LAB 2 – Special Signals

Objectives

- Understand degradation of a digital signal as it passes through a system
- Use step and pulse signal inputs to measure system response
- Investigate sine wave distortion and cutoff
- Construct a digital detector

Background

Bandwidth is a term that has been in the engineering vocabulary for many decades. Its usage has extended over time, especially in the context of digital systems. It has become commonplace now to mean information transfer rate, and all Internet users know that broadband stands for fast, and better. There are highly competitive markets demanding top performance – ever higher speed whilst maintaining a low probability of corruption. However, as speed is increased, obstacles emerge in the form of noise, interference and signal distortion. At the destination these limitations become digital errors, resulting in pixelated images, and audio breaking up.

The most important consideration affecting the speed of a digital signal is the switching process to produce a change of state. The switching time can never be instantaneous in a physical system because of energy storage in electronic circuitry, cabling and connecting hardware. This energy lingers in stray capacitance and inductance that cannot be completely eliminated in wiring and in electronic components. The effect is just like inertia in a mechanical system.

In Part A we investigate how digital signals are distorted when a system's response is affected by inertia. In Parts B and C we use a step and an impulse input, respectively, to measure system response. In Part 4 we introduce the sinewave, and study the phenomena of clipping.

New Modules

- AUDIO OSCILLATOR
- SEQUENCE GENERATOR
- BASEBAND CHANNEL FILTERS
- UTILITIES

Part A – Digital Pulse Sequence

A.1 – The AUDIO OSCILLATOR

The AUDIO OSCILLATOR provides a sinusoidal output tunable from 300 Hz to 10 kHz (typical audio frequency range). It can also provide a digital output signal by converting the sine wave into a square wave. Note that this digital signal is referred to as a "TTL Signal", referring to the transistor-transistor logic source of such a signal.

1. Insert the AUDIO OSCILLATOR module into the TIMS rack (shown here in slot 2 of Figure 1).

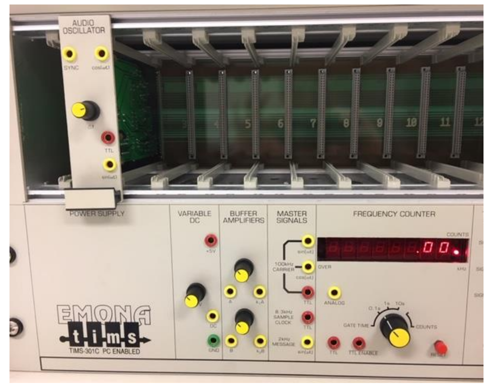


Fig 1 The AUDIO OSCILLATOR module shown inserted into Slot 2 of the TIMS rack.

2. The AUDIO OSCILLATOR output can be measured by connecting it to the frequency counter, and can be viewed by connecting it to the PicoScope.

- Connect the AUDIO OSCILLATOR TTL output to the TTL input of the FREQUENCY COUNTER.
- Also connect the AUDIO OSCILLATOR TTL output to Scope ChA. See Figure 2.
- The value on the FREQUENCY COUNTER is varied by adjusting the Δf knob on the AUDIO OSCILLATOR. Here it is 6.94k counts per second. For convenience, we will call this "6.94 kHz".

  Note: Technically a Hz is a cycle per second, referring to sinusoids, not count per second as we use it here. We will consider a kHz to also represent 1000 clock cycles per second.

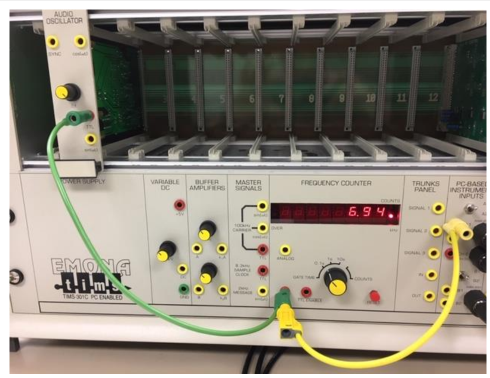


Fig 2 The oscillator is connected to the frequency counter and the scope.

3. Turn on the PicoScope

- Set the time scale to 200us/div. Note that the signal jumps around a lot. For triggering, select "single" that will take a freeze frame shot of the signal. This is referred to as "Single Shot Triggering". See Figure 3.
- Turn the triggering off again (Trigger = None) and also press the green triangle to get the scope running again.

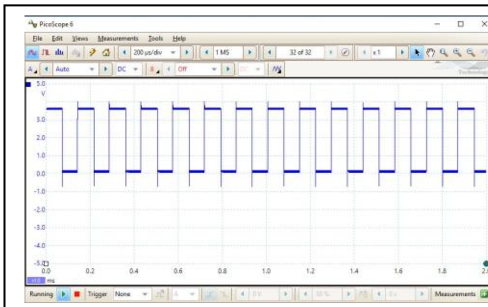


Fig 3 Single Shot Triggering used to display oscillator output.

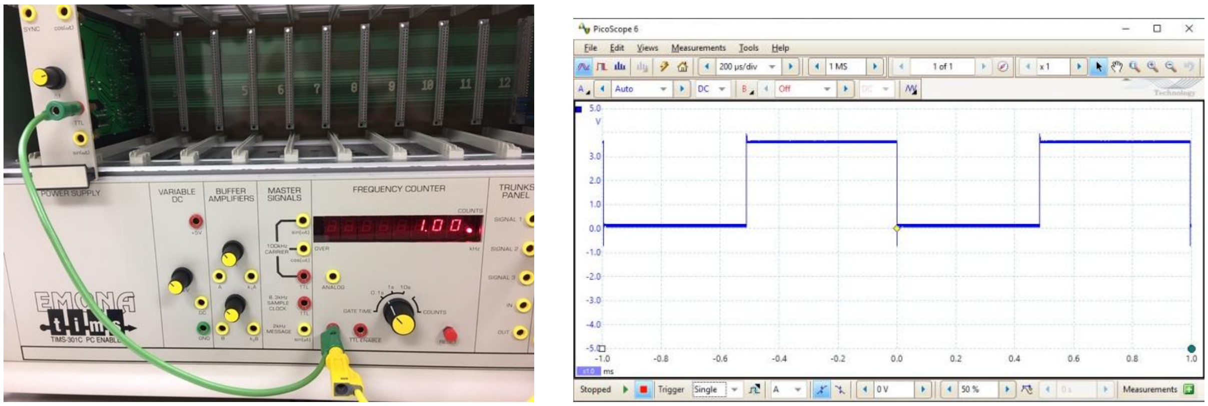


Fig 4 Δf knob adjustment achieves a 1 kHz signal seen on the counter and on the scope display.

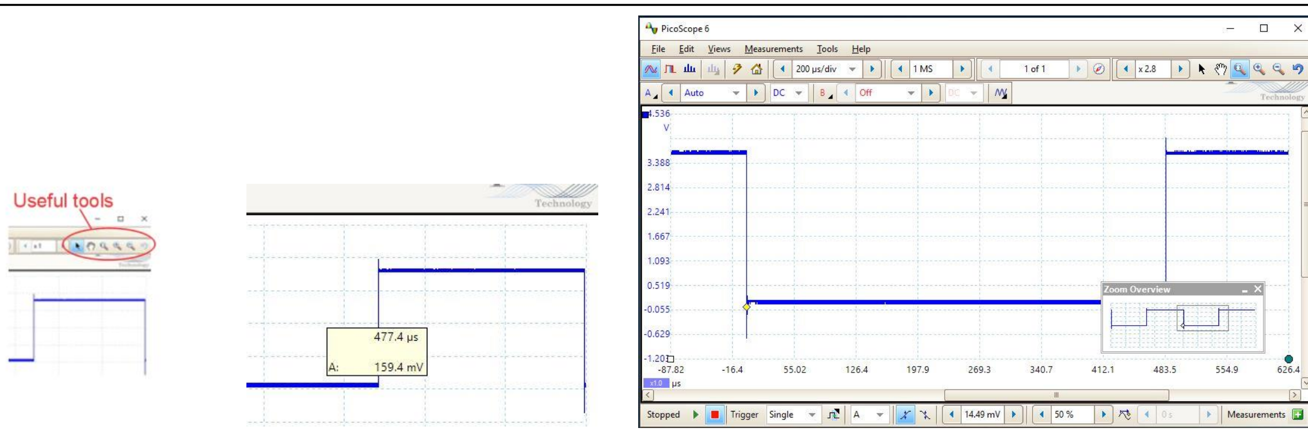
4. Adjust the AUDIO OSCILLATOR's Δf knob to achieve close to 1kHz on the counter. On the scope, perform single shot triggering to see what the clock signal looks like. See Figure 4.

5. Let's try out some useful tools on the upper right of the PicoScope window: the arrow, hand, and magnifying glass icons. See Figure 5a.

- Figure 5b shows the cursor location pop up when the arrow is clicked on the trace at a lower transition point.
- In Figure 5c, the "zoom window" magnifying glass does what you would expect. You can now use the arrow icon to more precisely read points on the scope.
- <u>Measure</u> the duration of one pulse width and one clock cycle and record it here:

Pulse Width = Around 0.5 mS ____________ Clock cycle = 1 kHz ____________

Note: the usual way to measure pulse width would be between the middles of the transitions. Here, this should correspond to half of a clock cycle.



(a) (b) (c)

Fig 5 Employ PicoScope tools to accurately determine the cycle duration

A.2 – The SEQUENCE GENERATOR

Using a common external clock signal, the SEQUENCE GENERATOR outputs two independent pseudorandom sequences **X** and **Y**. The **X** and **Y** sequences are available as either standard digital output (TTL output uses the red X and Y) or analog output.

1. Insert the SEQUENCE GENERATOR module into the TIMS rack (shown here in slot 4 of Figure 6).

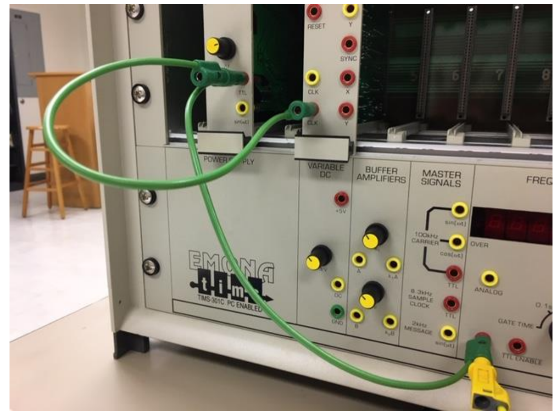


Fig 6 SEQUENCE GENERATOR module is shown inserted into Slot 4. The oscillator's TTL clock output is connected to the generator's input.

2. Referring to Figure 7a, connect the AUDIO OSCILLATOR TTL clock signal to:
- the SEQUENCE GENERATOR's red clock input (green lead)
- the FREQUENCY COUNTER (green lead)
- Scope ChA (yellow lead).

3. Connect the SEQUENCE GENERATOR's red X output to channel B for the scope. See Figure 7a (blue lead).

4. On the PicoScope, turn ChB on to Auto.
- Change the time base to 1ms/div for a better view of this digital signal.
- Employ single shot triggering. See Figure 7b.

5. Now employ the zoom and cursor features to read the briefest interval between consecutive transitions of the digital signal.
- Click on the green arrow to the left of the trigger to repeat the single shot triggering. Repeat this several times to ensure you are seeing a briefest transition.

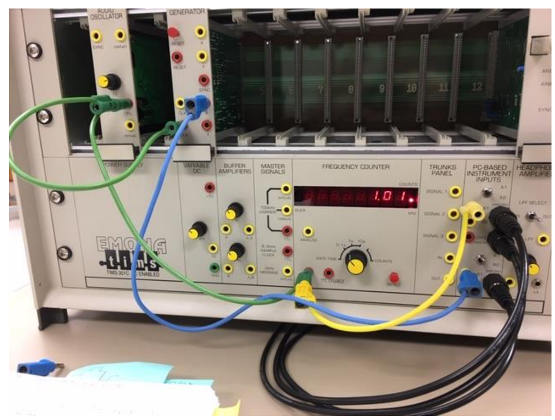
Record the minimum interval of the digital signal here:

1 mS

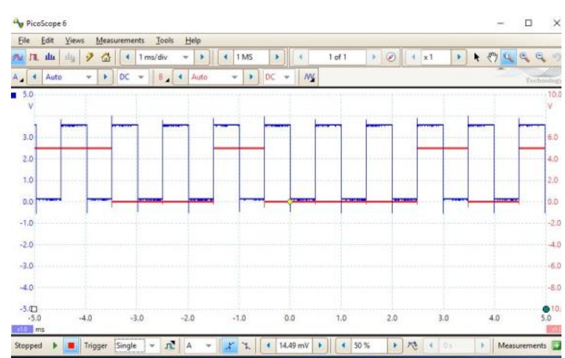
Question: How does this interval compare to the clock cycle duration? It is exactly one clock cycle

Convert the displayed portion of the digital signal to a binary code:

11001000101



(a)



(b)

Fig 7 (a) SEQUENCE GENERATOR connections and (b) the clock and digital signals displayed on the scope

A.3 – BASEBAND CHANNEL FILTERS

This module allows users to select from 4 channels. Channel 1 is simply a through connection, while channels 2, 3, and 4 are different low pass filters. These are abbreviated as BBLPF2, BBLPF3, and BBLPF4 (for BaseBand Low Pass Filter 2, etc.). The module description in the appendix provides insight into these filters.

In the current experiment, the filters represent response time limitations (i.e. the "system inertia" metaphor). We will examine the digital signal just before and just after the filter section.

1. Insert the BASEBAND CHANNEL FILTERS module into the TIMS rack (shown here in slot 6 of Figure 8)

2. Connect the modules as shown in Figure 8.

- It may be easiest to remove the connections from the previous experiment and start fresh.
- Connect the AUDIO OSCILLATOR TTL clock signal to:
  - the SEQUENCE GENERATOR's red clock input (green lead)
  - the FREQUENCY COUNTER (green lead)
- Connect the SEQUENCE GENERATOR's TTL clock output to the BASEBAND CHANNEL FILTERS input, and to Scope ChA (yellow leads).
- Connect the Baseband Channel Filters OUT to Scope Ch B (red lead).
- Select Channel 2 on the BASEBAND CHANNEL FILTERS module.

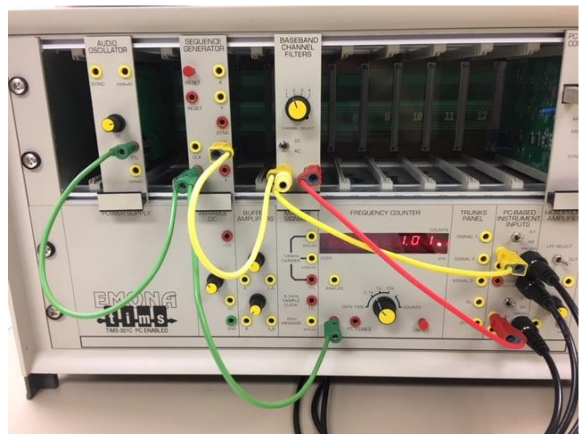


Fig 8 BASEBAND CHANNNEL FILTERS module is shown inserted into Slot 6.

3. Adjust the PicoScope as follows:

- Set the time scale to 1ms/div
- Employ AC coupling on Channel A
- Turn on channel B, with scaling set to +/-5V
- Employ AC coupling on Channel B

4. View several single shot trigger events. Figure 9 shows one such event.

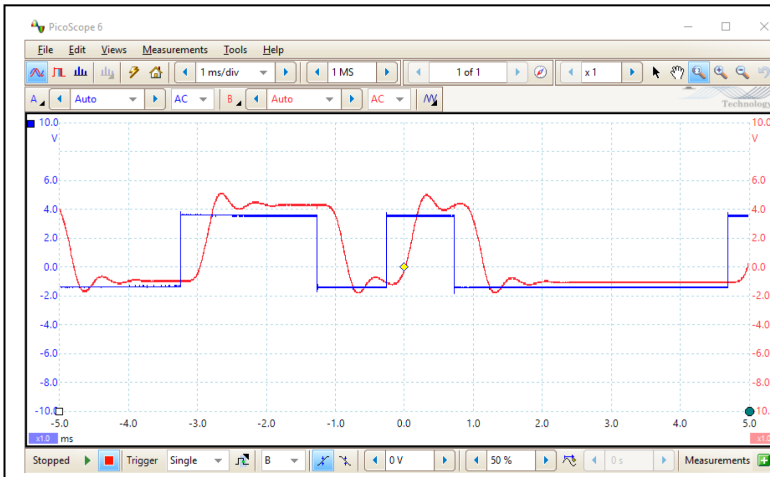


Fig 9 Typical digital signal before and after passing through BBLPF2.

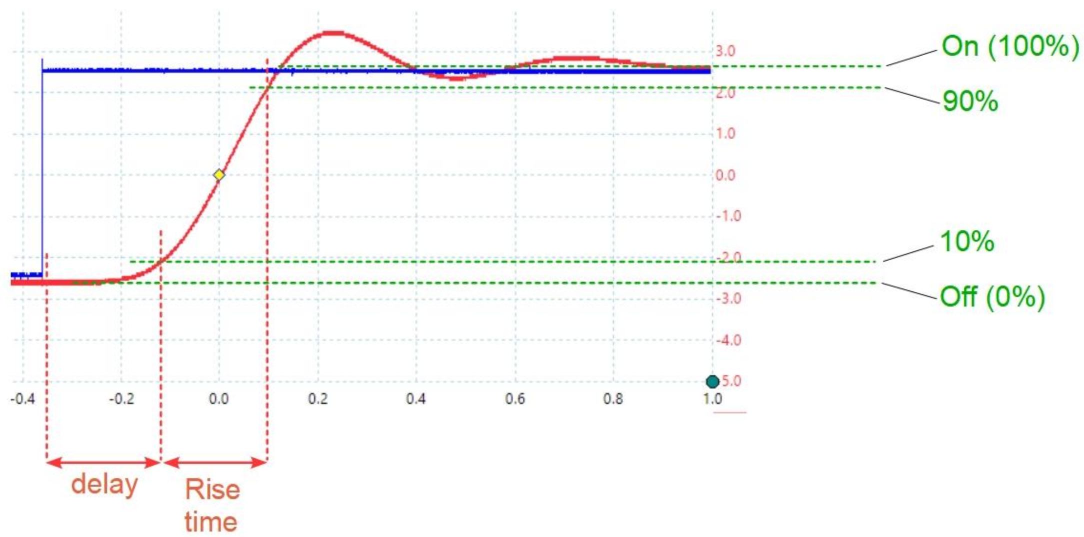


Fig 10 Delay and rise times for a digital signal transition from low to high.

Measuring Transition Time

When an actual digital signal transitions from a low state to a high state (or from a high state to a low state) it cannot do so instantly. There is a rise time associated with how long it takes to go from the low state (a digital logic "zero") to the high state (a digital logic "one"). Likewise, there is a fall time associated with going from the high to the low state. Together, rise time and fall time are referred to as "transition times".

Figure 10 indicates that rise time is measured from the time where the signal rises 10% of its transition time above the low state floor until it gets to 90% of its transition to the final voltage after the signal steadies out. Fall time is measured in a similar way. The figure also shows delay introduced to the signal by the system.

5. Use what you've learned so far to measure the delay and the rise and fall times of the digital signal passing through different filters. After making measurements at 1kHz, change the clock frequency to 1.5kHz and re-measure the rise and fall times.

Table 1: Delay and Transition Times for Sequence Data

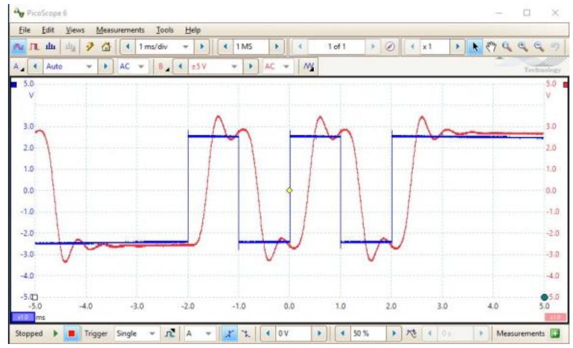
	BBLPF2		BBLPF3		BBLPF4	
	1 kHz	1.5 kHz	1 kHz	1.5 kHz	1 kHz	1.5 kHz
Delay	0.27 mS		0.262 mS		0.834 mS	
Rise time (10%-90%)	0.194 mS		0.286 mS		0.183 mS	
Fall time (90%-10%)	0.205 mS		0.286 mS		0.183 mS	

6. Figure 13(a) and (b) shows some typical results of the delay and distortion introduced by the low pass filter system.

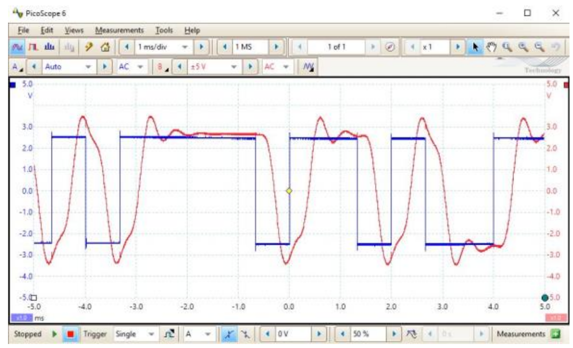
7. Increase Δf on the Audio Oscillator until the frequency counter reads 2.0 kHz. Observe the output (see Figure 13c).

8. Continue increasing Δf. At what frequency are you no longer able to accurately discern the digital signal?

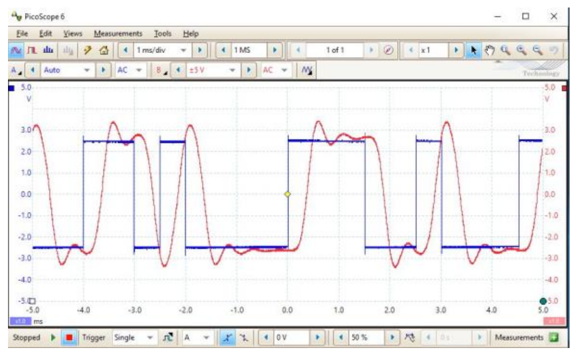
(Δf)max = 4.5 kHz on CH2



(a)



(b)



(c)

Fig 13 Typical scans to compare the digital signal just before and just after the filter section at (a) 1 kHz, (b) 1.5 kHz, (c) 2.0 kHz

Part B – Step Input

A common approach to characterize system delay and response is to employ a step function input. In this experiment, we will simulate the step function by viewing the initial response to a long duration pulse.

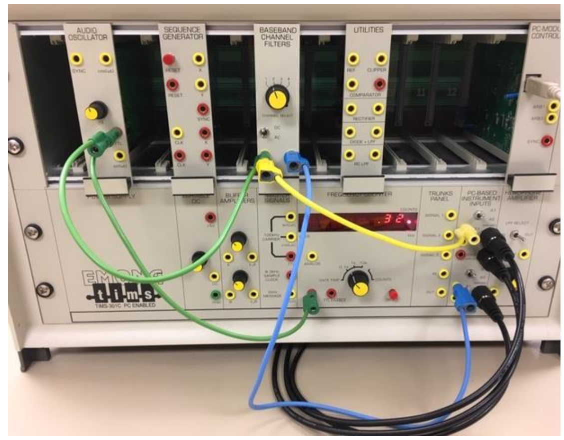


Fig 14 (a) configuration to simulate a step input.

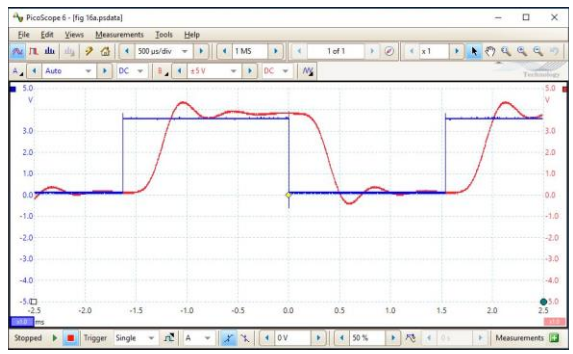
1. Connect the AUDIO OSCILLATOR output to the FREQUENCY COUNTER input and to the BASEBAND CHANNEL FILTERS input, as shown in Figure 14 (green leads).

- Connect Scope ChA to the input of the BASEBAND CHANNEL FILTERS input (yellow leads).
- Connect Scope ChB to the BASEBAND CHANNEL FILTERS output (blue lead).
- Adjust the Δf setting on the AUDIO OSCILLATOR to a low setting (fully counter clockwise is okay).
- Set the selection knob to Channel 2 for the BASEBAND CHANNEL FILTERS (BBLPF2).

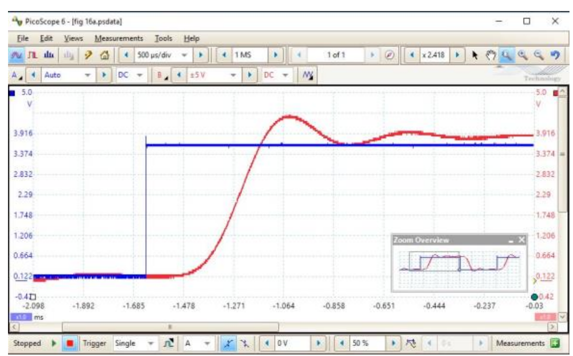
2. Now view the input and output signals on the PicoScope. With Δf set low, you should see only one or two pulses.

- Set the time scale to 500us/div, the output for Channel B to +/- 5V DC coupling, and the Trigger to Single. Press the green arrow to the left of the Trigger to view several scans, for instance like Figure 15a.
- Zoom in on a portion of the output signal as in Figure 15b. This will appear to be a step input signal plus system response to the step.

When the response to a step excitation is isolated in this way, so that there is no overlap with the responses of neighboring transitions, it is called the step response.



(a)



(b)

Fig 15 (a) output without zoom, (c) output with zoom appears as a step input and step response.

Notice the presence of oscillations and the relatively long settling time to the final value. This is known as ringing; a term that goes back to the days of manual telegraphy and Morse code.

3. Measure the system delay time. Refer to Figure 10.
- Find the time at the 10% point of the response, and subtract the time at the start of the step.
- Record the result in Table 2.

4. Measure the rise time. Refer to Figure 10.
- Record the result in Table 2.

5. Fill in the remainder of Table 2.

Table 2: Step Response for Various Filter Systems (low Δf setting)

	BBLPF2	BBLPF3	BBLPF4
Delay time	0.266 mS	0.26 mS	0.848 mS
Rise time	0.203 mS	0.291 mS	0.177 mS

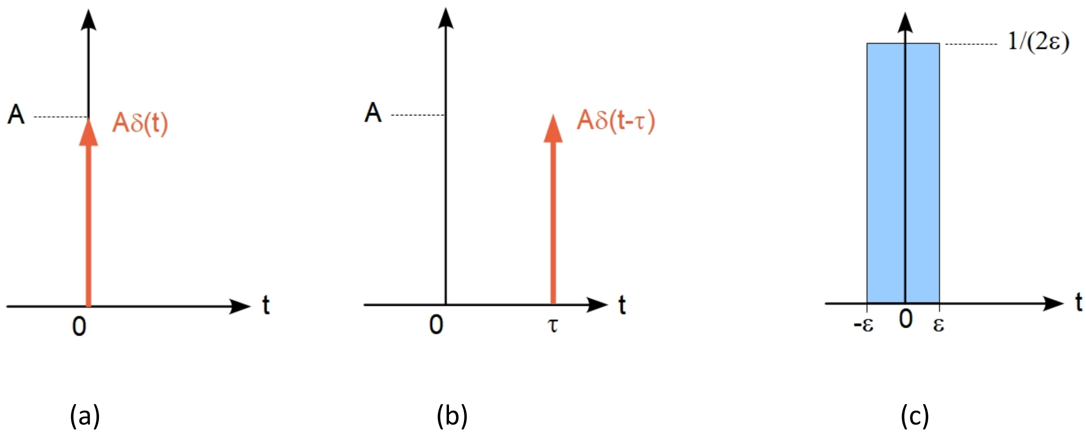


Fig. 16: (a) impulse function, (b) delayed impulse function, (c) model of impulse function.

Part C – Impulse

An ideal impulse function is shown in Figure 16a. As has likely been explained in lecture, the impulse function is a useful tool to express derivatives at discontinuities and for use in the sifting function. A system's response to an impulse function (i.e. the "impulse response") is very useful for linear time invariant systems. For LTI systems, any input signal can be expressed as a train of impulse functions. The system response to the signal is then simply the sum of the responses for the impulse train.

Mathematically, we can treat the impulse as a short duration pulse as shown in Figure 16c. For a unit impulse duration 2ε, the pulse has a height $1/(2\ \varepsilon)$ and thus unity area. For the ideal impulse, ε approaches zero. In our TIMS equipment experiment, we will approximate an impulse function with a short duration rectangular pulse, with the caveat that the height of the pulse will be system limited. Thus, our impulse signals will have an area less than 1, tantamount to feeding less power to the system. Consider a response in voltage vs time in Figure 17. The system's impulse response will have the same shape as that for an ideal impulse input, but will be compressed in voltage.

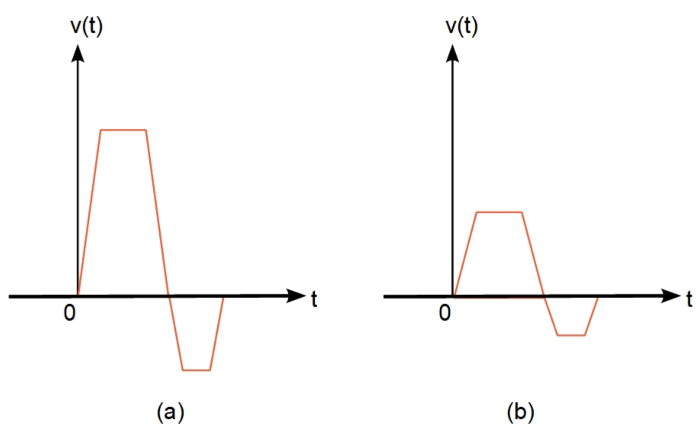


Fig 17 (a) system response to an ideal impulse input. (b) same system's response for a limited impulse input.

1. In order for the following instructions to match what you will be seeing, it is convenient to start fresh.

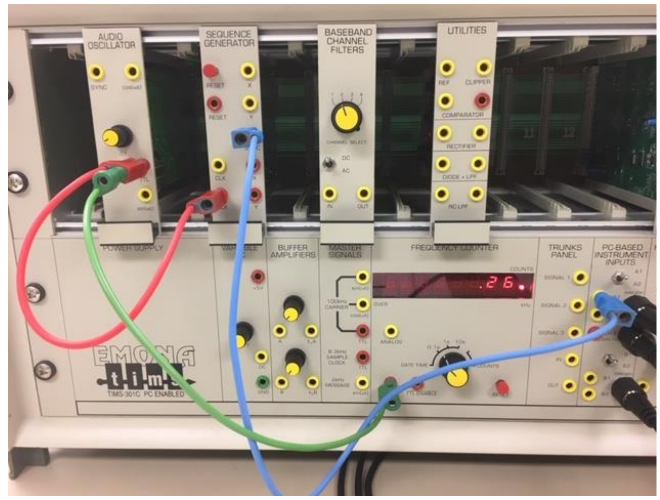
- Disconnect all leads and turn off the PicoScope.

2. The SEQUENCE GENERATOR's SYNC output will provide us with something similar to a single rectangular pulse signal.

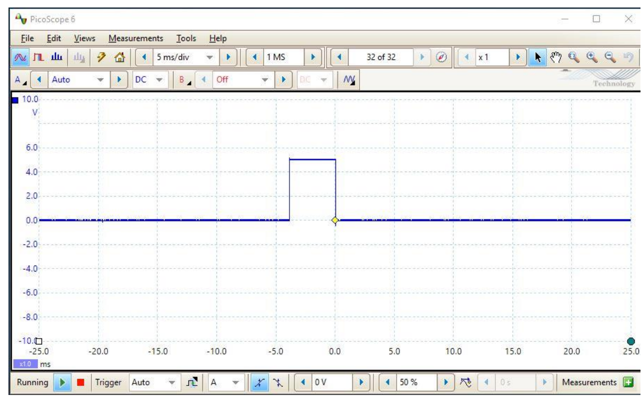
- Connect the AUDIO OSCILLATOR TTL LEVEL OUTPUT to the SEQUENCE GENERATOR's TTL CLOCK input, and to the FREQUENCY COUNTER's TTL input. See Figure 18a.
- Connect the SEQUENCE GENERATOR's SYNC output to Scope ChA.
- Turn Δf fully counterclockwise on the AUDIO OSCILLATOR.

3. Turn on the PicoScope.

- Set ChA to DC coupling
- Set the Trigger to AUTO. See Figure 18b.



(a)



(b)

Fig 18 (a) configuration to simulate an impulse. (b) the generated rectangular pulse.

4. Investigate the relationship between frequency and pulse width.

- Set Δf close to 0.5 kHz and measure the width of the pulse. Record in Table 3 and place the point on the blank graph of Figure 19.
- Repeat this process for the Δf values shown to complete Table 3 and Figure 18. You may wish to decrease the time scale to 1ms/div, and/or employ zoom in the vicinity of the pulse.

Table 3 – Pulse Width Data

Δf (cycles per second)	Pulse Width (us)
500	1952 uS
1000	1011 uS
1500	683 uS
2000	509 uS
3000	341.5 uS
4000	244.8 uS
6000	174.2 uS
8000	123.7 uS
10,000	98.45 uS

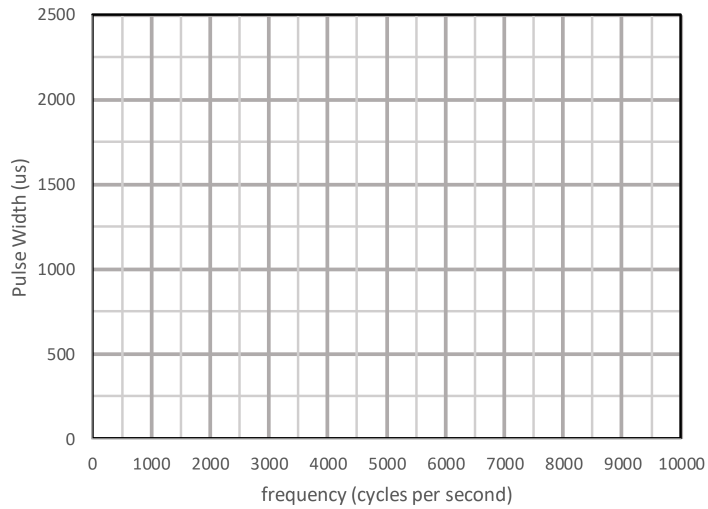


Fig 19 Relation between frequency and pulse width

5. Connect the SEQUENCE GENERATOR Sync output to the BASEBAND CHANNEL FILTERS input, and the BASEBAND CHANNEL FILTERS output to Scope ChB. See Figure 20.

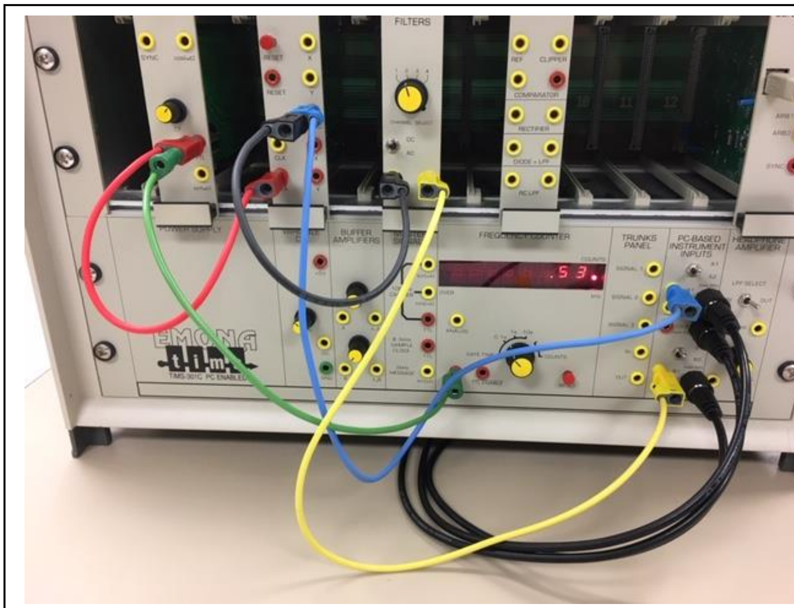


Fig 20 Configuration to view simulated impulse response.

6. On the PicoScope, turn on Ch B with DC coupling and +/- 10V.

- Set AUDIO OSCILLATOR Δf to 300 Hz.
- See Figure 21.

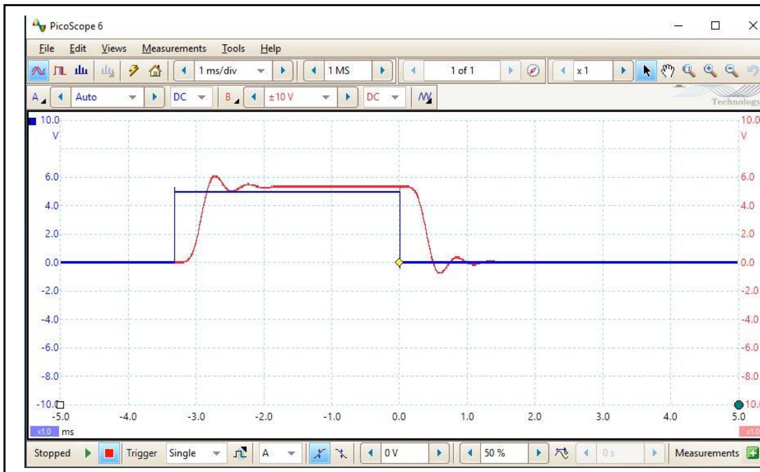


Fig 21 BBLPF2 response compared with input at 300 Hz.

7. Observe the response as you slowly increase Δf to 1000 Hz. See Figure 22.

- Note that the transitions are not affected, but as you continue to increase the frequency, and thus reduce the impulse width, the flat top between transitions gets shorter, and ultimately disappears.

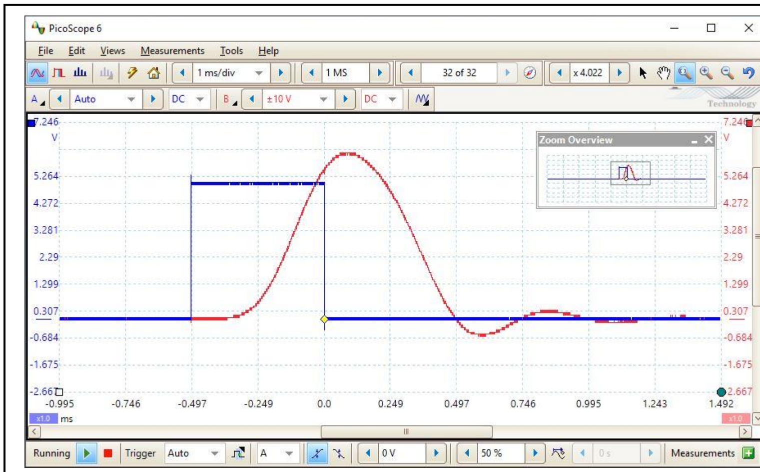
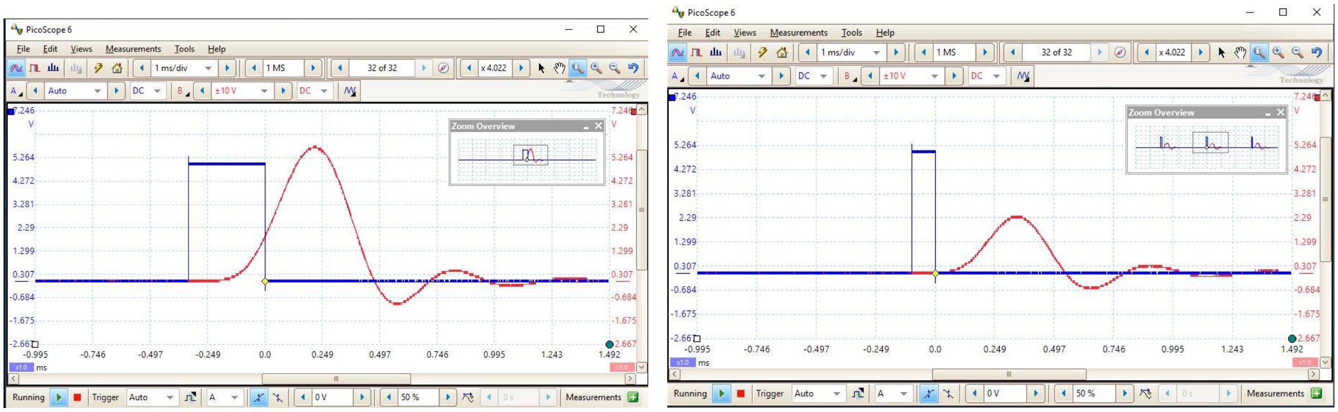


Fig 22 BBLPF2 response compared with input at 1000 Hz.



(a) (b)

Fig 23 Input and BBLPF2 response for (a) Δf = 2000; (b) Δf = 10,000

8. Increase Δf to 2000 Hz. See Figure 23(a).

9. Slowly increase Δf from 2000 to 10,000 Hz while observing the response. See Figure 23(b) for the response at 10,000 Hz.

Notice that when the pulse width becomes short enough, the general shape of the output (ignoring the amplitude) no longer changes. This is the impulse response for the system.

Q: Why does the amplitude change with continued decrease in pulse width?

The filtered signal does not have enough time to rise to its peak voltage before the falling edge of the un-filtered signal, so it falls back down.

Part D – Sinusoidal Signals

This section will focus on sinusoidal signals (i.e. sine waves). Why are sine waves important now in the digital age? One reason is that wireless communication systems transmit information by modulating sine waves. Also, the study of Fourier Series reveals that the spectrum of pulses is a collection of sine waves. So, the topic of sinusoidal signals remains of critical importance for electrical, computer, and wireless engineers.

D.1 – Basic Sine Waves

1. In order for the following instructions to match what you will be seeing, it is convenient to start fresh.
- Disconnect all leads and turn off the PicoScope.

2. Connect the AUDIO OSCILLATOR sin(ωt) output to the BUFFER AMPLIFIER input A. See Figure 24 (black lead)
- The BUFFER AMPLIFIER will allow control of the sinusoidal amplitude.

3. Connect the BUFFER AMPLIFIER output k1A to the FREQUENCY COUNTER, THE BASEBAND CHANNEL FILTERS input, and the Scope ChA. See Figure 24 (yellow leads)
- Adjust the AUDIO OSCILLATOR frequency to 300 Hz.
- Set BASEBAND CHANNEL FILTERS to Channel 2 (BBLPF2).

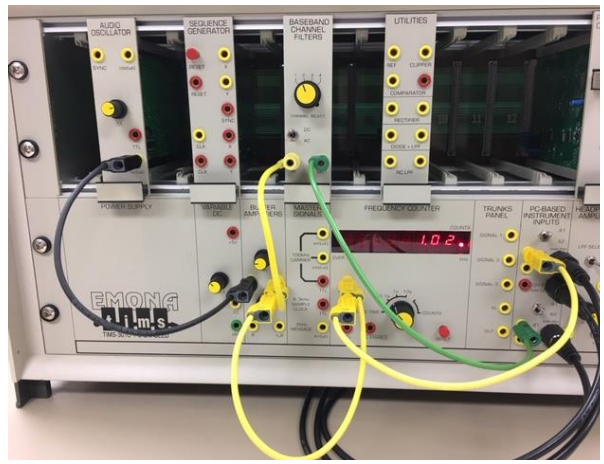


Fig 24 Configuration to view sinusoidal signals

4. Connect the BASEBAND CHANNEL FILTERS output to Scope ChB.
- See Figure 24, green lead.

5. Open the PicoScope
- Set timescale to 1ms/div and Trigger to Auto
- Turn on Channel B. Start with Auto Scale, but often you may need to adjust the scale.
- Adjust the BUFFER AMPLIFIER gain, knob k1, to achieve a 6 Vpp ("6 volts peak-to-peak") input signal.
- See Figure 25

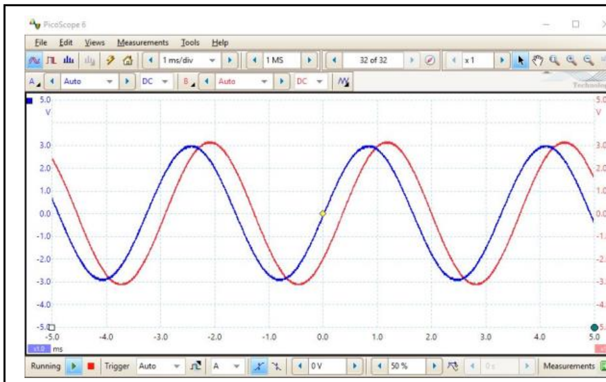


Fig 25 Input and output for the BBLPF2. The gain is set to provide a 6Vpp input sinusoid.

6. Read the output signal Vpp and record its value in Table 4.

7. Fill in the rest of Table 4 by adjusting the sinusoid's frequency and amplitude

- Maintain a constant 6 Vpp input signal.
- Adjust time scale as needed to view several cycles of the sinusoid
- Comment below on the signal quality. Note that Figure 26 shows the results at 4 kHz. Given the noisy nature of the signal, it will be hard to measure Vpp precisely. A good approach is to measure from the center of the fuzzy trace.

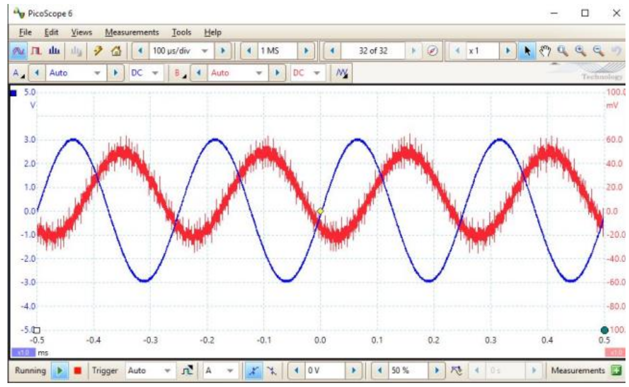


Fig 26 The output at 4 kHz is getting noisy.

Comments on Signal Quality:

The signal quality gets progressively worse as you increase the frequency.

Table 4

Frequency Hz	BBLPF2 Vpp	BBLPF3 Vpp	BBLPF4 Vpp
300	6.4 V	6 V	6.42 V
1000	6.4 V	4.822 V	6.97 V
2000	4.822 V	1.84 V	5.33 V
3000	526 mV	438.3 mV	32 mV
4000	100 mV	122 mV	175.3 mV
5000	88 mV	87.66 mV	175.3 mV

D.2 – Clipping

We will now use the UTILITIES module and show how the CLIPPER BIPOLAR OUTPUT (or CLIPPER for short) can convert a sine wave to a square wave.

1. In order for the following instructions to match what you will be seeing, it is convenient to start fresh.

- Disconnect all leads and turn off the PicoScope.

2. Connect the circuit as shown in Figure 27.

- Connect the AUDIO OSCILLATOR sin(ωt) output to the BUFFER AMPLIFIER input (black lead).
- Connect the BUFFER AMPLIFIER's ANALOG OUTPUT to the FREQUENCY COUNTER, the UTILITIES module's ANALOG SIGNAL INPUT, and Scope ChA (yellow leads).
- Connect the CLIPPER to Scope ChB (green lead).

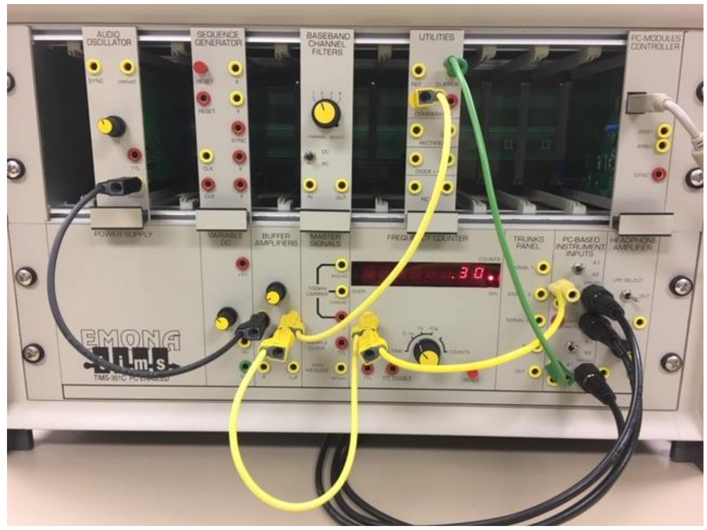


Fig 27 Configuration to study clipping.

3. PicoScope setup

- Timescale 1ms/div; set ChA trigger to Auto
- Adjust BUFFER AMPLIFIER k1 to achieve 6Vpp.

4. On the PicoScope, turn on ChB – set to the same scale as ChA. See Figure 28a.

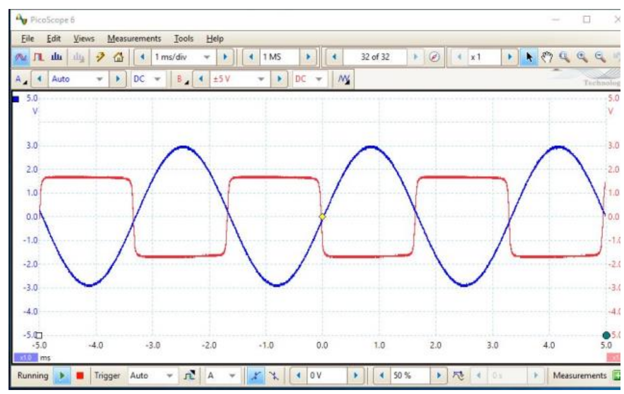
5. Adjust the BUFFER AMPLIFIER k1 to decrease the input signal to a Vpp of approximately 300 to 350 mV.

- On the PicoScope, use Single Shot Triggering.
- See Figure 28b.
- How has the output signal changed?

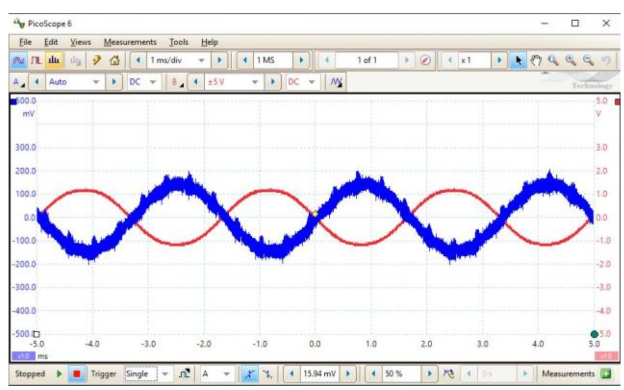
the output signal has less noise and is the inverse of the input signal.

6. Slowly increase the signal amplitude to 20 Vpp. Comment on the change to the output signal.

it becomes more of a square wave



(a)



(b)

Fig 28 (a) Clipped output for 6Vpp sinewave input. (b) output for weak input signal.

The signal to the Comparator must be strong enough to engage the Clipper.

E: – Digital Detector

A digital signal passing through a system (for instance, the BBLPF2) will suffer signal degradation. Then, provided the degradation is not too severe, a digital detector can recover the original signal. The ability to recover the signal despite degradation is a key advantage for digital systems.

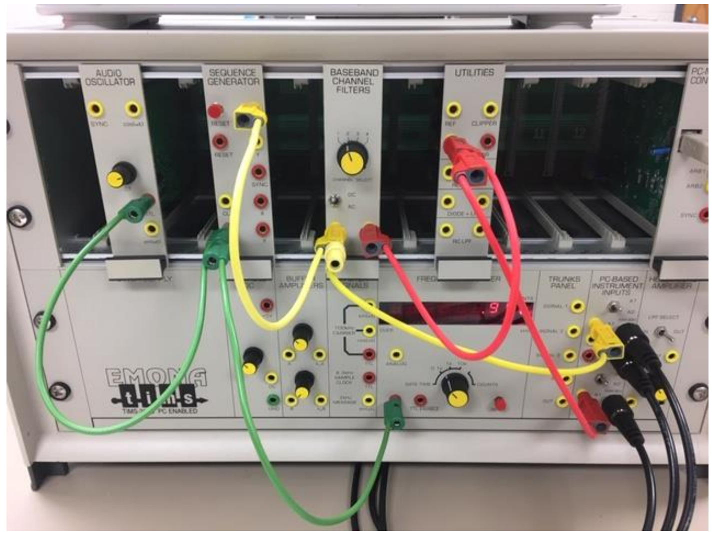


Fig 29 Initial digital detector configuration.

1. In order for the following instructions to match what you will be seeing, it is convenient to start fresh.

- Disconnect all leads and turn off the PicoScope.

2. Connect the circuit as shown in Figure 29

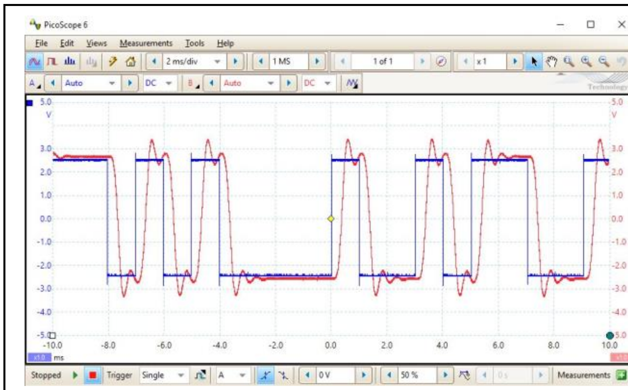
- Connect the AUDIO OSCILLATOR sin(ωt) output to the input of the SEQUENCE GENERATOR, and to the FREQUENCY COUNTER (green leads)
- Connect the SEQUENCE GENERATOR's AUDIO OUTPUT to the BASEBAND CHANNELS FILTER input, and to Scope ChA (yellow leads)
- Connect the BASEBAND CHANNELS FILTER output to the UTILITIES module ANALOG SIGNAL INPUT and to Scope ChB. (red leads)

3. Open the PicoScope and set it up as follows:

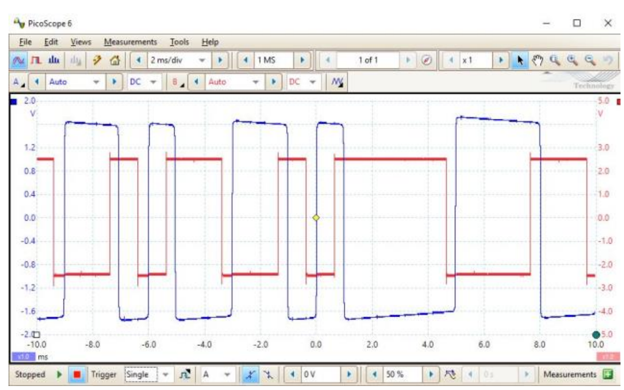
- Set timescale to 2ms/div
- Turn on ChB (to Auto)
- Select Single Shot Triggering
- See Figure 30a.

4. Disconnect Scope ChB from the BASEBAND CHANNELS FILTER output. Connect the UTILITIES module CLIPPER to Scope ChB.

- See Figure 30b. *Note the delay and the inversion of the signal.*
- Observe that the digital signal is accurately recovered, but inverted.



(a)



(b)

Fig 30 (a) The output of the filter is compared to the input digital signal. (b) the clipper output (i.e. the digital detector output) is compared with the input digital signal.

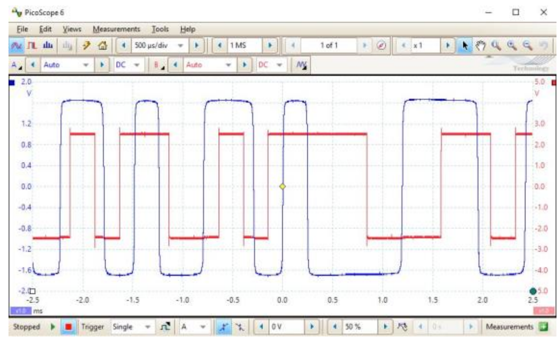
5. As the frequency is increased, the digital signal passing through the system (i.e. the BBLPF2) becomes more degraded. This will eventually lead to errors in the recovered digital signal.

- Observe the input digital signal and the recovered digital signal as frequency is increased. See Figure 31 for several examples.

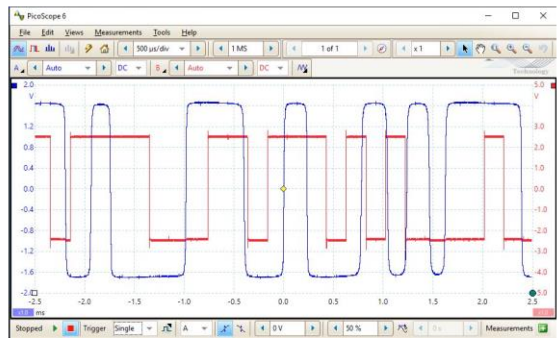
6. In Table 5, convert a convenient portion of your input and output signals to their digital equivalents

Table 5 Comparing recovered digital signal to original digital signal

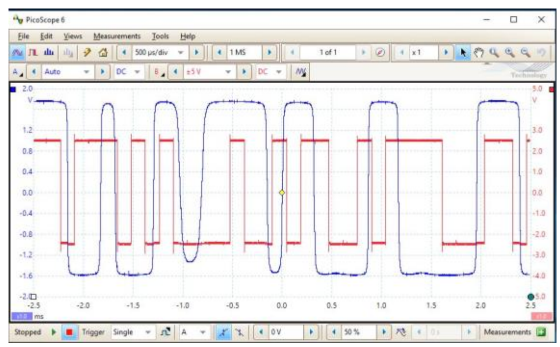
f (kHz)	Input digital signal	Output digital signal
4	1011001011100 1101	1011001011100 1101
5	01110011011010100010	01110011011010100001
7	011010100010010110 01011100110	011001000010011100 111001



(a) 4kHz



(b) 5kHz



(c) 7kHz

Fig 31 Recovery of the digital signal for increasing frequencies.

Lab was long but not overwhelming. I was able to learn a lot about how to operate the TIMS and the included software that goes along with it. This lab gives the student a good idea of what to expect out of the class and future labs.

END OF LAB 2