Name: ______________________

Class: ______________________

11 - Sampling and reconstruction

Experiment 11 – Sampling and reconstruction

Preliminary discussion
So far, the experiments in this manual have concentrated on communications systems that transmit analog signals. However, digital transmission is fast replacing analog in commercial communications applications. There are several reasons for this including the ability of digital signals and systems to resist interference caused by electrical noise.

Many digital transmission systems have been devised and several are considered in later experiments. Whichever one is used, where the information to be transmitted (called the *message*) is an analog signal (like speech and music), it must be converted to digital first. This involves *sampling* which requires that the analog signal's voltage be measured at regular intervals.

Figure 1a below shows a pure sinewave for the message. Beneath the message is the digital *sampling signal* used to tell the sampling circuit when to measure the message. Beneath that is the result of "naturally" sampling the message at the rate set by the sampling signal. This type of sampling is "natural" because, during the time that the analog signal is measured, any change in its voltage is measured too. For some digital systems, a changing sample is unacceptable. Figure 1b shows an alternative system where the sample's size is fixed at the instant that the signal measured. This is known as a *sample-and-hold* scheme (and is also referred to as *pulse amplitude modulation*).

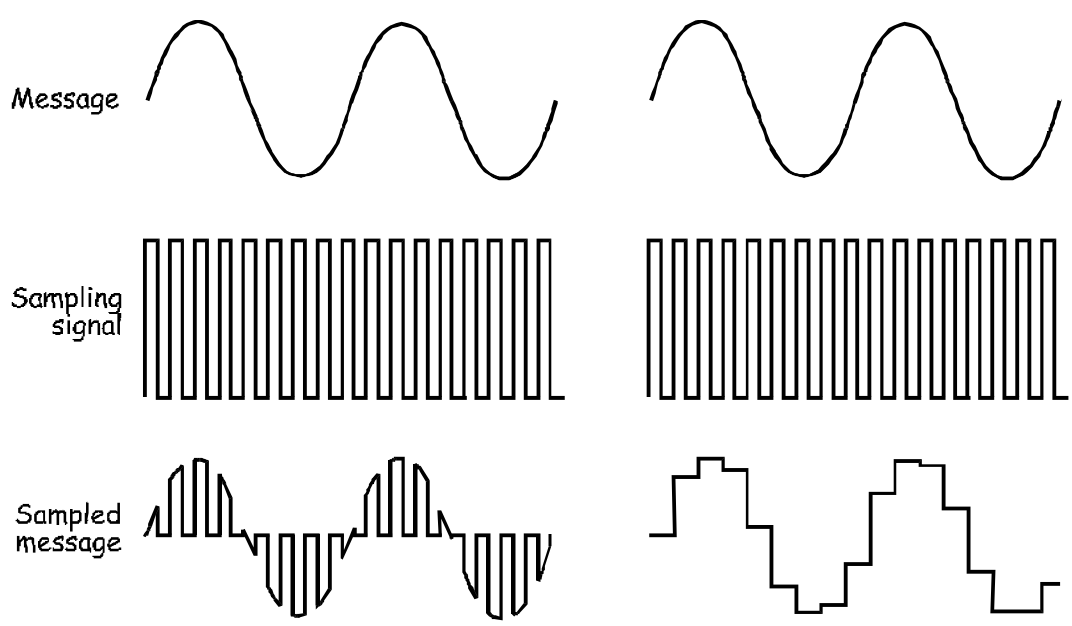


Figure 1a **Figure 1b**

Regardless of the sampling method used, by definition it captures only pieces of the message. So, how can the sampled signal be used to recover the whole message? This question can be answered by considering the mathematical model that defines the sampled signal:

Sampled message = the sampling signal × the message

As you can see, sampling is actually the multiplication of the message with the sampling signal. And, as the sampling signal is a digital signal which is actually made up of a DC voltage and many sinewaves (the fundamental and its harmonics) the equation can be rewritten as:

Sampled message = (DC + fundamental + harmonics) × message

When the message is a simple sinewave (like in Figure 1) the equation's solution (which necessarily involves some trigonometry that is not shown here) tells us that the sampled signal consists of:

- A sinewave at the same frequency as the message
- A pair of sinewaves that are the sum and difference of the fundamental and message frequencies
- Many other pairs of sinewaves that are the sum and difference of the sampling signals' harmonics and the message

This ends up being a lot of sinewaves but one of them has the same frequency as the message. So, to recover the message, all that need be done is to pass the sampled signal through a low-pass filter. As its name implies, this type of filter lets lower frequency signals through but rejects higher frequency signals.

That said, for this to work correctly, there's a small catch which is discussed in Part C of the experiment.

The experiment

In this experiment you'll use the Emona Telecoms-Trainer 101 to sample a message using natural sampling and a sample-and-hold scheme. You'll then reconstruct the message from the sampled signal and examine the effect of aliasing.

It should take you about 50 minutes to complete this experiment.

Equipment

- Emona Telecoms-Trainer 101 (plus power-pack)
- Dual channel 20MHz oscilloscope
- two Emona Telecoms-Trainer 101 oscilloscope leads
- assorted Emona Telecoms-Trainer 101 patch leads

Part A – Sampling a simple message
The Emona Telecoms-Trainer 101 has a Dual Analog Switch module that has been designed for sampling. This part of the experiment lets you use the module to sample a simple message using two techniques.

Procedure

1. Gather a set of the equipment listed above.

2. Connect the set-up shown in Figure 2 below.

   Note: Insert the black plugs of the oscilloscope leads into a ground (*GND*) socket.

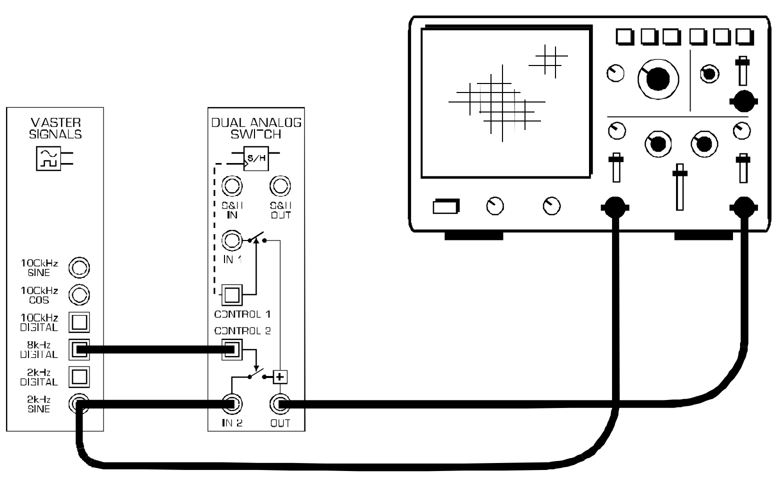


Figure 2

The set-up in Figure 2 can be represented by the block diagram in Figure 3 below. It uses an electronically controlled switch to connect the message signal (the *2kHz SINE* output from the Master Signals module) to the output. The switch is opened and closed by the *8kHz DIGITAL* output of the Master Signals module.

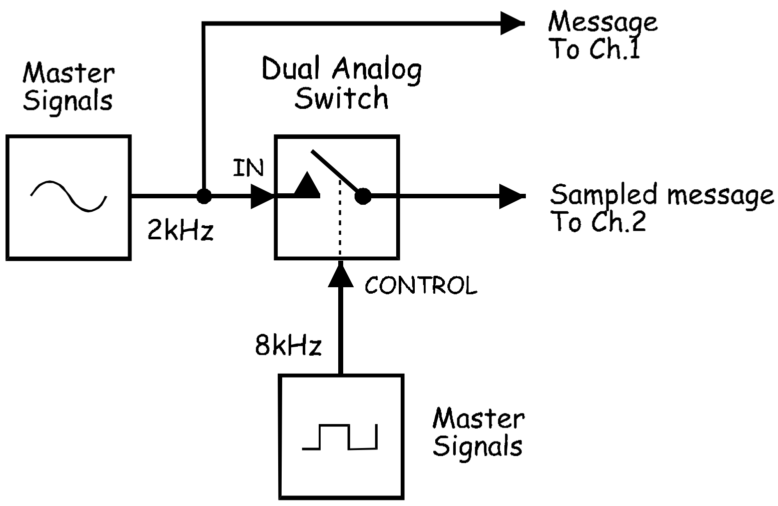


Figure 3

3. Set up the scope per the instructions in Experiment 1. Ensure that:

   - the *Trigger Source* control is set to the *CH1* (or *INT*) position.
   - the *Mode* control is set to the *CH1* position.

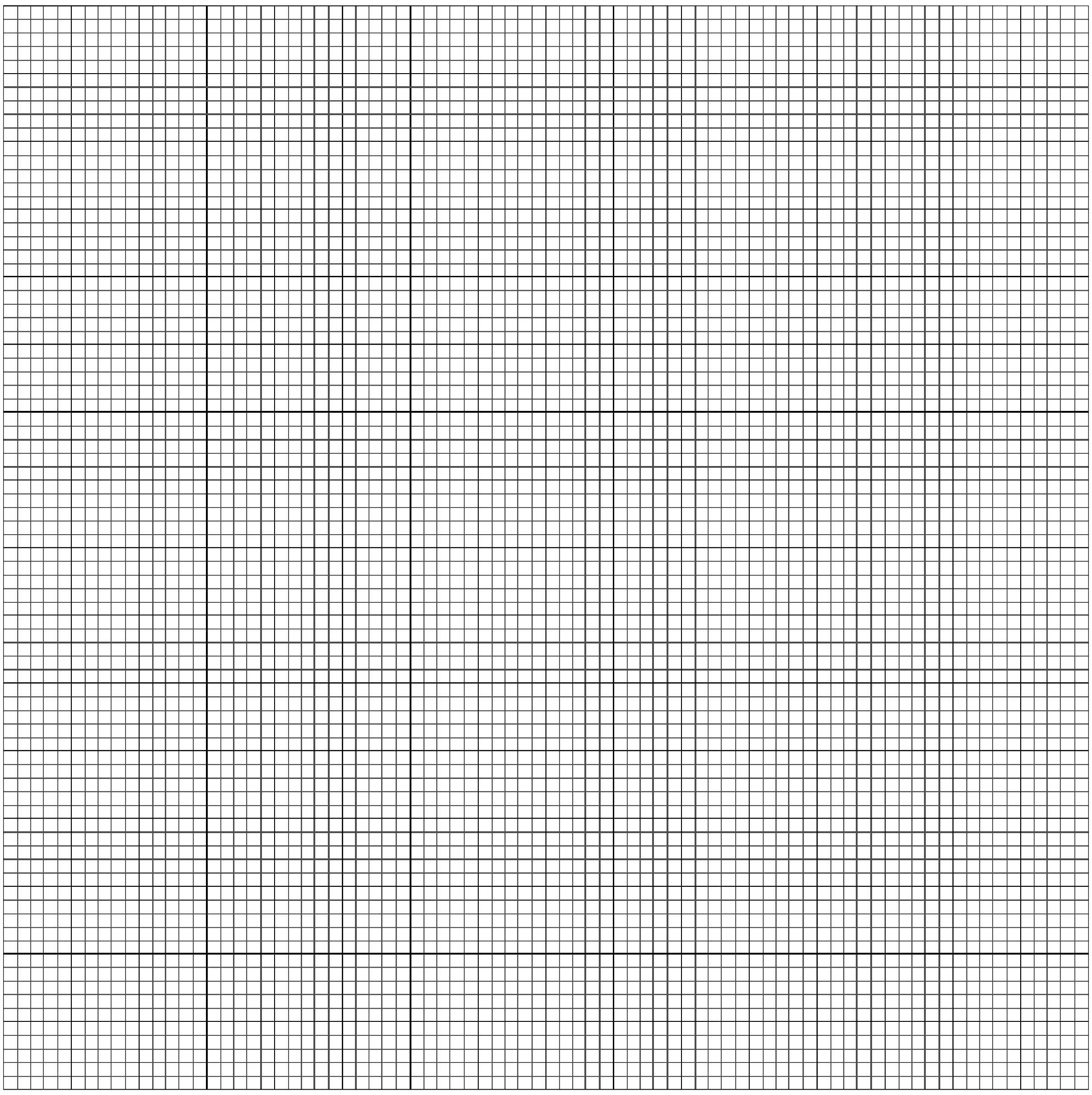
4. Adjust the scope's *Timebase* control to view two or so cycles of the Master Signals module's *2kHz SINE* output.

5. Set the scope's *Mode* control to the *DUAL* position to view the sampled message out of the Dual Analog Switch module as well as the message.

6. Set the scope's *Vertical Attenuation* controls to the *1V/div* position.

7. Draw the two waveforms to scale in the space provided on the next page leaving room to draw a third waveform.

   Tip: Draw the message signal in the upper third of the graph and the sampled signal in the middle third.



Question 1
What type of sampling is this an example of?

- ☐ Natural ✓
- ☐ Sample-and-hold

Question 2

What two features of the sampled signal confirm this?

1) The sample voltages change during sampling.

2) The signal's voltage returns to zero volts between the samples.

Ask the instructor to check your work before continuing.

8. Modify the set-up as shown in Figure 4 below.

Before you do…

The set-up in Figure 4 below builds on the set-up that you've already wired so don't pull it apart. To highlight the changes that we want you to make, we've shown your existing wiring as dotted lines.

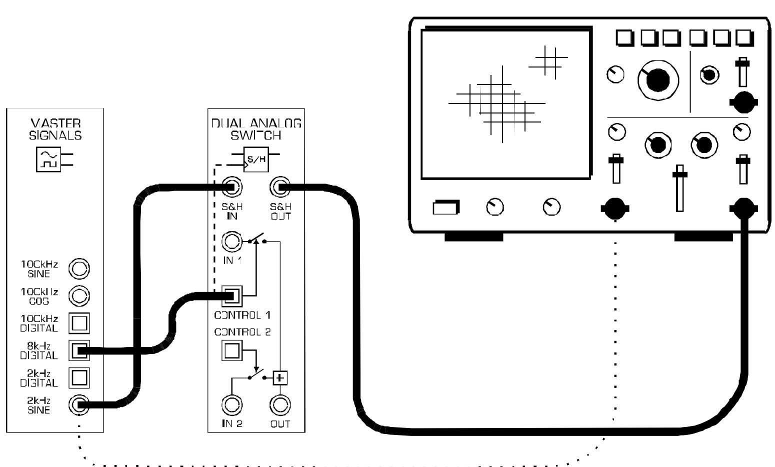


Figure 4

The set-up in Figure 4 can be represented by the block diagram in Figure 5 below. The electronically controlled switch in the original set-up has been substituted for a sample-and-hold circuit. However, the message and sampling signals remain the same (that is, a 2kHz sinewave and an 8kHz pulse train).

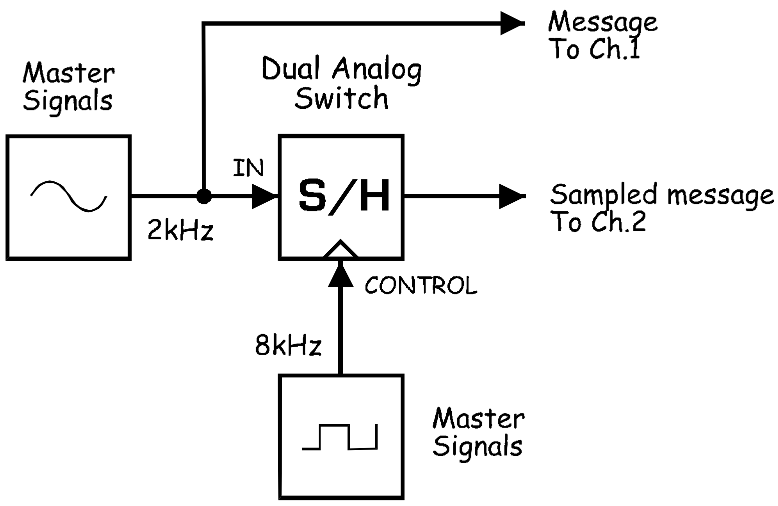


Figure 5

9. Draw the new sampled message to scale in the space that you left on the graph paper.

Question 3
What two features of the sampled signal confirm that the set-up models the sample-and-hold scheme?

1) The sample voltages don't change during sampling.

2) There's no space between the samples.

Ask the instructor to check your work before continuing.

Part B – Sampling speech

This experiment has sampled a 2kHz sinewave. However, the message in commercial digital communications systems is much more likely to be speech and music. The next part of the experiment lets you see what a sampled speech signal looks like.

10. Disconnect the plugs to the Master Signals module's *2kHz SINE* output.

11. Connect them to the Speech module's output as shown in Figure 6 below.

    Remember: Dotted lines show leads already in place.

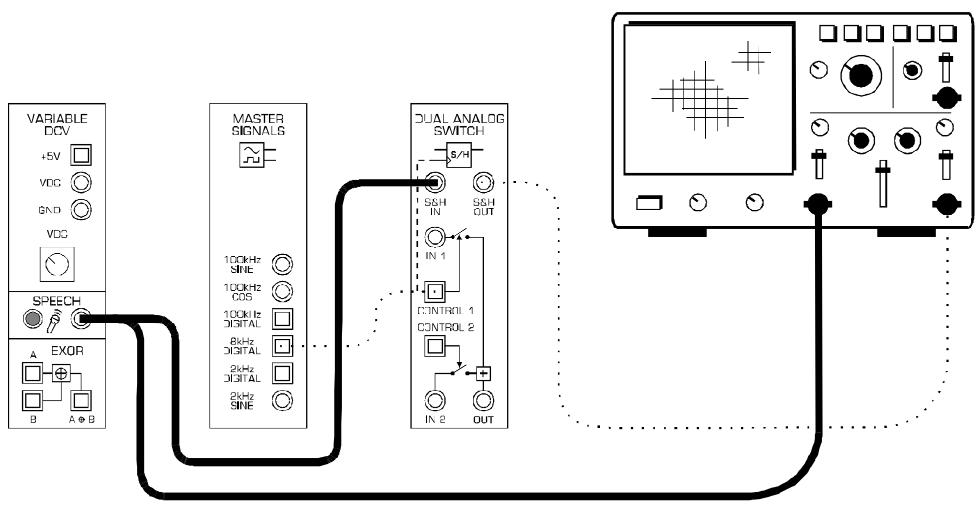


Figure 6

12. Set the scope's *Timebase* control to the *2ms/div* position.

13. Talk, sing or hum while watching the scope's display.

Ask the instructor to check your work before continuing.

Part C – Reconstructing a sampled message

Recall that the sampled message is made up of many sinewaves. Importantly, for every sinewave in the message, there's a sinewave in the sampled message with the same frequency. So "reconstructing" the original message involves passing the sampled message signal through a low-pass filter. This lets the sinewave (or sinewaves) with the same frequency as the message through while rejecting the other sinewaves. The next part of the experiment lets you do this.

14. Return the scope's *Timebase* control to the *0.1ms/div* position.

15. Locate the Tuneable Low-pass Filter module and set its *Gain* control to about the middle of its travel.

16. Turn the Tuneable Low-pass Filter module's *Cut-off Frequency Adjust* control fully anti-clockwise.

17. Disconnect the plugs to the Speech module's output.

18. Modify the set-up as shown in Figure 7 below.

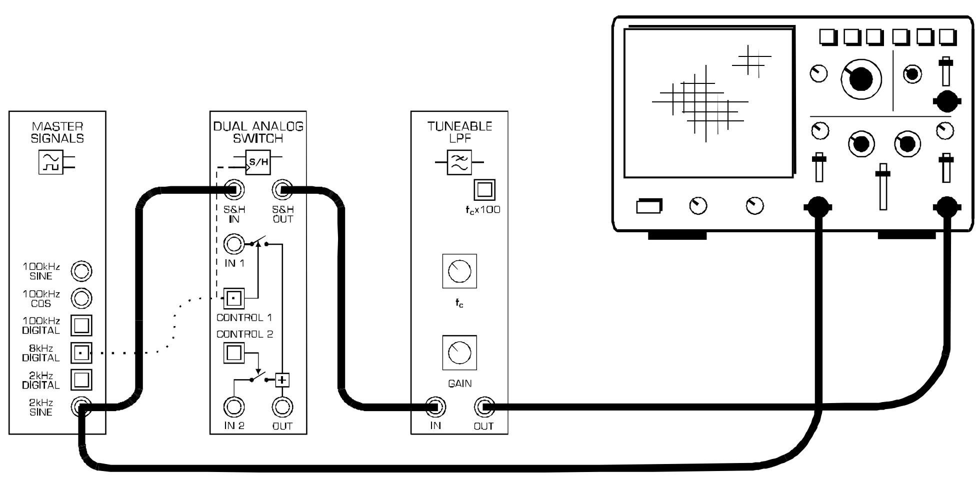


Figure 7

The set-up in Figure 7 can be represented by the block diagram in Figure 8 below. The Tuneable Low-pass Filter module is used to recover the message. The filter is said to be "tuneable" because the point at which frequencies are rejected (called the *cut-off frequency*) is adjustable.

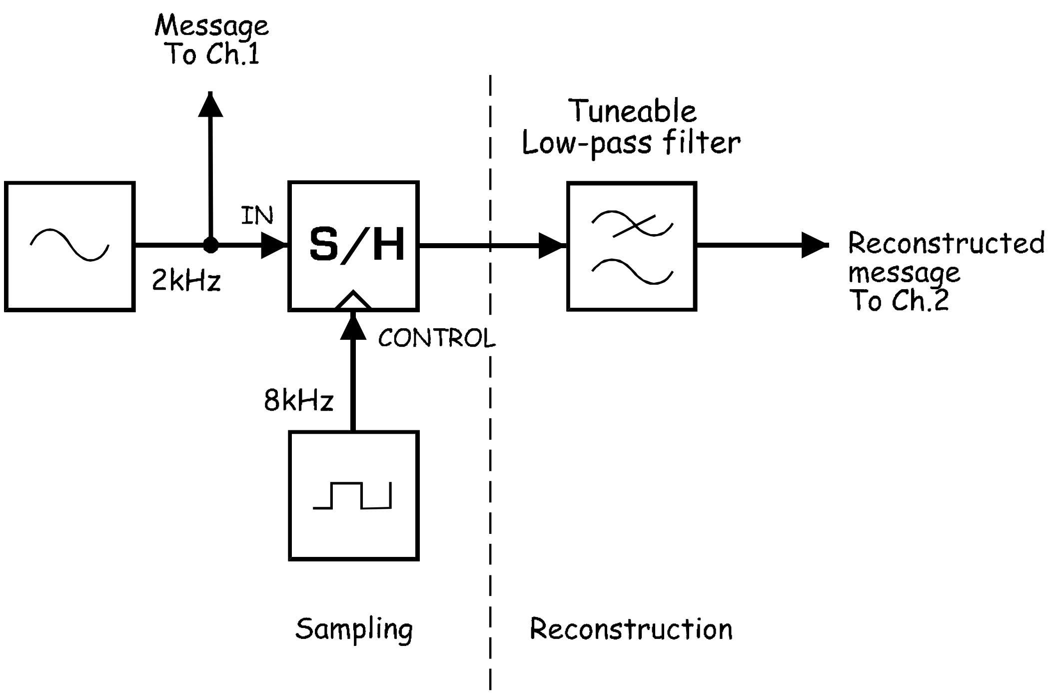


Figure 8

At this point there should be nothing out of the Tuneable Low-pass Filter module. This is because it has been set to reject almost all frequencies, even the message. However, the cut-off frequency can be increased by turning the module's *Cut-off Frequency Adjust* control clockwise.

19. Slowly turn the Tuneable Low-pass Filter module's *Cut-off Frequency* control clockwise and stop the moment the message signal has been reconstructed.

Ask the instructor to check your work before continuing.

Part D – Aliasing

At present, the filter is only letting the message signal through to the output. It is comfortably rejecting all of the other sinewaves (called *aliases*) that make up the sampled message. This is only possible because the frequency of these other sinewaves is high enough. But, this isn't an accident. Their frequency is determined by the sampling rate (that is, the sampling signal's frequency).

To explain, recall that the sampled message consists of the following:

- A sinewave at the same frequency as the message
- A pair of sinewaves that are the sum and difference of the fundamental and message frequencies
- Many other pairs of sinewaves that are the sum and difference of the sampling signals' harmonics and the message

In your set-up, a 2kHz sinewave message is sampled using an 8kHz sampling signal. That being the case, the lowest frequency components of the sampled message are:

- 2kHz
- 6kHz
- 10kHz

Now, suppose the frequency of the sampling signal is lowered. You'd still get the message but the frequency of the aliases would go down as well. So for example, if the sampling signal is 7kHz, the lowest frequency components of the sampled message become 2kHz, 5kHz and 9kHz. Clearly, if the sampling signal's frequency is low enough, one or more of the lower frequency aliases can pass through the filter along with the reconstructed message. Obviously, this would distort the reconstructed message which is a problem known as *aliasing*.

To avoid aliasing, the sampling signal's theoretical minimum frequency is twice the message frequency (or twice the highest frequency in the message if it contains more than one sinewave and is a baseband signal). This figure is known as the *Nyquist Sample Rate*. So for this set-up, the minimum sampling rate is 4kHz and the lowest frequency components would be 2kHz, 2kHz and 6kHz.

That said, filters aren't perfect. Their rejection of frequencies beyond the cut-off is gradual rather than instantaneous. So in practice the sampling signal's frequency needs to be a little higher than the Nyquist Sample Rate.

The next part of the experiment lets you vary the sampling signal's frequency to observe aliasing.

20. Locate the VCO module and set its *Frequency Adjust* control fully clockwise.

21. Set the VCO module's *Range* control to the *LO* position.

22. Modify the set-up as shown in Figure 9 below.

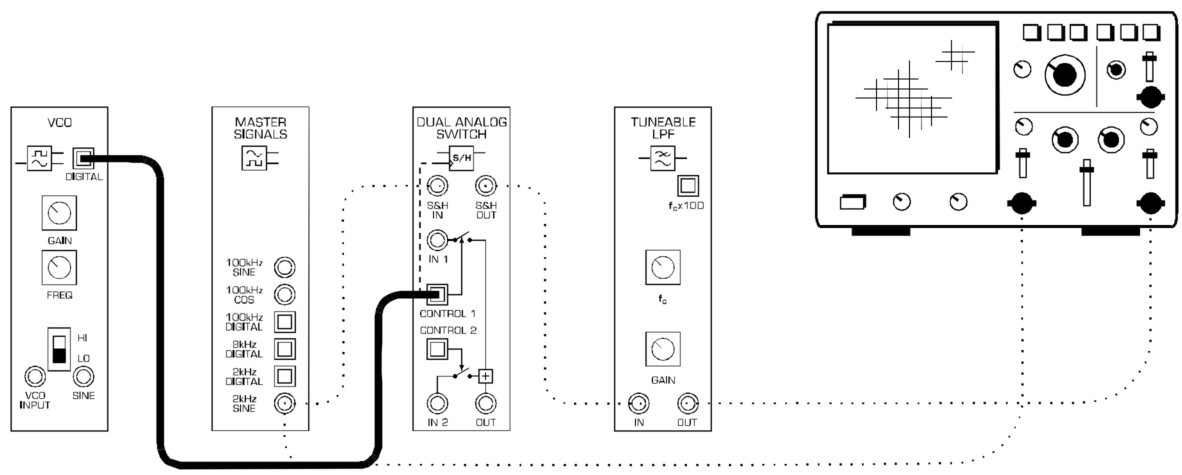


Figure 9

This set-up can be represented by the block diagram in Figure 10 below. Notice that the sampling signal is now provided by the VCO module which has a manually adjustable frequency.

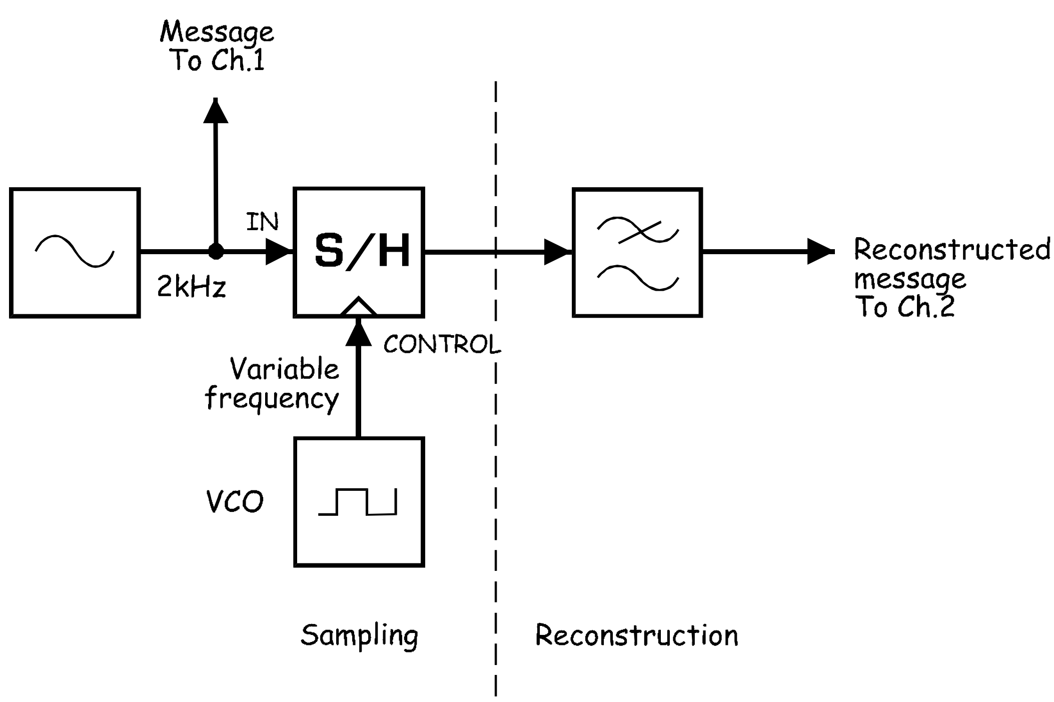


Figure 10

At this point, the sampling of the message and its reconstruction should be working normally.

23. Slowly reduce the frequency of the VCO module's output (by turning its *Frequency Adjust* control anti-clockwise) while watching the reconstructed message signal.

Question 4
What's the name of the distortion that appears when the VCO module's *Frequency Adjust* control is turned far enough?

Aliasing.

Question 5
Given the message is a 2kHz sinewave, what's the theoretical minimum frequency for the sampling signal? **Tip:** If you're not sure, see the notes on page 11-12.

4kHz (2 × 2kHz)

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24. Turn the VCO module's *Frequency Adjust* control clockwise and stop the moment the reconstructed message is no-longer distorted.

25. Connect the scope's Channel 1 input to the VCO module's *DIGITAL* output.

26. Set the scope's *Mode* control to the *CH1* position.

27. Adjust the scope's *Timebase* control to view two or so cycles of the VCO module's *DIGITAL* output.

28. Measure the signal's period and record this in Table 1 on the next page.

    Tip: If you're not sure how to measure the signal's period, see Experiment 1 (page 1-7).

29. Use the period to calculate and record the signal's frequency.

    Tip: If you're not sure how to calculate the signal's frequency, see Experiment 1 (page 1-8).

Table 1	Period	Frequency
VCO module's DIGITAL output		

Question 6
Why is the actual minimum sampling frequency higher than the theoretical minimum that you calculated for Question 5?

Because filters aren't perfect - their cut-off is not instantaneous.

Ask the instructor to check your work before finishing.