded edges
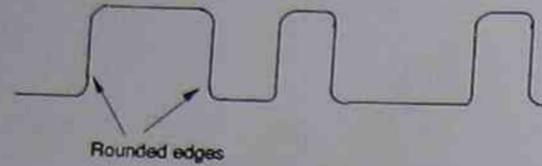
9. Both the phase/frequency response and the amplitude/frequency response affect the shape of the waveform. The shape of the waveform is very important for digital signals.

10. ▪ to select a particular signal
    ▪ to reject unwanted signals and noise

Section 4

1. (a)

Source — Transducer — Transmitter — Channel — Receiver — Transducer — Destination
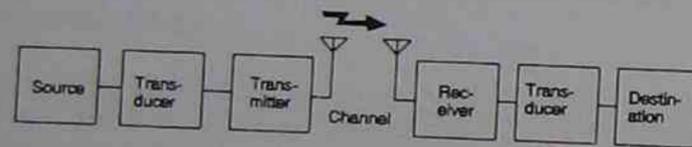
   (b) A channel is a means of one-way communication. A bearer may carry a number of channels.

   (c) ▪ Source: Microphone, strain gauge, video camera
       ▪ Destination: Loudspeaker, printer, video monitor

2. No, the source would be the sound, the microphone is the transducer.

3. • To allow multiplexing
   • To allow use of a more suitable frequency range.

4. Any three of the following:
   ▪ noise
   ▪ crosstalk
   ▪ limited frequency response
   ▪ delay distortion
   ▪ non-linearities.

5. FDM and TDM

6. (a) VHF
   (b) UHF
   (c) UHF
   (d) HF

7. ▪ Super High Frequency
   ▪ 3GHz - 30GHz
   ▪ used for satellite communications

8. Its amplitude would be restricted to two levels.

9. A modem is a modulator/demodulator which converts a digital signal to a form suitable for transmission through an analog channel and reconverts to digital.

10. Codec.

Section 5

1. (a)

0   15 17   32   47 49   64
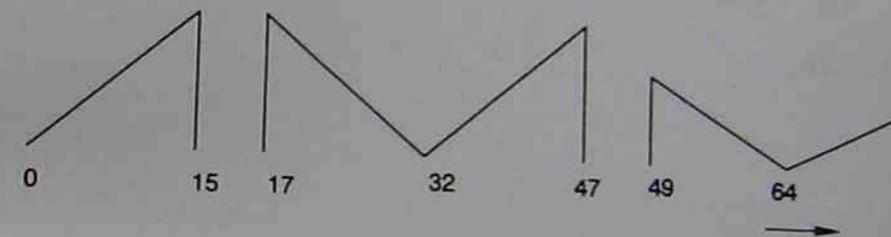
   (b) With a 15kHz LPF.

   (c) Yes, provided fairly sharp cut-off filters are used.

   (d) An alias would occur at 32kHz - 19kHz = 13kHz.

2. Analog to digital conversion (quantisation and encoding) and parallel to serial conversion.

3. (a) 256
   (b) 49.8dB

4. ▪ Serial to parallel conversion
   ▪ Digital to analog conversion
   ▪ Low-pass filtering

5. (a) 768 kbps
   (b) 384kHz

Section 6

1. Motor brushes, relays (virtually anything electrical!).

2. Lightning and cosmic radiation.

3. Thermal, shot and flicker.

4. To reduce shot noise. (Noise produced in the first stage of an amplifier is amplified by all the other stages.)

5. To minimise flicker noise.

6. No, it has more power at lower frequencies whereas white noise has uniform power over the spectrum.

7. White noise.

8. Impulse noise.

9. (a)   182nV.
   (b)   81dB.

10. 1uW.