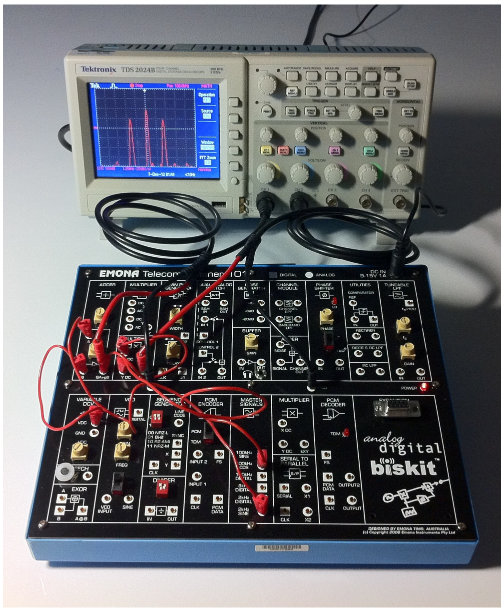
T8

Modulation and Demodulation

In this lab, the EMONA Telecoms-Trainer 101 (ETT-101) is used to demonstrate various analogue and digital modulation techniques. More specifically, the lab covers Amplitude Modulation (AM), Quadrature Amplitude Modulation (QAM), Frequency Modulation (FM), On-Off Keying (OOK), Binary Phase Shift Keying (BPSK), Quaternary Phase Shift Keying (QPSK) and Frequency Shift Keying (FSK). Both non-coherent and coherent demodulation techniques are considered.



Schedule

Preparation time	:	3 hours
Lab time	:	3 hours

Items provided

Tools	:	None
Components	:	None
Equipment	:	Oscilloscope, EMONA Telecoms-Trainer 101 (ETT-101)
Software	:	Tektronix OpenChoice Desktop

Items to bring

Essentials. A full list is available on the Laboratory website at https://secure.ecs.soton.ac.uk/notes/ellabs/databook/essentials/

Before you come to the lab, it is essential that you read through this document and complete ***all*** of the preparation work in section 2. If possible, prepare for the lab with your usual lab partner. Only preparation which is recorded in your laboratory logbook will contribute towards your mark for this exercise. There is no objection to several students working together on preparation, as long as all understand the results of that work. Before starting your preparation, read through all sections of these notes so that you are fully aware of what you will have to do in the lab.

> **Academic Integrity** – *If you undertake the preparation jointly with other students, it is important that you acknowledge this fact in your logbook. Similarly, you may want to use sources from the internet or books to help answer some of the questions. Again, record any sources in your logbook.*

Revision History

December 6, 2012	Rob Maunder (rm)	First version of this lab created

1 Aims, Learning Outcomes and Outline

This laboratory exercise aims for you to:

- investigate various modulation and demodulation techniques;
- experience the real-time transmission of analogue, digital and speech signals;
- observe the modulated signals in both the time and frequency domains.

Having successfully completed the lab, you will be able to:

- understand how the various modulation and demodulation techniques work;
- appreciate when non-coherent and coherent demodulation is most appropriate;
- appreciate the characteristics of various modulated signals.

Section 2 of this document details the preparation that should be completed before attending the corresponding lab session. The work that every student is expected to complete within the lab session is detailed in Section 3. Section 4 provides some additional work that can be completed during the lab session, in order to receive higher marks. Finally, an appendix is provided at the end of this document, in order to explain the conventions adopted by the ETT-101.

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2 Preparation

Read through the course handbook statement on safety and safe working practices, and your copy of the standard operating procedure. Make sure that you understand how to work safely. Read through this document so you are aware of what you will be expected to do in the lab.

Before the laboratory session begins, complete the following pieces of preparation in your **logbook**. To help you with this preparation and with the laboratory session itself, you should look at the notes for your ELEC1207 Electronic Systems lectures and the core textbook for that course, Communication Engineering Principles by Ifiok Otung.

✎ 1. Write some bulletpoints about the relative advantages and disadvantages of Amplitude Modulation (AM), Double Side-Band Suppress Carrier (DSBSC), Quadrature Amplitude Modulation (QAM), Frequency Modulation (FM), Amplitude Shift Keying (ASK), Phase Shift Keying (BPSK) and Frequency Shift Keying (FSK).

✎ 2. Using Equations 1 and 2, determine the value of the DC offset A required to achieve an AM modulation index of 90%, when modulating the message signal $x(t) = \cos(2\pi f_\mathrm{m} t)$.

✎ 3. Figure 5 shows the amplitude spectrum of the message signal $x(t) = \cos(2\pi f_\mathrm{m} t)$ that is AM modulated onto a carrier having a frequency f_c. Sketch a prediction for the amplitude spectrum of an *audio* signal that is AM modulated onto a 100 kHz carrier. Note that audio signals are comprised of components having frequencies in the range 20 Hz to 20 kHz, where those with higher frequencies tend to have lower amplitudes.

✎ 4. For the case where the signal $x(t) = \cos(2\pi 2000 t)$ is AM modulated onto a 100 kHz carrier using a modulation index of 90%, sketch some predictions for the signals that are output by the rectifier and the peak detector, which are employed by the envelope detector shown in Figure 6.

✎ 5. For the case where an audio signal is AM modulated onto a 100 kHz carrier, determine the range of cutoff frequencies that are suitable for the Low Pass Filter (LPF), which is employed by the product detector shown in Figure 9. Note that audio signals are comprised of components having frequencies in the range 20 Hz to 20 kHz.

✎ 6. Using a similar process to that employed in Equations 1, 5, 6 and 7, show that the QAM schematic of Figure 11 allows two signals to be modulated onto a single carrier and then separated in the demodulator. You may need the trigonometric identities $\cos(A)\cos(B) = \frac{1}{2}\cos(A-B) + \frac{1}{2}\cos(A+B)$, $\sin(A) = \cos(A-\pi/2)$ and $-\cos(A) = \cos(A-\pi)$.

✎ 7. Figure 19 provides constellation diagrams for the OOK, BPSK and QPSK schemes of Sections 3.3 – 4.1. Find and sketch at least five more examples of constellation diagrams by looking through your lecture notes, text book and the Internet. Make sure you note the names of the constellation diagrams and state how many bits are transmitted at a time in each case. Also, state the relationship between the number of constellation points M and the number of bits k that are transmitted by each symbol.

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3 Laboratory Work

3.1 Amplitude Modulation

3.1.1 Modulation

When a Radio Frequency (RF) signal is placed onto the antenna of a transmitter, it will propagate through free space and can be detected on the antenna of a receiver. The higher the frequency of this signal, the smaller the antennas that are required. However, we are often interested in communicating relatively low frequency message signals, such as audio. Hence, we must modulate our low frequency message signal onto a high frequency *carrier*, in order to transmit it. This has the added benefit of allowing us to modulate different message signals onto different carrier frequencies, in order to transmit them without interfering with each other.

Amplitude Modulation (AM) uses the message signal $x(t)$ to vary the amplitude of the carrier sinusoid $\cos(2\pi f_c t)$, where the carrier frequency f_c is usually much higher than the highest frequency in the message signal. Figure 1 below shows a simple message signal $x(t)$ and an unmodulated carrier $\cos(2\pi f_c t)$. It also shows the AM signal $y(t)$ that results from modulating the message onto the carrier.

The dotted lines in the AM signal $y(t)$ of Figure 1 track the signal's envelopes (that is, its positive peaks and negative peaks). If you look at the envelopes closely you'll notice that they have the same shape as the message signal $x(t)$.

As shown in the block diagram of Figure 2, the mathematical model that defines the AM signal is

$$y(t) = [A + x(t)]\cos(2\pi f_c t), \tag{1}$$

where A is a constant DC offset.

The value of the DC offset A affects the maximum peak-to-peak voltage V_{ppmax}, as well as the minimum peak-to-peak voltage V_{ppmin}, as shown in Figure 1. The *modulation factor* m is defined as

$$m = \frac{V_{\text{ppmax}} - V_{\text{ppmin}}}{V_{\text{ppmax}} + V_{\text{ppmin}}}. \tag{2}$$

When the DC offset A is large, V_{ppmax} and V_{ppmin} will both be large and the modulation factor m will be close to zero. As A is reduced, V_{ppmin} will approach zero and the modulation

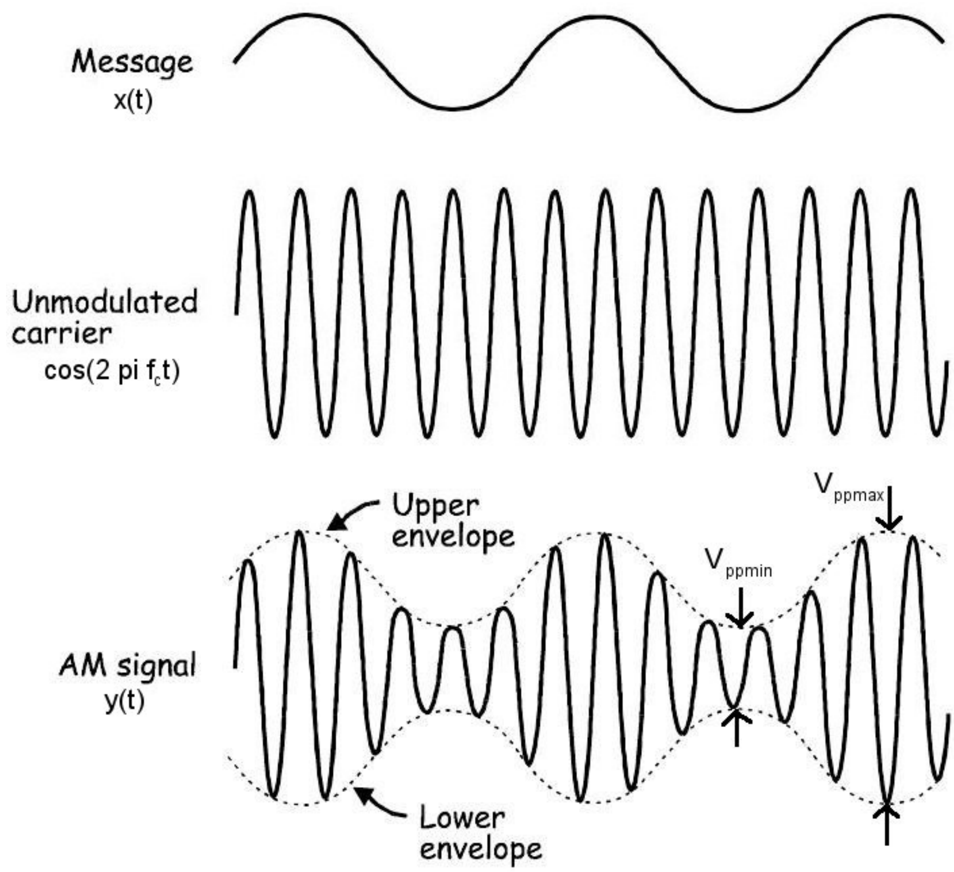

Figure 1

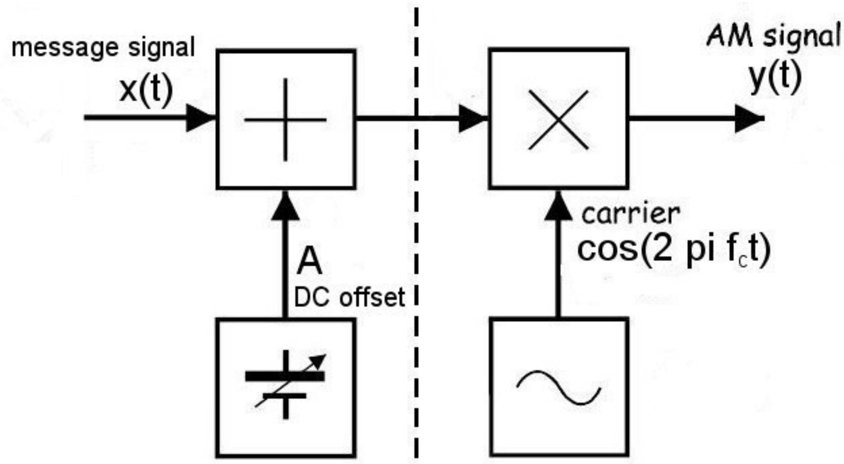

Figure 2

factor m will increase towards one. Since the modulation factor is in the range $[0, 1]$, it can be expressed as a percentage, whereupon it is called the *modulation index*. When $V_{\text{ppmin}} > 0$, the modulation index will be less than 100% and the AM signal $y(t)$ is said to be *undermodulated*. When $V_{\text{ppmin}} = 0$, the modulation index will be 100% and the AM signal $y(t)$ is said to be *100% modulated*. If the DC offset A is further reduced after V_{ppmin} has been reduced to zero, then the AM signal $y(t)$ becomes *overmodulated*. When the DC offset A is reduced to 0, the signal $y(t)$ stops being AM modulated and instead becomes Double Side-Band Suppressed Carrier (DSBSC) modulated.

When the message $x(t)$ is a simple sinewave having a frequency $f_{\text{m}} < f_{\text{c}}$ (like in Figure 1), Equation 1 becomes

$$y(t) = [A + \cos(2\pi f_{\text{m}} t)] \cos(2\pi f_{\text{c}} t). \tag{3}$$

Using the trigonometric identity $\cos(\alpha)\cos(\beta) = \frac{1}{2}\cos(\alpha - \beta) + \frac{1}{2}\cos(\alpha + \beta)$, we obtain

$$y(t) = A\cos(2\pi f_{\text{c}} t) + \frac{1}{2}\cos(2\pi[f_c - f_m]t) + \frac{1}{2}\cos(2\pi[f_c + f_m]t). \tag{4}$$

This shows that the AM signal $y(t)$ is the sum of three sinusoids, when $x(t) = \cos(2\pi f_{\text{m}} t)$:

- one with a frequency equal to the difference between the carrier and message frequencies $[f_c - f_m]$;
- one with a frequency equal to the carrier frequency f_c;
- one with a frequency equal to the sum of the carrier and message frequencies $[f_c + f_m]$.

Equation 4 implies that, for every sinusoid in the message $x(t)$, a pair of sinusoids is generated, having frequencies on either side of the carrier frequency f_{c}. Fourier theory tells us that message signals such as speech and music can be considered to comprise a sum of many different sinusoids, having different frequencies. When these signals are modulated onto a carrier, the various frequencies will be mapped to pairs of frequencies on either side of the carrier frequency. These two groups are called the sidebands. It is these sidebands that contain all of the message information. While the sinusoid at the carrier frequency f_{c} does not contain any message information, it is useful for helping the receiver 'lock-on' to the transmission.

Note that when $A = 0$, the sinusoid at the carrier frequency f_c is removed or *suppressed* from Equation 4, leaving only the two sidebands. This is why the modulation that results when $A = 0$ is called Double Side-Band Suppressed Carrier (DSBSC) modulation. By omitting the sinusoid at the carrier frequency, DSBSC modulation is more power efficient than AM (at least two thirds of the transmission power is used by the carrier in AM!). However, it becomes more difficult for the receiver to 'lock-on' to the transmission when DSBSC modulation is used.

In this experiment you'll use the ETT-101 to generate an AM signal by implementing the block diagram of Figure 2.

1. Build the left half of the block diagram shown in Figure 2 by making the connections shown in Figure 3. Remember to insert the black plugs of the oscilloscope leads into ground (GND) sockets. The ETT-101 leads come in a variety of colours; you may like to pick a particular colour to carry the *baseband* signals shown in the left half of Figure 2.

2. Point the potentiometers in the 'ADDER' and 'VARIABLE DCV' modules to the 12 o'clock position.

3. Ensure that the 'DC coupling' and the '×1 probe' settings are used in the oscilloscope channel you are using. Use 1 V per division on this channel and adjust the scope's timebase control to 100 μs per division. Move the scope's trigger level to the highest position that allows reliable triggering.

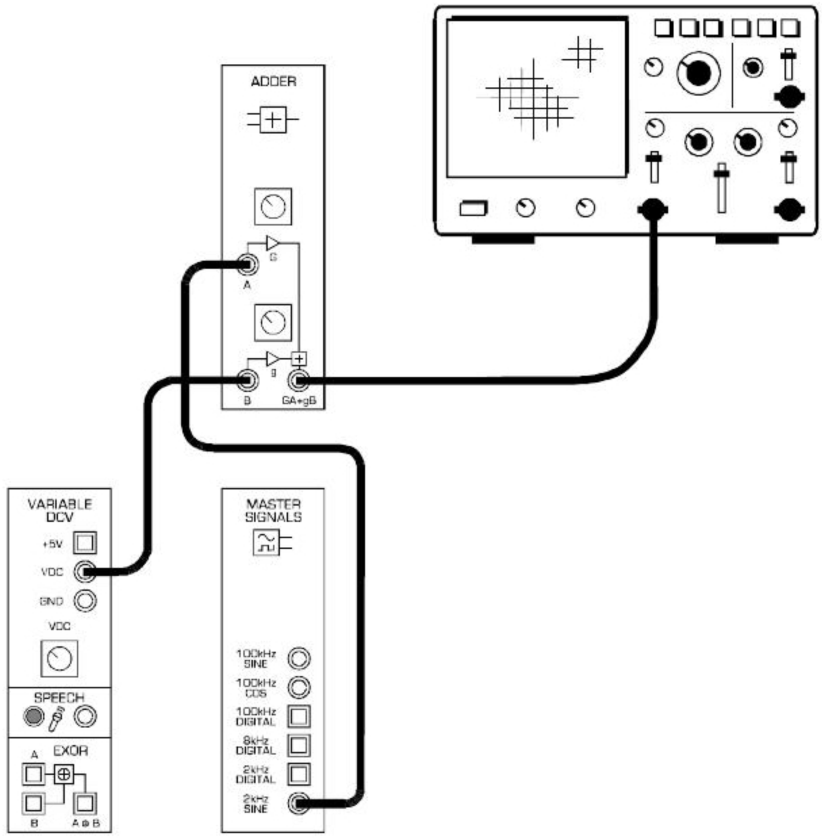

Figure 3

4. The potentiometer in the 'VARIABLE DCV' module controls the value of the DC offset A in Equation 1. Turning the potentiometer as far as it will go in an anti-clockwise direction gives $A = +2.5$ V, the 12 o'clock position gives $A = 0$ V and turning the potentiometer as far as it will go in a clockwise direction gives $A = -2.5$ V. Adjust the value of A within the range 0 to 2.5 V and make sure that you can see the sinusoid moving up and down on the oscilloscope. If you find that the sinusoid is not moving up and down, then the oscilloscope is probably using AC coupling instead of DC coupling for the channel you are using. When you are finished, leave the potentiometer in the position that gives the highest value for A that allows reliable triggering.

5. Modify the setup as shown in Figure 4, in order to complete the block diagram of Figure 2. The dotted lines in Figure 4 show connections that are already in place. In order to carry the RF signals shown in the right half of Figure 2, you may like to use ETT-101 leads that have a different colour to the ones you used for the baseband signals.

6. Turn on the scope's second channel and use the 'DC coupling' and the '×1 probe' settings as well as 2 V per division for this channel, which is displaying the output 'kXY' from the 'MULTIPLIER' module.

7. The oscilloscope should be displaying something that looks like the AM signal in Figure 1. Sketch this display in your **logbook**. Alternatively, you may like to print out the oscilloscope's display by capturing it on the computer. You can do this using the Tektronix OpenChoice Desktop software, which is described at
`https://secure.ecs.soton.ac.uk/notes/ellabs/databook/equip/`
Also, state the values of f_m and f_c in your **logbook** and use these to calculate the frequencies of the three sinusoids that comprise the AM signal $y(t)$, as shown in Equation 4. Finally, calculate the modulation factor m of your AM signal $y(t)$ using Equation 2.

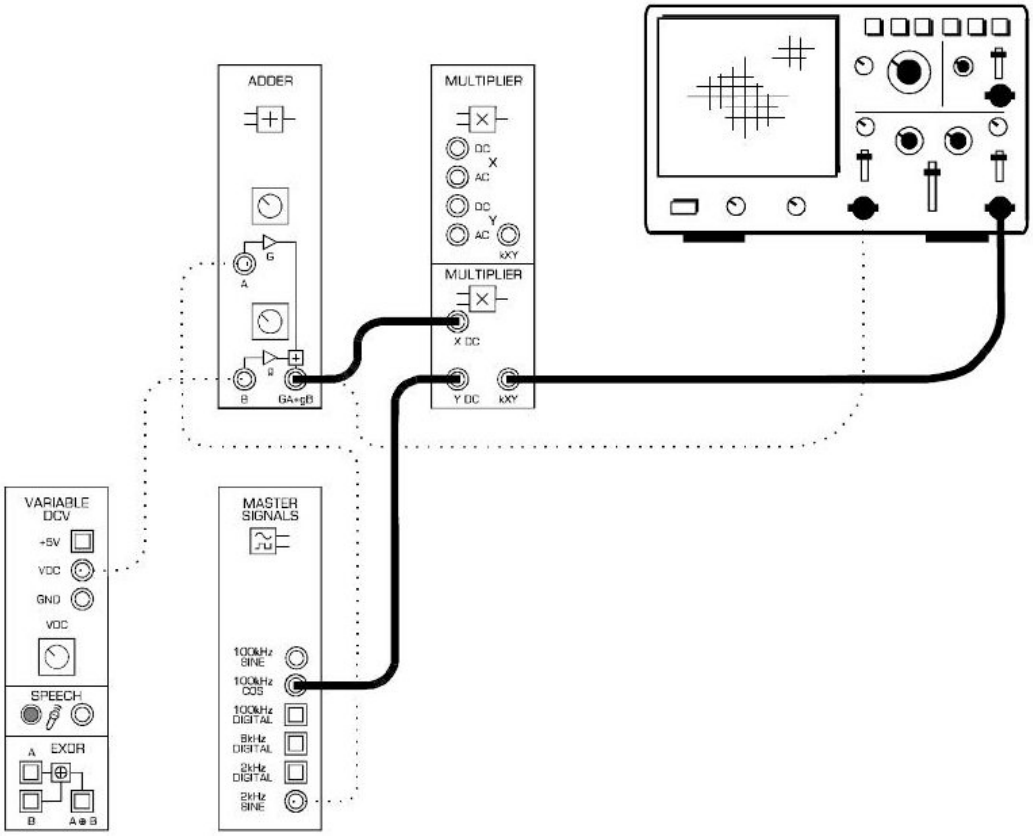

Figure 4

8. Now that we've looked at the AM signal in the *time domain*, let's take a look at its amplitude spectrum, which exists in the *frequency domain*. Use the 'MATH' button on the oscilloscope to obtain a Fast Fourier Transform (FFT) for your AM signal. Use the zoom, timebase and horizontal position controls on the oscilloscope in order to clearly display the spikes that correspond to the three components of the AM signal $y(t)$. You should end up with something that approximates Figure 5. Sketch the FFT of the AM signal in your **logbook** and annotate it with the correct frequencies.

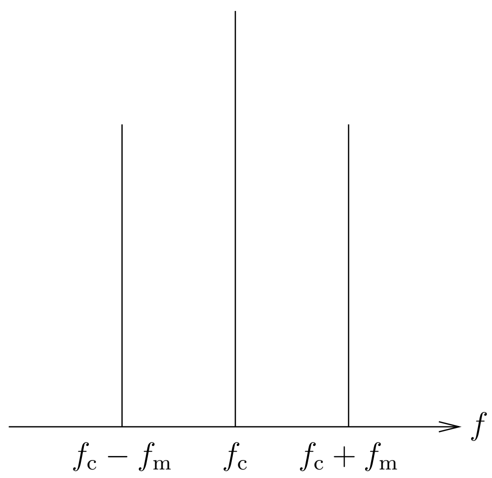

Figure 5

9. Gradually reduce the value of the DC offset A towards 0 V and observe how the AM signal $y(t)$ changes in the time and frequency domain as it goes from being undermodulated, to

being 100% modulated, to being overmodulated and finally to being DSBSC modulated. Comment on these changes in your **logbook**. When you are finished, return the DC offset A to the highest value that allows reliable triggering, which should give undermodulation.

3.1.2 Non-coherent demodulation

Provided that an AM signal is not overmodulated, it can be recovered simply using *envelope detection*. This is a form of non-coherent demodulation, since it does not require any knowledge of the frequency and phase of the carrier. A block diagram of an envelope detector is provided in Figure 6. The rectifier of Figure 6 may be implemented using a diode, while the peak detector can be implemented using a resistor and a capacitor to form a low pass filter (RC LPF).

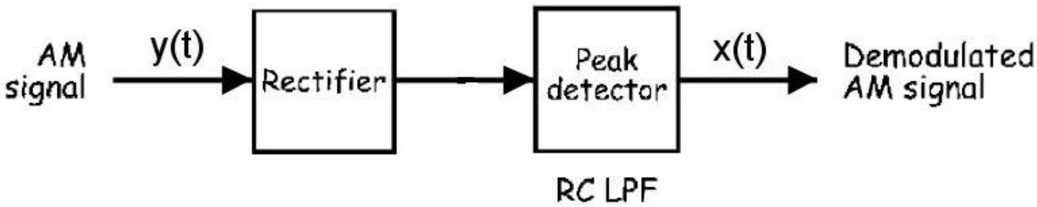

Figure 6

1. You can build the block diagram of Figure 6 by modifying your connections as shown in Figure 7.

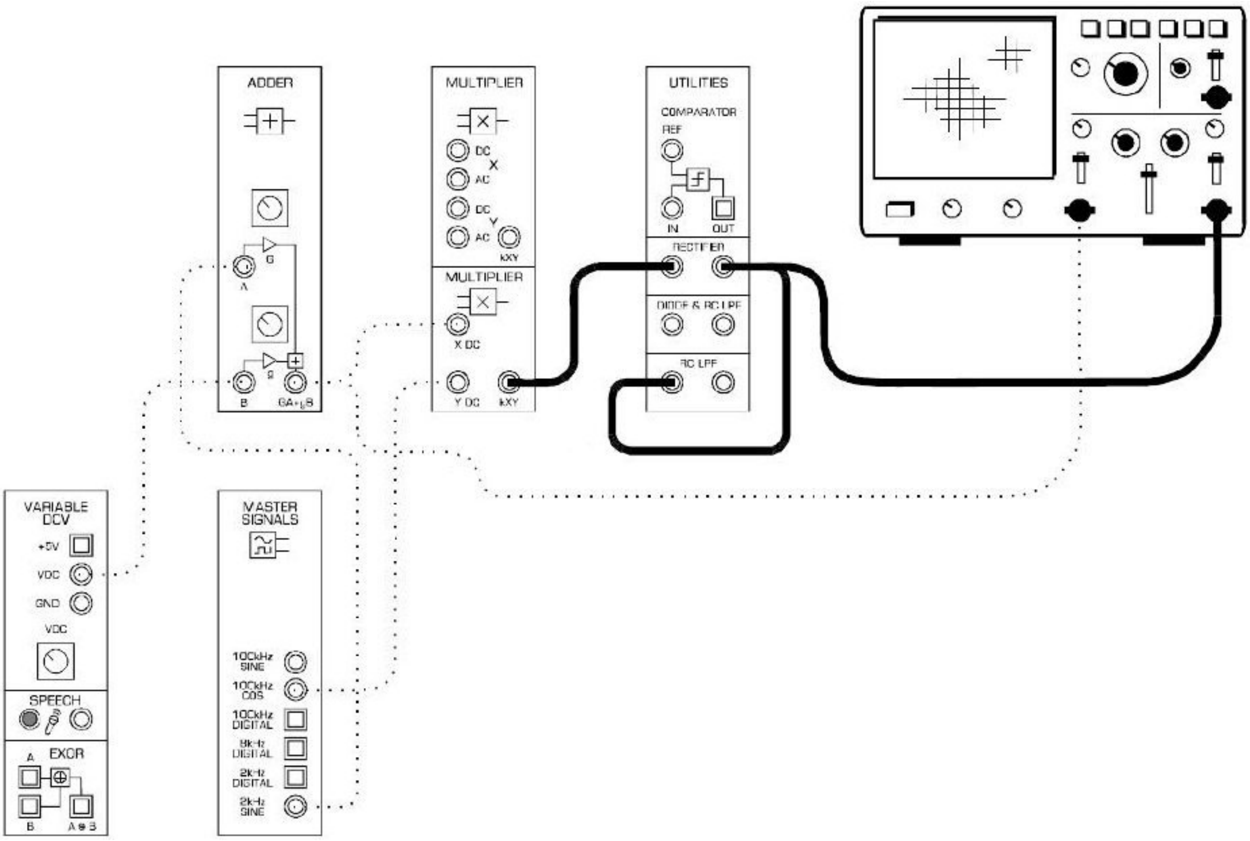

Figure 7

✎ 2. The oscilloscope display should be showing the output of the rectifier shown in Figure 6. Sketch this in your **logbook** and explain what the rectifier has done to the AM signal. Also, comment on how the signal compares to the prediction you made when preparing for this lab.

✎ 3. Gradually reduce the value of the DC offset A towards 0 V and observe how the rectified signal changes as the AM signal $y(t)$ goes from being undermodulated, to being 100% modulated, to being overmodulated and finally to being DSBSC modulated. Comment on these changes in your **logbook**. When you are finished, return the DC offset A to the highest value that allows reliable triggering, which should give undermodulation.

✎ 4. Connect a third oscilloscope probe to the output of the 'RC LPF' module. If you don't have a third probe available, then move the probe from the output of the 'RECTIFIER' module. By 'smoothing' the output of the rectifier, the RC LPF reconstructs the message signal $x(t)$. Sketch the oscilloscope display in your **logbook** and explain why rectification is necessary before 'smoothing'. Also, comment on how the signal compares to the prediction you made when preparing for this lab.

✎ 5. Again, gradually reduce the value of the DC offset A towards 0 V and observe how the reconstructed signal changes as the AM signal $y(t)$ goes from being undermodulated, to being 100% modulated, to being overmodulated and finally to being DSBSC modulated. Comment on these changes in your **logbook** and explain why overmodulation prevents the use of non-coherent demodulation. When you are finished, return the DC offset A to the highest value that allows reliable triggering, which should give undermodulation.

6. By making the connections shown in Figure 8 and plugging in some headphones, you can use your setup to modulate and demodulate speech. Give this a go and gradually reduce the value of the DC offset A towards 0 V. You should notice that the intelligible reconstruction of the speech fails when the AM signal $y(t)$ becomes overmodulated. When you are finished, return the DC offset A to its starting position.

3.1.3 Coherent demodulation

In the previous section, we saw that overmodulation prevents non-coherent demodulation. However, the demodulator can use *coherent demodulation* to demodulate an overmodulated AM signal, provided that it has exact knowledge of the carrier frequency and phase. This knowledge can be obtained using a phase locked loop, for example. In the second year lab C8, you'll see what happens if the receiver's knowledge of the carrier frequency and phase is not exactly correct, but don't worry about this for now. Coherent demodulation is achieved using a *product detector*, as shown in the block diagram of Figure 9.

As its name implies, the product detector uses multiplication and so mathematics is necessary to explain its operation. The incoming AM signal is multiplied by a pure sinewave that must be the same frequency and phase as the AM signal's carrier. This sinewave is generated by the receiver and is known as the local carrier. To see how this process recovers the message, let's describe product detection mathematically as

$$u(t) = y(t)\cos(2\pi f_c t). \tag{5}$$

Substituting Equation 1 in gives

$$u(t) = [A + x(t)]\cos(2\pi f_c t)\cos(2\pi f_c t). \tag{6}$$

Using the trigonometric identity $\cos(\alpha)\cos(\alpha) = \frac{1}{2} + \frac{1}{2}\cos(2\alpha)$, we obtain

$$u(t) = \frac{1}{2}[A + x(t)] + \frac{1}{2}[A + x(t)]\cos(2\pi[2f_c]t). \tag{7}$$

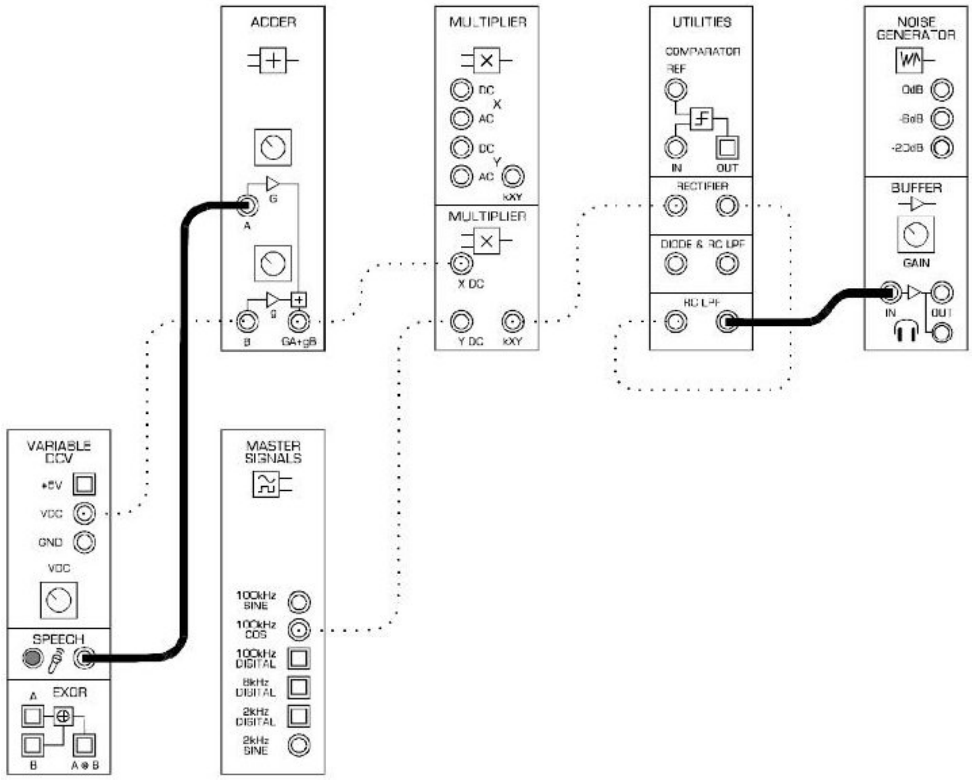

Figure 8

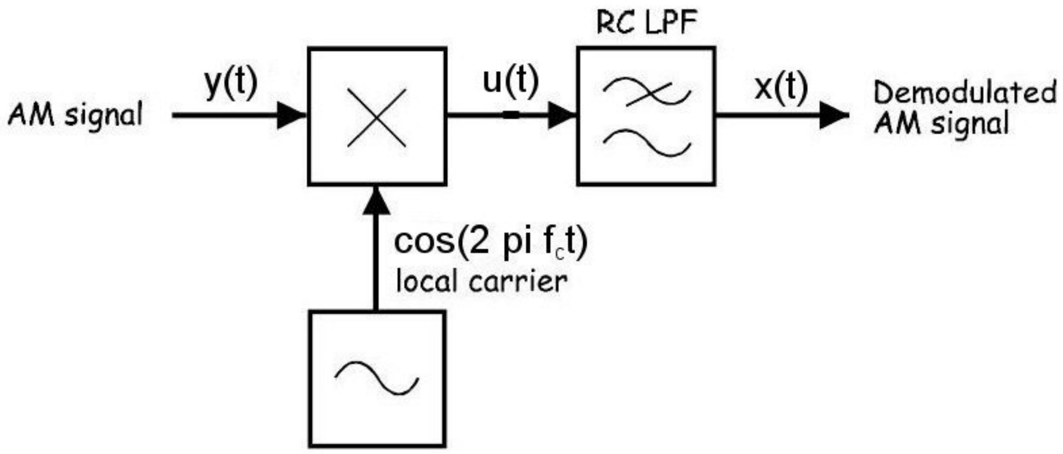

Figure 9

By comparing the $\frac{1}{2}[A + x(t)]\cos(2\pi[2f_c]t)$ term in Equation 7 with Equation 1, we can see that it is our message signal $x(t)$, but AM modulated onto a carrier having a frequency of $2f_c$. This AM signal can be removed from $u(t)$ by the low pass filter shown in the block diagram of Figure 9, leaving only the $\frac{1}{2}[A + x(t)]$ term from Equation 7. This term is our message signal $x(t)$, but with a DC offset A and attenuated by half, which are transformations that are easily fixed.

1. Return your setup to the state shown in Figure 4
2. Modify the set-up as shown in Figure 10.

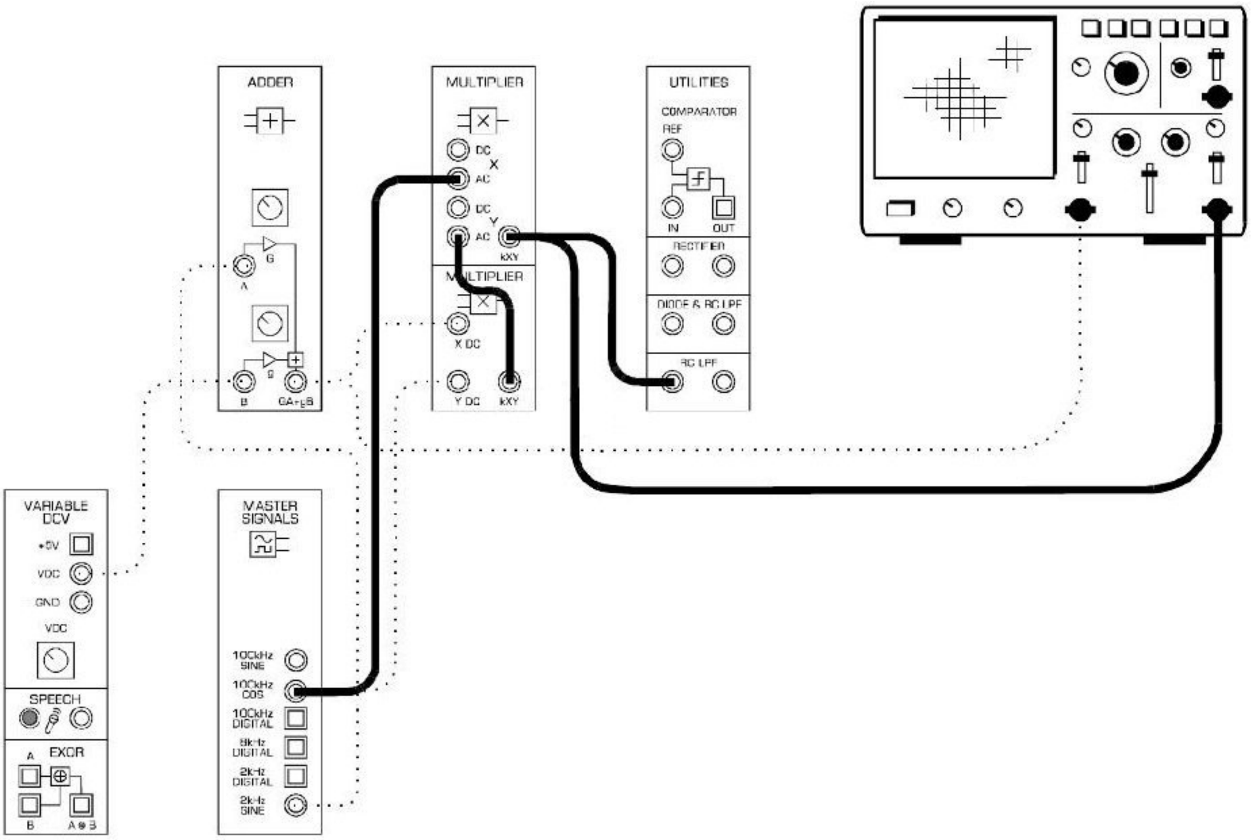

Figure 10

3. The oscilloscope display should be showing the output of the multiplier $u(t)$ shown in Figure 9. Sketch this in your **logbook**.
4. Gradually reduce the value of the DC offset A towards 0 V and observe how the signal $u(t)$ changes as the AM signal $y(t)$ goes from being undermodulated, to being 100% modulated, to being overmodulated and finally to being DSBSC modulated. Comment on these changes in your **logbook**. When you are finished, return the DC offset A to the highest value that allows reliable triggering, which should give undermodulation.
5. Connect a third oscilloscope probe to the output of the 'RC LPF' module. If you don't have a third probe available, then move the probe from the output of the 'MULTIPLIER' module. By 'smoothing' the output of the multiplier, the RC LPF reconstructs the message signal $x(t)$. Sketch the oscilloscope display in your **logbook**.
6. Again, gradually reduce the value of the DC offset A towards 0 V and observe how the

reconstructed signal changes as the AM signal $y(t)$ goes from being undermodulated, to being 100% modulated, to being overmodulated and finally to being DSBSC modulated. Comment on these changes in your **logbook** and explain why coherent demodulation is able to reconstruct the message signal $x(t)$, even when the AM signal $y(t)$ is overmodulated. When you are finished, return the DC offset A to the highest value that allows reliable triggering, which should give undermodulation.

7. By making similar connections to the ones shown in Figure 8, you can use your setup to modulate and demodulate speech. Give this a go and gradually reduce the value of the DC offset A towards 0 V. You should notice that the reconstruction of the speech succeeds, even when the AM signal $y(t)$ becomes overmodulated. When you are finished, return the DC offset A to its starting position.

3.2 Quadrature Amplitude Modulation

In Equations 1 and 5, we considered the use of a cosine carrier wave. If we had used sine carrier waves instead, we would have obtained very similar results. This is because the sine and cosine functions differ only by a phase shift of $\pi/2$ radians, ie $\cos(\alpha) = \sin(\alpha + \pi/2)$. As a result, $\sin(\alpha) = 0$ if $\cos(\alpha) = \pm 1$ and $\cos(\alpha) = 0$ if $\sin(\alpha) = \pm 1$. These results indicate that the cosine and sine functions are *orthogonal* to each other. This means that a message signal $x_{\mathrm{i}}(t)$ that is amplitude modulated onto a cosine carrier wave will not interfere with another message signal $x_{\mathrm{q}}(t)$ that is modulated onto a sine carrier wave having the same frequency f_{c}. In this way, we can transmit two message signals at once, which is useful for stereo audio for example. We refer to these message signals as the *in-phase signal* $x_{\mathrm{i}}(t)$ and the *quadrature-phase signal* $x_{\mathrm{q}}(t)$. The additional presence of the quadrature-phase signal gives *Quadrature Amplitude Modulation* (QAM) its name. A schematic for a QAM scheme is shown in Figure 11

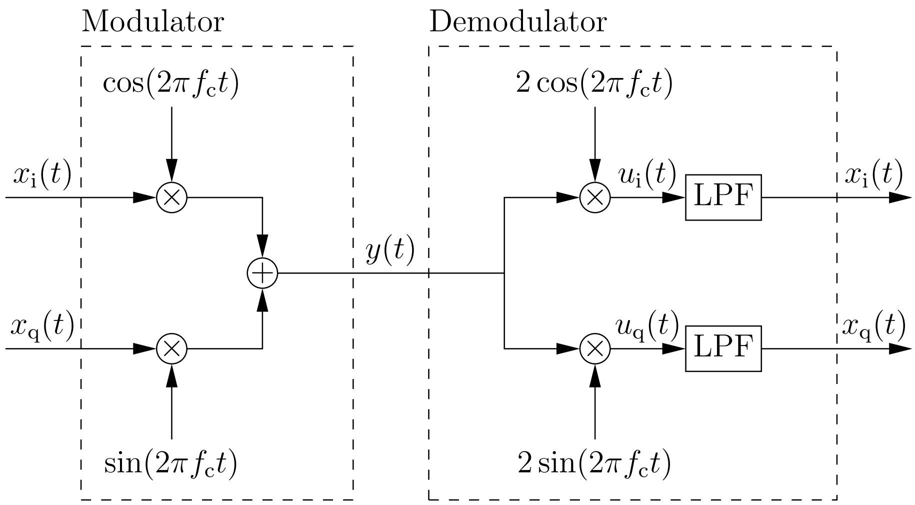

Figure 11

1. Implement the schematic of Figure 11 by making the connections shown in Figure 12. This uses a 2 kHz sinusoid for the in-phase signal $x_{\mathrm{i}}(t)$ and the signal from the microphone for the quadrature-phase signal $x_{\mathrm{q}}(t)$.
2. Set the potentiometers in the 'ADDER' module to the 12 o'clock position.
3. Since the ETT-101 has only three multipliers, we cannot demodulate both the in-phase signal and the quadrature-phase signal at once. However, the 'PHASE SHIFTER' module

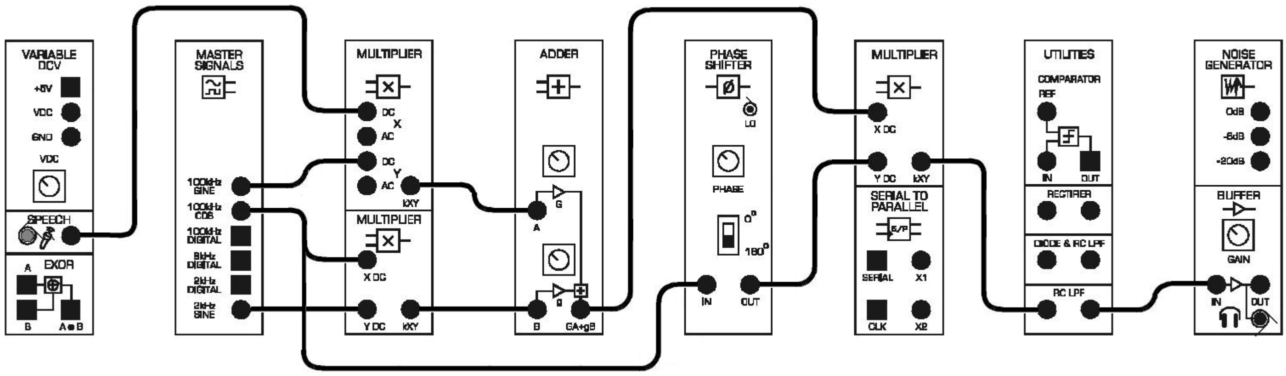

Figure 12

lets us listen to one signal or the other. When this shifts the cosine wave generated in the receiver by 0 radians, we get the in-phase signal. When it shifts the cosine wave by $\pi/2$ radians, we get the quadrature-phase signal. Turn the potentiometer on the 'PHASE SHIFTER' module very slowly to listen to the two different signals. Note that this potentiometer is very sensitive. In your **logbook**, explain what happens when the phase shift is not a multiple of $\pi/2$ radians. Also, explain why QAM must therefore use coherent demodulation, even if we added DC offsets to the message signals before modulating them onto the carriers.

3.3 On-Off Keying

So far, we've been considering *analogue* modulation schemes. Let's look at some digital modulation schemes now. These can be used to transmit a sequence of binary digits (bits) at a particular *rate* f_b, which is measured in bits per second. Each bit therefore has a *period* of $1/f_b$ seconds. On-Off Keying (OOK) is a special type of Amplitude Shift Keying (ASK) in which the carrier sinusoid is switched on or off for the duration of each bit period, depending on the value of the corresponding bit. In this experiment, the carrier is amplitude modulated using a digital signal that takes on a value of 0 V during the period of 0-valued bits and a value of 5 V during the period of 1-valued bits. The result is shown in Figure 13.

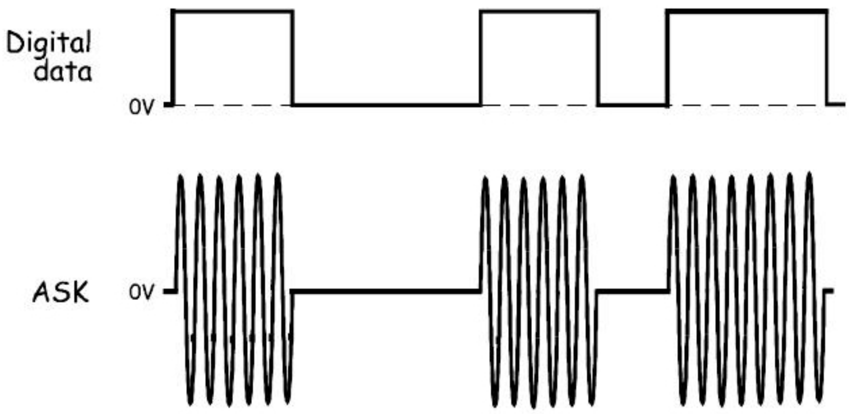

Figure 13

In the block diagram of Figure 14, OOK is achieved by using the digital signal to switch the carrier sinusoid on and off. As shown in Figure 13, the envelope of the OOK signal matches the original digital signal. For this reason, the envelope detector of Section 3.1.2 is employed in the block diagram of Figure 14. Following this, the digital signal is restored using the comparator shown in Figure 14. This outputs 5 V when the demodulated signal is higher than a reference DC voltage and 0 V when the demodulated signal is lower.

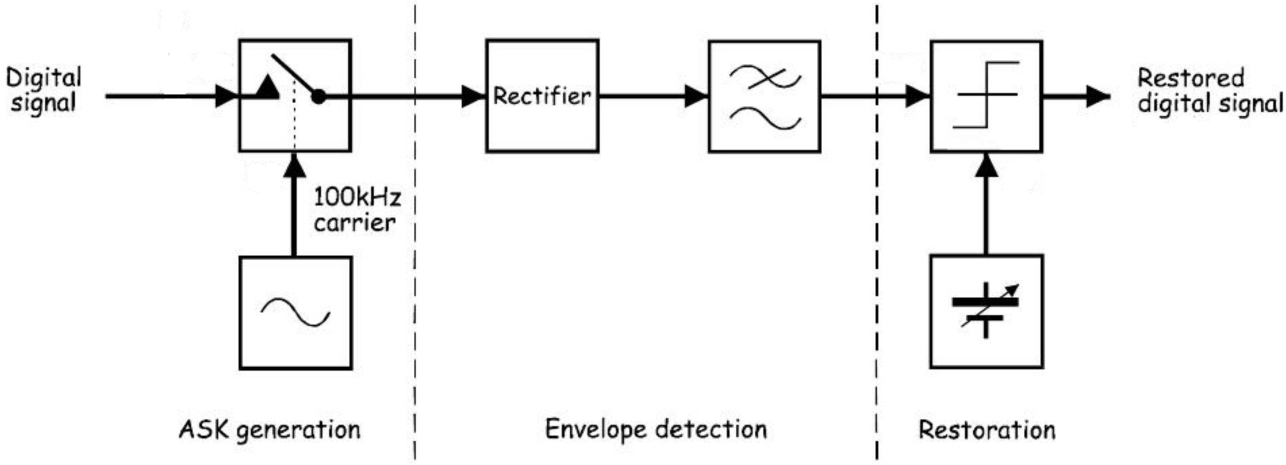

Figure 14

1. Implement the block diagram of Figure 14 using the connections shown in Figure 15. In order to carry the digital signals shown in Figure 14, you may like to use ETT-101 leads that have a different colour to ones you used for the baseband and RF signals.

2. Make sure that the oscilloscope is triggering on the channel that is connected to the 'SYNC' output of the 'SEQUENCE GENERATOR' module. Also, make sure that both channels of the oscilloscope are DC coupled.

✎ 3. The 'X' output of the 'SEQUENCE GENERATOR' module repeatedly outputs a particular 31-bit Pseudo Random Binary Sequence (PRBS). Write it in your **logbook** and sketch the corresponding digital signal.

✎ 4. Connect a third oscilloscope probe to the output of the 'DUAL ANALOGUE SWITCH' module. If you don't have a third probe available, then move the probe from the 'X' output of the 'SEQUENCE GENERATOR' module. Sketch the OOK signal in your **logbook**. You may need to adjust the timebase of the oscilloscope to properly see what's going on.

✎ 5. Now move an oscilloscope probe to the output of the 'LPF' module and look at the recovered signal in the time domain. You should see that the recovered signal is not a perfect square wave. Sketch this signal in your **logbook** and explain its imperfections.

✎ 6. Finally, move an oscilloscope probe to the output of the 'COMPARATOR' module. Adjust the potentiometer on the 'VARIABLE DCV' module until you obtain the digital signal. Sketch this in your **logbook**.

3.4 Binary Phase Shift Keying

In Binary Phase Shift Keying (BPSK), the phase of a carrier sinusoid is keyed to either 0 radians or π radians for the duration of each bit period $1/f_{\mathrm{b}}$, depending on the value of the corresponding bit, as shown in Figure 16.

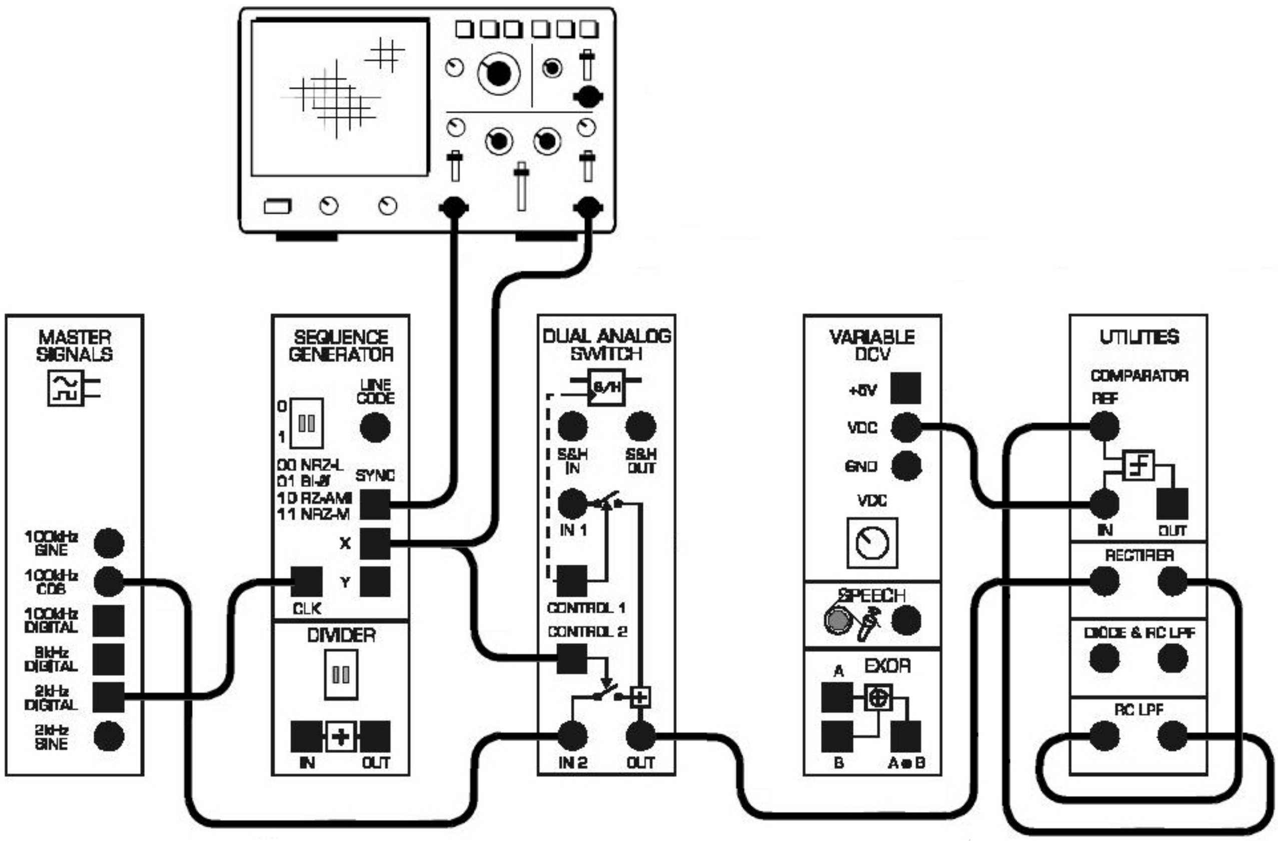

Figure 15

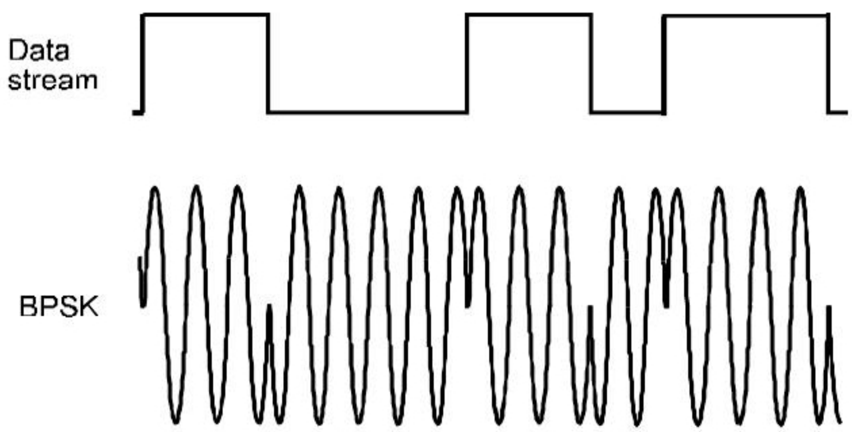

Figure 16

As shown in the block diagram of Figure 17, BPSK can be achieved by amplitude modulating the carrier using a digital signal that takes on a particular positive voltage (the ETT-101 uses +2.5 V) during the period of 1-valued bits and the corresponding negative voltage (−2.5 V) during the period of 0-valued bits. This works because multiplying a sinusoid by −1 shifts its phase by π radians. In other words, $-\cos(\alpha) = \cos(\alpha + \pi)$.

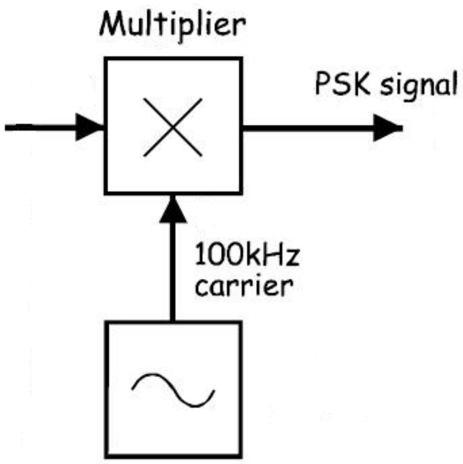

Figure 17

1. Implement the block diagram of Figure 17 using the connections shown in Figure 18.
2. Position the switches in the 'SEQUENCE GENERATOR' module to their upper positions. As before, make sure that the oscilloscope is triggering on the channel that is connected to the 'SYNC' output of the 'SEQUENCE GENERATOR' module. Also, make sure that both channels of the oscilloscope are DC coupled. Sketch the digital signal in your **logbook**.
3. Connect a third oscilloscope probe to the output of the 'MULTIPLIER' module. If you don't have a third probe available, then move the probe from the 'LINE CODE' output of the 'SEQUENCE GENERATOR' module. Sketch the PSK signal in your **logbook**. You may need to adjust the timebase of the oscilloscope to see what's going on properly.
4. Decide whether coherent or non-coherent demodulation is required for PSK and build the corresponding demodulator. Use the 'COMPARATOR' module to restore the demodulated signal. Draw a block diagram of the PSK modulator and demodulator in your **logbook**.

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4 Optional Additional Work

Additional marks can be earned by completing one of the following subsections, of your choice. Marks will only be awarded for this section if you have already completed all of Section 3 to an excellent standard and with excellent understanding.

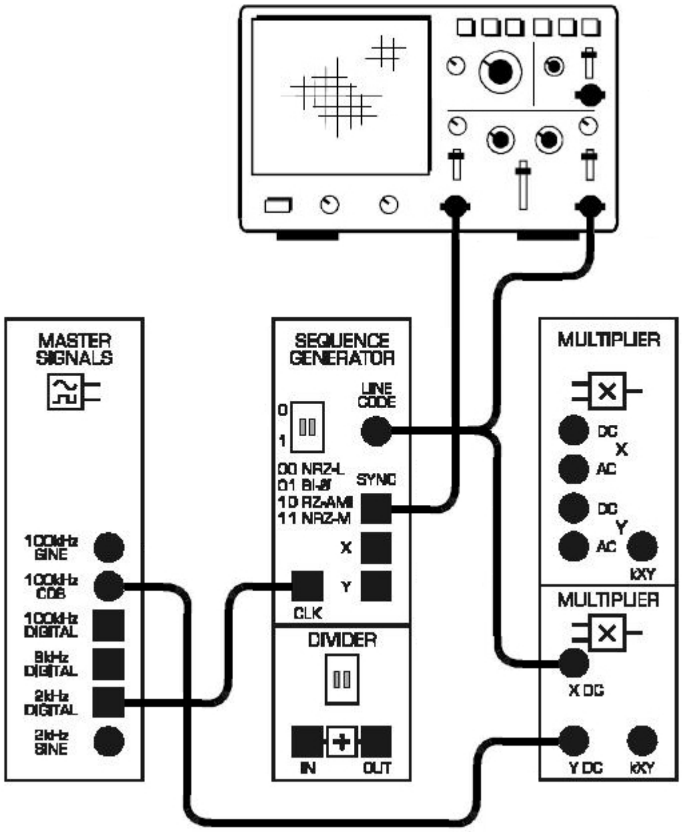

Figure 18

4.1 Quaternary Phase Shift Keying

So far, we've looked at two digital modulation schemes, OOK and BPSK, which transmit one bit at a time, using the same rate f_t of transmissions per second as the rate f_b of bits per second. During each transmission duration of $1/f_\text{t}$ seconds, OOK selects one of two *amplitudes* for the carrier depending on the value of the bit being transmitted. By contrast, BPSK selects one of two *phases* for the carrier depending on the value of the bit. The schemes can be characterised by the *constellation diagrams* shown in Figures 19a and 19b, respectively.

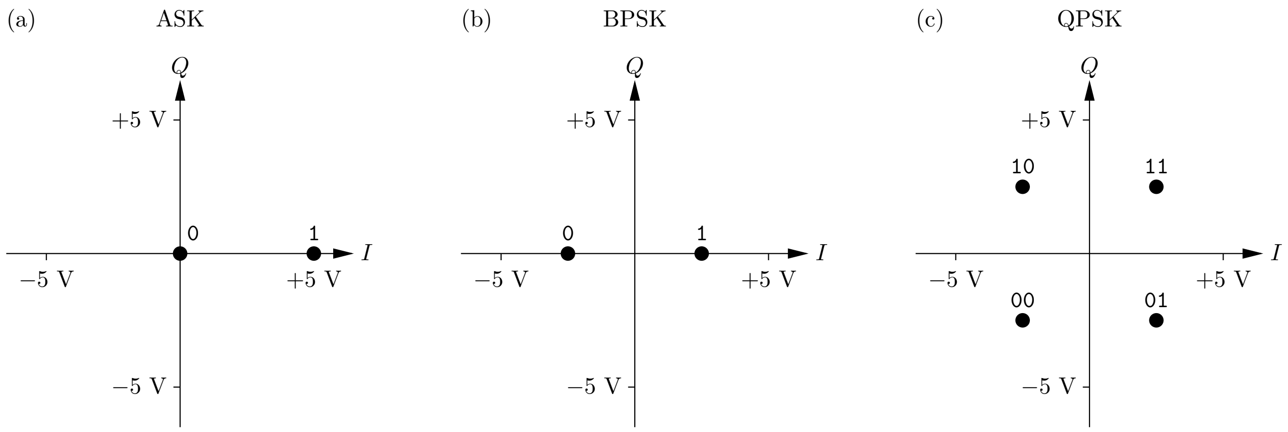

Figure 19

In Figure 19, each combination of amplitude and phase that can be selected for the carrier is indicated by a *constellation point*. Each of these points is labelled with the bit value that imposes the corresponding amplitude and phase on the carrier for a duration of $1/f_t$ seconds. The Euclidean distance between a particular constellation point and the origin of the diagram is equal to the amplitude that is used during the corresponding transmission. Similarly, a constellation point's rotation from the axis pointing to the right is equal to the phase that is imposed upon the carrier. So, a constellation point on the axis pointing to the right is associated with a phase shift of 0 radians, a constellation point on the axis pointing upwards is associated with a phase shift of $\pi/2$ radians, the axis pointing to the left is associated with a shift of π radians, and a shift of $3\pi/2$ radians is associated with the axis pointing downwards.

While OOK and BPSK transmit one bit at a time, Quaternary Phase Shift Keying (QPSK) transmits $k = 2$ bits at a time, at a rate of $f_t = \frac{1}{k} f_b$ transmissions per second. Since there are $2^k = 4$ possible combinations of $k = 2$ bits, QPSK requires four constellation points, as shown in Figure 19c. Note that each of the constellation points in Figure 19c is labelled with a different combination of two bit values. Also note that each of the QPSK constellation points has the same amplitude (distance from the origin) but a different phase shift (rotation from the axis pointing to the right). More specifically, the bit combination 11 is mapped to a phase shift of $\pi/4$ radians, 10 is mapped to $3\pi/4$ radians, 00 is mapped to $5\pi/4$ radians and 01 is mapped to $7\pi/4$ radians.

A carrier having the phase shifts indicated in Figure 19c can be obtained by summing cosine and sine waves having particular amplitudes. The amplitude required for the cosine carrier wave is given by the particular constellation point's position along the horizontal *in-phase* axis of the constellation diagram. Similarly, a constellation point's position along the vertical *quadrature-phase* axis indicates the amplitude required for the sine carrier wave. For example, the constellation point labelled 01 in Figure 19c is achieved by summing a cosine carrier wave having an amplitude of +2.5 V with a sine carrier having an amplitude of −2.5 V.

Take a look at the constellation points that are labelled 11 and 10 in Figure 19c. Notice that both these constellation points have labels in which the first bit is a 1 and both are associated with an quadrature-phase amplitude of +2.5 V. Meanwhile, the constellation points having labels in which the first bit is a 0 are both associated with an quadrature-phase amplitude of −2.5 V. Therefore, the value of the first bit can be used to select the amplitude for the sine carrier wave. In a similar way, Figure 19c shows that the value of the second bit can be used to select the amplitude of the cosine carrier wave. We can therefore achieve QPSK using the block diagram of Figure 20, in which the digital signal is +2.5 V during the period of 1-valued bits and −2.5 V during the period of 0-valued bits.

1. Implement the digital signal and the serial-to-parallel converter of Figure 20 using the connections shown in Figure 21.

2. Make sure that the oscilloscope is triggering on the channel that is connected to the output of the 'DIVIDER' module. Also, make sure that both channels of the oscilloscope are DC coupled.

✎ 3. The 'X1' output of the 'SERIAL TO PARALLEL' module repeatedly outputs a particular 31-bit PRBS. Write it in your **logbook** and sketch the corresponding digital signal.

✎ 4. Connect a third oscilloscope probe to the 'X2' output of the 'SERIAL TO PARALLEL' module. If you don't have a third probe available, then move the probe from the 'X1' output of the 'SERIAL TO PARALLEL' module. The 'X2' output repeatedly provides a different 31-bit PRBS. Write it in your **logbook** and sketch the corresponding digital signal.

✎ 5. You can use the connections shown in Figure 12 to implement the QPSK modulator of Figure 20, as well as the corresponding demodulator. In this case, the quadrature-phase

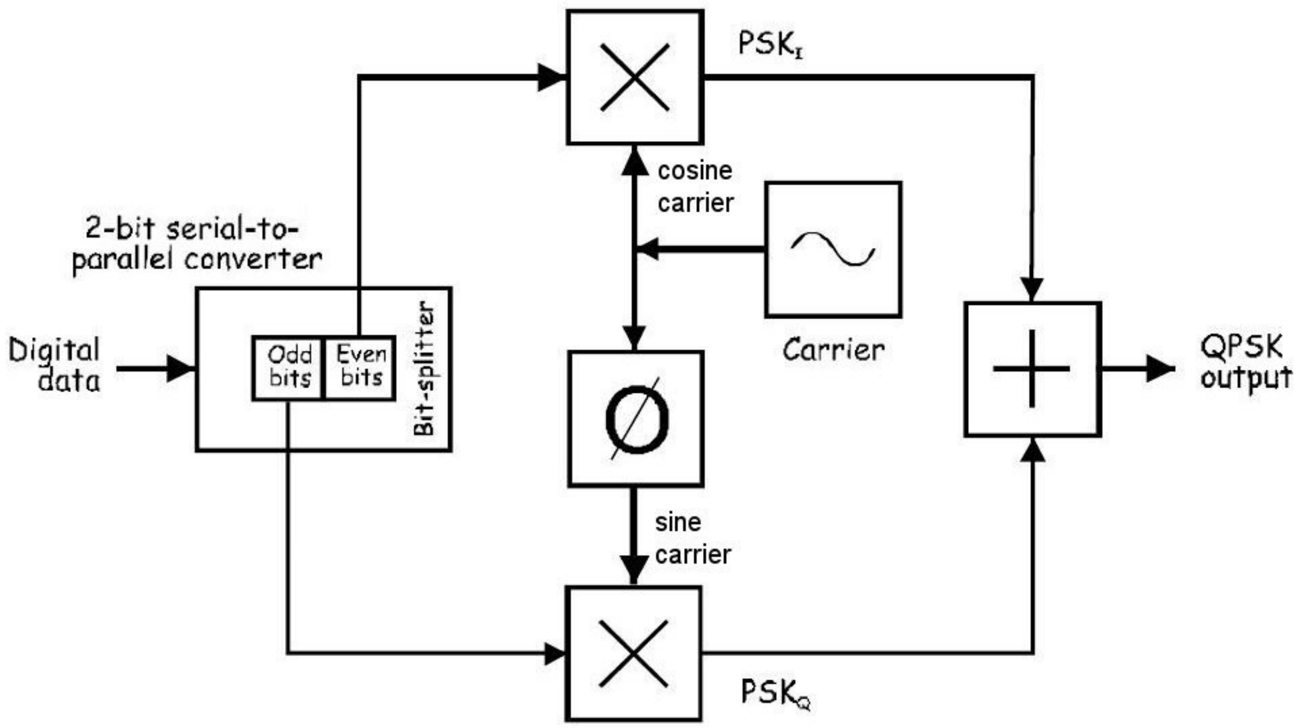

Figure 20

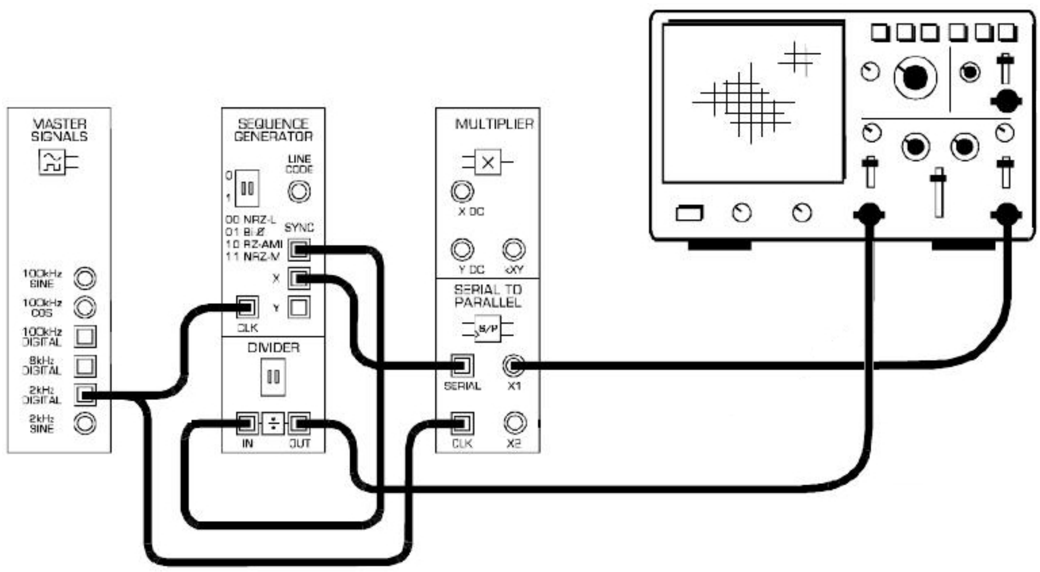

Figure 21

signal should be provided by the 'X1' output of the 'SERIAL TO PARALLEL' module, instead of by the 'SPEECH' module of Figure 12. Similarly, the in-phase signal should be provided by the 'X2' output, rather than the 2 kHz sine wave. As in Sections 3.3 and 3.4, use the 'COMPARATOR' module to reconstruct your digital signal. Draw a block diagram of the QPSK demodulator in your **logbook**.

✎

6. Move an oscilloscope probe to the output of the 'COMPARATOR' module. As in Section 3.2, turn the potentiometer on the 'PHASE SHIFTER' module very slowly to view the in-phase and quadrature-phase digital signals. Notice that, while the QAM scheme of Section 3.2 was extremely sensitive to the selection of the correct phase, the QPSK scheme is not as sensitive. Explain this advantage of QPSK in your **logbook**.

4.2 Frequency Modulation

As we saw in Section 3.1.1, we can convey a signal by using it to modulate the amplitude of a carrier. However, our signal will become corrupted in the presence of additive noise, which can disrupt the carrier's amplitude. In the second year lab C8, you'll see this for yourself, but don't worry about it for now. The presence of additive noise motivates *Frequency Modulation* (FM), which modulates the frequency of the carrier and is therefore significantly less sensitive to the disruptive effects of additive noise. However, FM doesn't gain this benefit for free; it requires significantly more bandwidth than AM. The FM concept is demonstrated in Figure 22, which shows that the frequency of the carrier is modulated by the amplitude of the message signal.

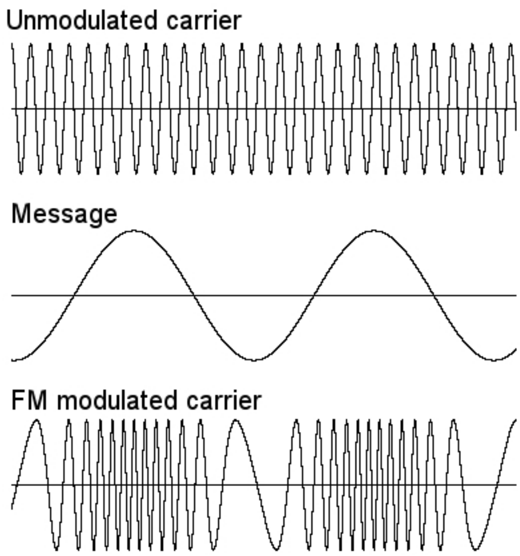

Figure 22

1. Use the connections shown in Figure 23 to build an FM modulator and demodulator.

✎

2. The 'VCO' (Voltage Controlled Oscillator) module implements the FM modulator. Set the switch to the 'HI' position, the 'GAIN' potentiometer to its most clockwise position and the 'FREQ' potentiometer to the 12 o'clock position. Observe how the frequency of the resultant carrier varies as you change the DC message signal using the potentiometer in the 'VARIABLE DCV' module. Write the range of frequencies that you can obtain in your **logbook**.

✎

3. As shown in the block diagram of Figure 24, an FM demodulator can be implemented

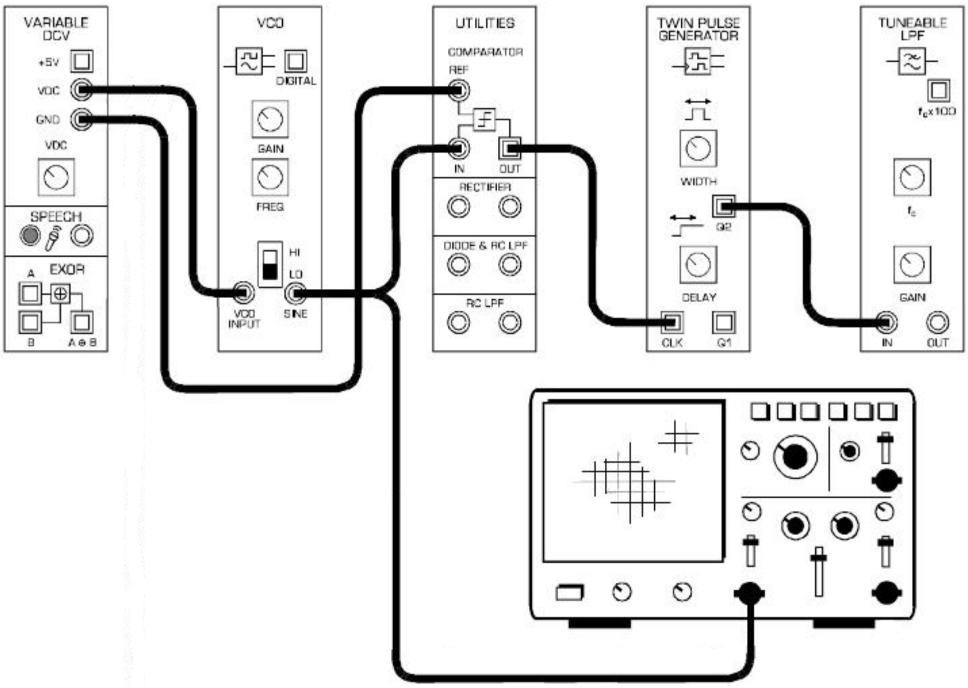

Figure 23

using a Zero Crossing Detector (ZCD) comprising a comparator, a pulse generator and a Low Pass Filter (LPF). Connect the second oscilloscope channel to the output of the 'COMPARATOR' module, making sure that it is DC coupled. Sketch the FM signal and the output of the comparator in your **logbook**.

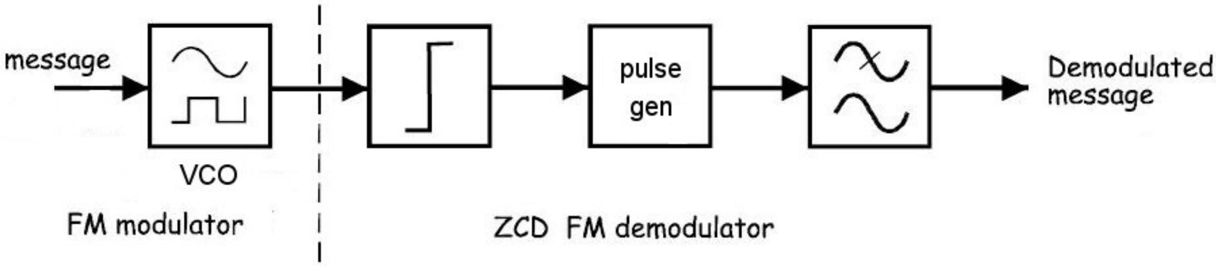

Figure 24

4. Turn both potentiometers in the 'TWIN PULSE GENERATOR' module as far as possible in an anti-clockwise direction. Move the second oscilloscope probe from the output of the 'COMPARATOR' module, to the output of the 'PULSE GENERATOR' module. Sketch this output in your **logbook**. Note the effect of adjusting the DC message signal level on the *mark* and *space* of the pulse generator's output.

5. Finally, move the second oscilloscope probe to the output of the 'TUNEABLE LPF' module. Set the 'GAIN' potentiometer to the 12 o'clock position and the 'f_c' potentiometer to its furthest clockwise position. Slowly adjust the 'f_c' potentiometer in an anti-clockwise direction until the output of the 'TUNEABLE LPF' module is a steady DC voltage for all positions of the potentiometer in the 'VARIABLE DCV' module. Notice that the DC voltage output by the 'TUNEABLE LPF' module varies with the voltage of the DC message signal generated by the 'VARIABLE DCV' module. The ZCD is therefore achieving

FM demodulation. With reference to your sketches, explain in your **logbook** how the ZCD works and how it gets its name.

6. Instead of a DC voltage, use the 2 kHz sine wave generated by the 'MASTER SIGNALS' module as the message signal. Move the oscilloscope's *first* probe from the output of the 'VCO' module to its input in order to check that the sinusoidal message signal is correctly FM modulated and demodulated.

7. Finally, use the microphone and headphones to perform the FM modulation of speech.

4.3 Frequency Shift Keying

In the same way that the digital version of AM is ASK, the digital version of FM is *Frequency Shift Keying* (FSK). While ASK uses the bit values to select between different *amplitudes* for the carrier, FSK uses the bit values to select between different *frequencies* for the carrier, as shown in Figure 25.

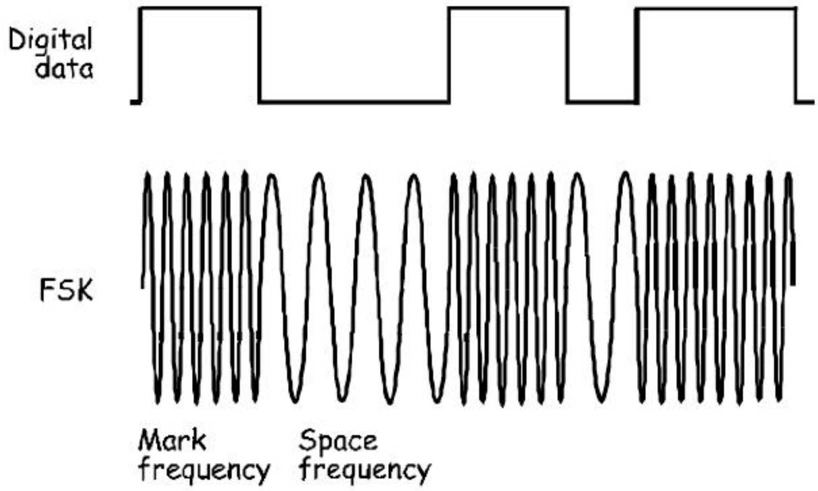

Figure 25

1. You can build an FSK modulator by making the connections shown in Figure 26.

2. Set the switch in the 'VCO' module to the 'LO' position and set the 'GAIN' potentiometer to its 12 o'clock position.

3. Make sure that the oscilloscope is triggering on the channel that is connected to the 'SYNC' output of the 'SEQUENCE GENERATOR' module. Also, make sure that both channels of the oscilloscope are DC coupled. Take a look at the digital signal provided by the 'LINE CODE' output of the 'SEQUENCE GENERATOR' module.

4. Now, connect a third oscilloscope probe to the output of the 'VCO' module. If you don't have a third probe available, then move the probe from the 'LINE CODE' output of the 'SEQUENCE GENERATOR' module.

5. ✎ Adjust the 'FREQ' potentiometer and see how the frequency of the carrier is keyed by the digital signal. Sketch the relationship between the digital signal and the modulated carrier in your **logbook**. When you are finished, leave the 'FREQ' potentiometer in its 2 o'clock position. With the 'FREQ' potentiometer in this position, write the frequencies that correspond to bit values of 0 and 1 in your **logbook**. You may find the oscilloscope's 'RUN/STOP' button to be helpful for capturing a clean image of the FSK signal.

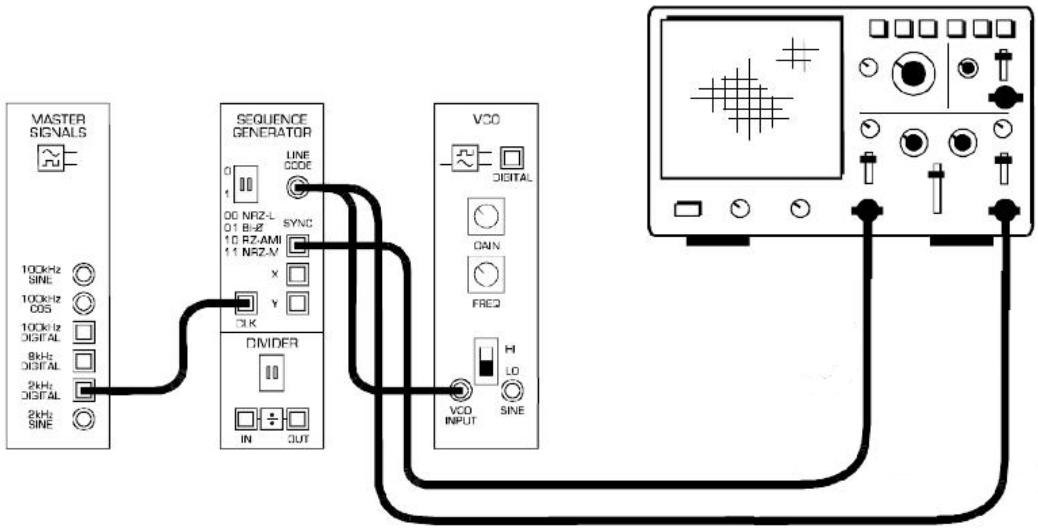

Figure 26

6. You could perform FSK demodulation using the same method as Section 4.2. However, this would require two comparators: one for the ZCD and one to reconstruct the digital signal. Since the ETT-101 has only one comparator, you need to come up with another method for demodulation. Here's a hint: think about how you could remove one or other of the frequencies from the FSK signal and what the result would look like. You should be left with a signal that can be demodulated using one of the techniques we considered earlier. Build your demodulator and draw a block diagram for it in your **logbook**. Sketch the signal in the time domain at each stage of your demodulator.

Appendix: ETT-101 System Conventions

The front panel of the EMONA Telecoms-Trainer 101 (ETT-101) has been laid out following a series of front panel conventions. All ETT-101 modules, for example the module shown in Figure 27, conform to the following mechanical and electrical conventions:

- Signal interconnections are made via front panel, 2mm sockets.
- Sockets on the **LEFT HAND SIDE** are for signal **INPUTS**. All inputs are high impedance, either 10k ohms or 56k ohms depending on the module, in order to reduce effects when connections are made and broken.
- Sockets on the **RIGHT HAND SIDE** are for signal **OUTPUTS**. All analog outputs are low impedance, typically 330 ohms. Again, this is to reduce effects when connections are made and broken. Digital outputs are typically 47 ohms.
- ROUND sockets, "○", are only for ANALOG signals. ANALOG signals are typically held near the ETT-101 standard reference level of 4V pk-pk.
- SQUARE sockets, "□", are only for DIGITAL signals. DIGITAL level signals are TTL level, 0 to 5 V.
- ROUND sockets labelled GND, "○", are common, or system GROUND.

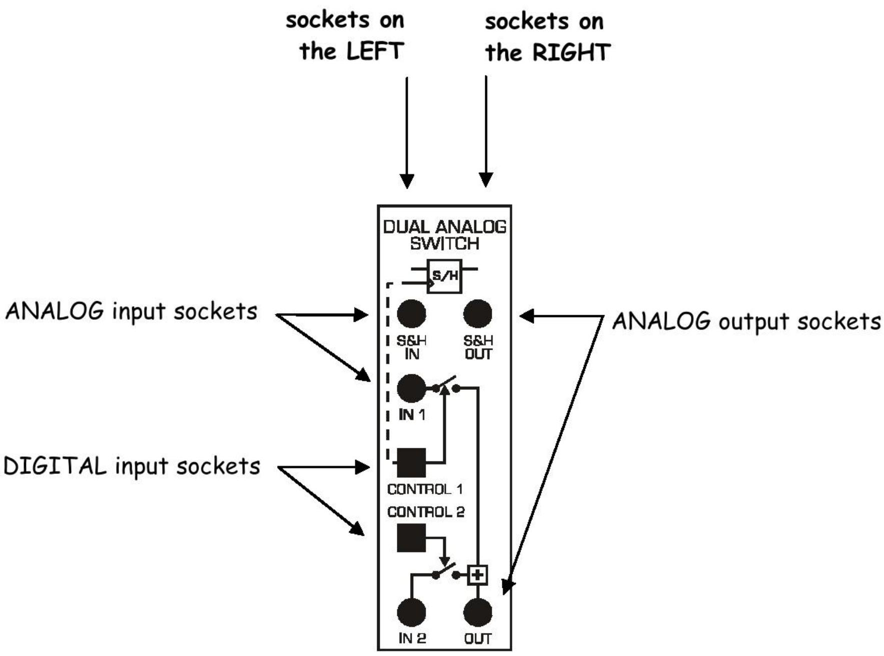

Figure 27

⇨ **Warning:** Each of the scope probe has a red lead and a black lead. Plug the **black lead** to the ground (GND) socket. It will not work if you plug the red lead to the GND.

⇨ **Note:** There are two GND sockets at the middle of the trainer and one GND socket at the 'VARIABLE DCV' module. It is possible to create more than 3 GNDs using the jumpers provided.