The Sampling Theorem:

Aim:

Experimental verification of the sampling theorem; sampling and message reconstruction (interpolation).

Experimental Procedure:

Taking Samples:

In the first part of the experiment you will set up the arrangement illustrated in Figure 1. Conditions will be such that the requirements of the Sampling Theorem, not yet given, are met. The message will be a single audio tone.
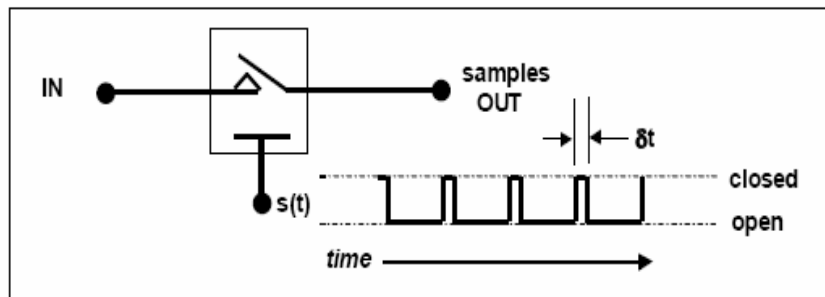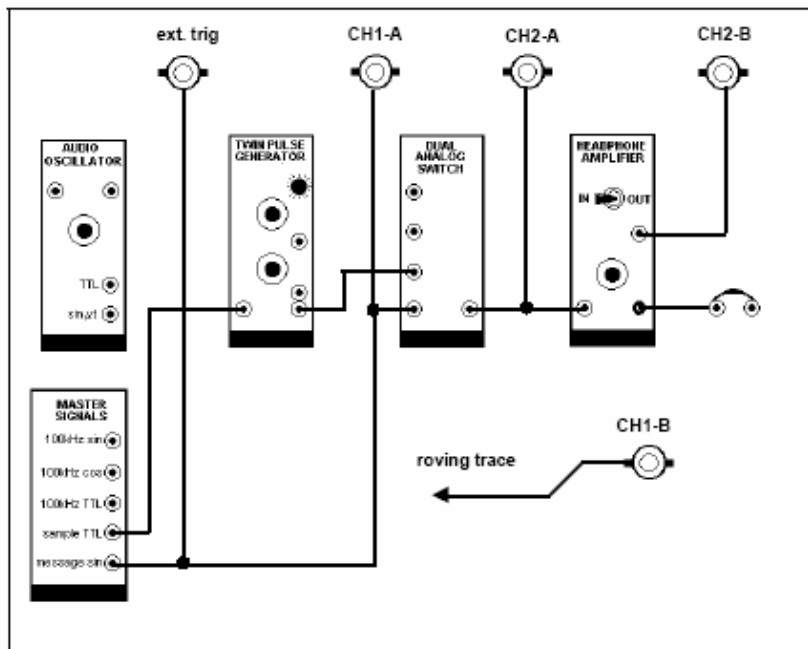

Figure 1: an analog sampler

To model the arrangement of Figure 1 with TIMS the modules required are a TWIN PULSE GENERATOR (only one pulse is used), to produce s(t) from a clock signal, and a DUAL ANALOG SWITCH (only one of the switches is used). The TIMS model is shown in Figure 2 below.

Figure 2: the TIMS model of Figure 1

T1 patch up the model shown in Figure 2 above. Include the oscilloscope connections. Note the oscilloscope is externally triggered from the message.

T2 adjust the width of the pulse from the TWIN PULSE GENERATOR with the pulse width control. The pulse is the switching function s(t), and its width is δt. You should be able to reproduce the sampled waveform of Figure 3.
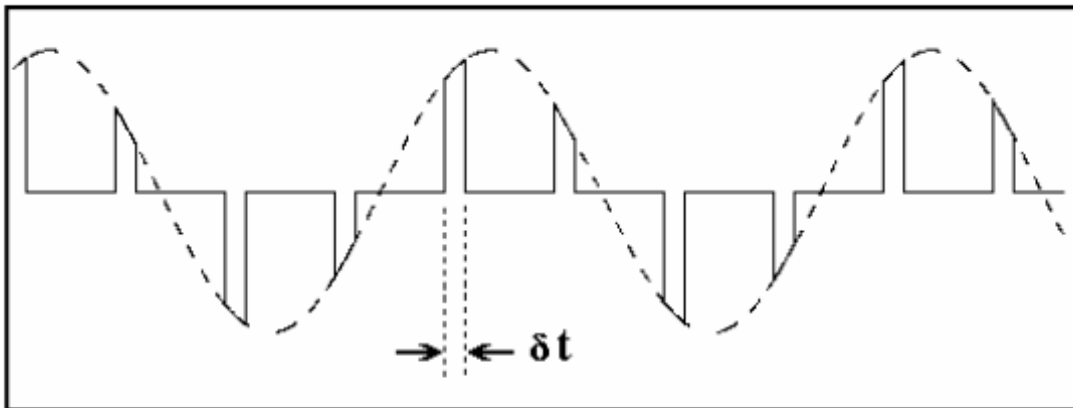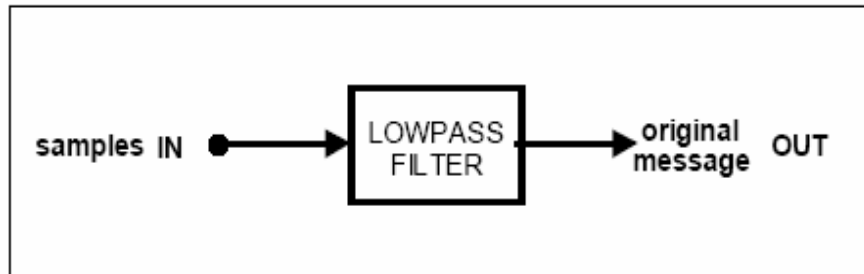

Figure 3: four samples per period of a sine wave.

<u>Why the message and the samples are stationary on the screen?</u>

<u>Reconstruction / Interpolation:</u>

The reconstruction circuitry is illustrated in Figure 4.

Figure 4: reconstruction circuit.

You can confirm that it recovers the message from the samples by connecting the output of the DUAL ANALOG SWITCH to the input of the 3 kHz LPF in the HEADPHONE AMPLIFIER module, and displaying the output on the oscilloscope.

T3 connect the message samples, from the output of the DUAL ANALOG SWITCH, to the input of the 3 kHz LPF in the HEADPHONE AMPLIFIER module, as shown in the patching diagram of Figure 2.

T4 switch to CH2-B and there is the message. Its amplitude may be a little small, so use the oscilloscope CH2 gain control. If you choose to use a BUFFER AMPLIFIER, place it at the output of the LPF. **<u>Why not at the input ?</u>**

T5 vary the width of the samples, and report the consequences as observed at the filter output

<u>Reconstruction Filter Bandwidth:</u>

T6 replace the 2 kHz message from the MASTER SIGNALS module with one from an AUDIO OSCILLATOR. In the first instance set the audio oscillator to about 2 kHz, and observe CH1-A and CH2-A simultaneously as you did in an earlier Task. You will see that the display is quite different.

<u>What change do you see in the oscilloscope display?</u>

T7 change the oscilloscope triggering to the sample clock. Report results.

T8 return the oscilloscope triggering to the message source. Try fine adjustments to the message frequency (sub-multiples of the sampling rate).

T9 look again at the reconstructed message on CH2-B. Lower the message frequency, so that if any distortion products are present (harmonics of the message) they will pass via the 3 kHz LPF.

## To find the minimum sampling rate:

MDSDR

In the procedure to follow you are going to report when it is just visibly obvious, in the time domain, when a single sinewave has been corrupted by the presence of another. You will use frequencies which will approximate those present during a later part of the experiment.

The frequencies are:

- wanted component - 3 kHz
- unwanted component - 4 kHz

Suppose initially the amplitude of the unwanted signal is zero volt. While observing the wanted signal, in the time domain, how large an amplitude would the unwanted signal have to become for its presence to be (just) noticed ?

A knowledge of this phenomenon will be useful to you throughout your career. An estimate of this amplitude ratio will now be made with the model illustrated in Figure 5.
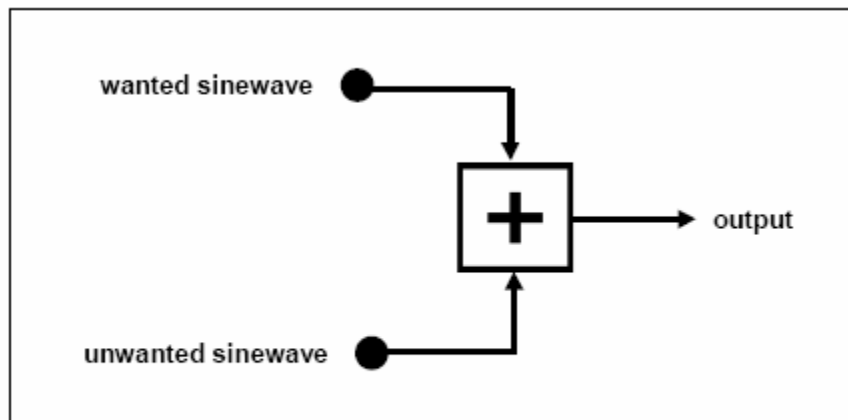
Figure 5: corruption measurement

T10 obtain a VCO module. Set the 'FSK - VCO' switch, located on the circuit board, to 'VCO'. Set the front panel 'HI - LO' switch to 'LO'. Then plug the module into a convenient slot in the TIMS unit.

T11 model the block diagram of Figure 5. Use a VCO and an AUDIO OSCILLATOR for the two sinewaves. Reduce the unwanted signal to zero at the ADDER output. Set up the wanted signal output amplitude to say 4 volt peak-to-peak. Trigger the oscilloscope to the source of this signal. Increase the amplitude of the unwanted signal until its presence is

just obvious on the oscilloscope. Measure the relative amplitudes of the two signals at the ADDER output. This is your MDSDR - the maximum detectable signal-to-distortion ratio. It would typically be quoted in decibels.

Minimum Sampling Rate measurement:

T12 remove the patch lead from the 8.333 kHz SAMPLE CLOCK source on the MASTER SIGNALS module, and connect it instead to the VCO TTL OUTPUT socket. The VCO is now the sample clock source.

T13 use the FREQUENCY COUNTER to set the VCO to 10 kHz or above.

T14 use the FREQUENCY COUNTER to set the AUDIO OSCILLATOR to f_c, the edge of the 3 kHz LPF passband.

T15 synchronize the oscilloscope to the message, direct from the AUDIO OSCILLATOR, and confirm that the message being sampled, and the reconstructed message, are identical in shape and frequency (the difference in amplitudes is of no consequence here).

T16 whilst continuing to monitor both the message and the reconstructed message, slowly reduce the sampling rate (the VCO frequency). As soon as the message shows signs of distortion (aliasing distortion), increase the sampling rate until it just disappears. The sampling rate will now be the minimum possible.

Questions:

1) Plot the spectrum of the sampled signal

2) even if the signal to be sampled is already bandlimited, why is it good practice to include an anti-aliasing filter ?

3) consider a sampling scheme as illustrated in Figure 1. The sampling rate is determined by the distance between the pulses of the switching function s(t). Assume the message was reconstructed using the scheme of Figure 4.
Suppose the pulse rate was slowly increased, whilst keeping the pulse width fixed.
Describe and explain what would be observed at the lowpass filter output.

4) Apart from varying the time interval between samples, what effect upon the message reconstruction does the sample width have ?