Lab Manual:
Signals & Systems

Using the EMONA Signals & Systems board for ADS MAX

Introduction & Sample Chapter

EMONA tims
WIRELESS SIGNALS LAB EQUIPMENT

Documented capabilities of the Emona Signals & Systems board

Introduction: Lab descriptions

Lab 1: Using the EMONA Signals & Systems board with ADS MAX

Lab 2: Introduction to the EMONA Signals & Systems board

Lab 3: Special signals – characteristics and applications

Lab 4: Systems – Linear and nonlinear

Lab 5: Unravelling convolution

Lab 6: Integration, Convolution and Correlation

Lab 7: Exploring complex numbers & exponentials

Lab 8: Build a Fourier Series analyser

Lab 9: Spectrum analysis

Lab 10: Time domain analysis of an RC circuit

Lab 11: Poles & zeros in the Laplace domain

Lab 12: Sampling and Aliasing

Lab 13: Getting started with analog-digital conversion

Lab 14: Discrete time structures

>>> SAMPLE CHAPTER Lab 15: Poles & zeros in z-plane-IIR

Lab 16: Discrete time filters – practical applications

Introduction

The EMONA Signals & Systems Lab Manual covers a broad range of introductory Signals & Systems topics through a series of hands-on laboratory experiments. Each experiment is written to support the theoretical concepts introduced in the class work of a first course in Signals & Systems.

In order to make the student's learning experience more memorable, the student is able to view a variety of signals on the ADS MAX oscilloscope.

Each lab experiment presents an interesting, hands-on learning experience for the student. In each experiment the student is challenged to build, measure and consider: there are no “instant” or “cookbook-style” experiments. The EMONA Signals & Systems Board (ESSB) is actually a true engineering modeling system where students see that the block diagrams so common in their textbooks represent real functioning systems.

Background: Teaching methodology behind the “block diagram approach”

The Emona Signals & Systems board draws on a well established experimental methodology that brings to life the “universal language” of Signals & Systems, the BLOCK DIAGRAM. Originally developed in the 1970’s by Tim Hooper, a senior lecturer in telecommunications at The University of New South Wales, Australia, and further developed by Emona Instruments, this modeling approach is used by thousands of students around the world, **to implement many different experiments across a variety of disciplines.**

Block Diagrams

Block diagrams are used to explain the principle of operation of electronic systems (like a radio transmitter for example) without worrying about how the circuit works. Each block represents a part of the circuit that performs a separate task and is

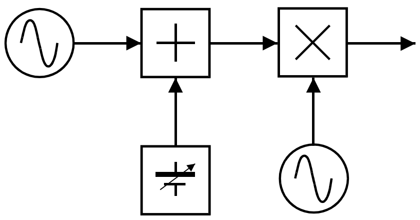

Figure 1: Example block diagram involving addition and multiplication of signals

named according to what it does. Examples of common blocks in communications equipment include the *adder*, *multiplier, oscillator*, and so on.

The board is a collection of blocks (called modules) that are patched together to implement dozens of Signals & Systems experiments. Experiments make use of the board together with the ADS MAX providing the instrumentation.

This approach to implementing Signals & Systems experiments through realizing BLOCK DIAGRAMS has the following benefits in the educational environment:

- Students gain practical experience with true mathematical modeling hardware, designed specifically for implementing Signals & Systems theory.
- Students actually build each experiment stage-by-stage, in an engineering manner, by following the BLOCK DIAGRAM.
- Students are free to try “what-if” scenarios to validate their understanding of the theory being investigated, by viewing real, real-time electrical signals which represent the mathematics being studied.
- The board is designed to allow students to make mistakes, hence students will learn from their hands-on experiences as they investigate their findings.

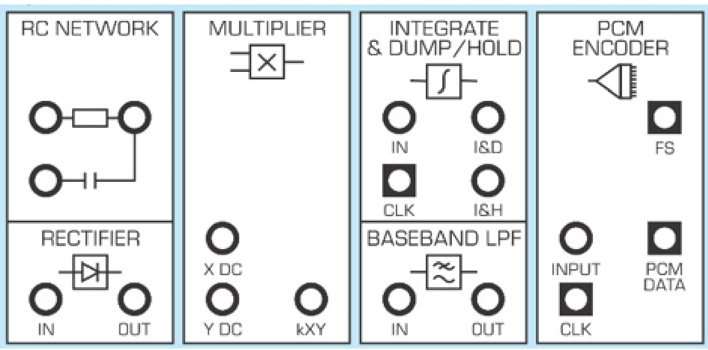

Figure 2: Example individual circuit blocks

One-to-One Relationship

The figure above shows a few of the blocks contained on the board and that they are independent of each other and closely related to their BLOCK DIAGRAM.

The functional blocks of the board are used and re-used in experiments, just as blocks of the block diagram reappear in many different implementations, and just as LabVIEW blocks are interconnected to form program flows.

Guidelines for Using the Lab Manual

This manual covers a broad range of concepts, from fundamental topics through to the advanced topics which form an introduction to Digital Signal Processing. In each experiment, the core technology is revealed to the student, at its most fundamental level. The first chapters also provide a solid introduction to the ADS MAX platform.

Chapters can be covered in any order, however, it is imperative that all students complete the first two chapters before proceeding to the subsequent chapters.

- Lab 1 Introduction to the EMONA Signals & Systems board with ADS MAX
- Lab 2 Introduction to the EMONA Signals & Systems board

Learning Objectives

Lab 1 - Using the EMONA Signals & Systems board with ADS MAX

In this Lab, students are familiarised with the use of the EMONA Signals & Systems board in the ADS MAX unit.

Lab 2 - Introduction to the EMONA Signals & Systems board

In this Lab, students are guided through an explanation of each circuit block on the Signals & Systems board in order to familiarise them with the capabilities of each and every block on the board. In this way they are able to quickly and easily relate electrical blocks with the block diagram elements in the theory. They are also presented with the specifications of each circuit block.

Lab 3 - Special signals - characteristics and applications

In this Lab students investigate how signals are distorted when a system's response is affected by inertia, and discover signals that are useful for probing a system's behaviour. Signals used are the impulse, step and sinusoid and as this experiment is a process of discovery, we will name the blocks which represent the channel " System Under Investigation" until we have familiarized ourselves with their actual characteristics through the use of these "probing" signals.

Lab 4 – Systems – Linear & Nonlinear

In this Lab students will focus on the "system" and discover a signal set which is perfectly matched to a special class of systems. Explorations of various linear and nonlinear systems as well as memory effect will take place. Finally, an introductory investigation of frequency response.

Lab 5 – Unravelling convolution

This lab offers a more evocative experience of convolution. By tracing the passage of some basic signals through a simple linear system, the student will be able to observe the underlying process in action, and, with a little arithmetic, discover a formula as it emerges from the hardware. Superposition is introduced and utilized.

Lab 6 - Integration, Convolution & Correlation

In this Lab students will bring to life the essential mathematical principles of Integration, Convolution and Correlation using real world signals interacting with real world systems. Exploring system responses to real signals by manually executing point by point correlations and convolutions will reinforce the meaning and the relationships between these often-used processes.

Lab 7 – Exploring complex number and exponentials

In this Lab, students will implement realizations of signals which can be described using complex numbers and by so doing, develop an in depth understanding of one use of complex numbers. As well, the creation of exponentially decaying signals will explore the meaning of exponential decay in processes through the use of real electrical signals.

Lab 8 – Build a Fourier Series analyser

In this Lab, students will explore the Fourier series by manually decomposing a complex sinusoid into its constituent sine and cosine components using real electrical signals. In this way they will discover and prove to themselves how the fourier series is defined and why it works. As well, the principle of Inverse Fourier Transform will be introduced.

Lab 9 – Spectrum Analysis

In this lab, students will compare time and frequency domain representation of numerous important signals, including impulses, sinc pulses and pseudo random noise sequences. As well, non-linear processes are explored and their frequency domain qualities recognised. Real signals are investigated throughout.

Lab 10- Time domain analysis of an RC circuit

In this lab, students will analyse a simple RC circuit as a network, using tools such as steps, impulses, exponential pulses and sinusoids to compare responses with theory. As well they will synthesize an equivalent network and test its performance with real signals.

Lab 11 - Poles and zeros in the Laplace domain

In this lab, students will implement a continuous time structures with integrators and study its CT responses. As well they will learn how poles and zeros can be used to visualize frequency responses graphically, and how to apply this to higher order systems.

Lab 12 - Sampling and aliasing

In this lab, students will sample a continuous signal and investigate the spectrum of this signal, relating this back to the sampling theorems and aliasing issues. As well, undersampling is introduced and its use in other applications.

Lab 13 - Introduction to analog-digital conversion

In this lab, students study real digital streams to understand quantization, synchronization and encoding issues relating to PCM data streams. A conversion process from analog to digital and back to analog is implemented.

Lab 14 - Discrete-time filters with FIR systems

In this lab, students implement a FIR structure and use this to discover the relationship of a discrete-time FIR filter to its transfer function.

Lab 15 - Poles and zeros in the z plane with IIR systems

In this lab, students will evaluate a real circuit implementation and investigate how to intuitively design recursive/IIR discrete-time responses. They will use a Bode Analyzer and design software to automate selection of poles & zeros in IIR structures.

Lab 16 - Discrete Time filters - practical applications

In this lab, students will construct a 2nd order IIR structure and investigate the importance of sensitivity and dynamic range issues that arise in the practical implementation of discrete-time filters, as well as studying the ussies relating to optimum sampling rate selection.

Lab Manual:
Signals & Systems

Using the EMONA Signals & Systems Explorer board for ADS MAX

Lab 15: Poles & zeros in z-plane-IIR

List of Updates

Date	Details
9/7/2019	Completed final document
5/6/2025	Updated to ADS MAX implementation

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Lab 15: Poles & zeros in z-plane-IIR

Learning Objectives

After completing this lab, you should be able to complete the following activities.

1. Discuss the poles and zeros of the transfer function of discrete-time filters to visualize frequency responses graphically at a glance, without math.
2. Describe how to intuitively design recursive/IIR discrete-time responses
3. Use design software to automate the selection of poles, zeros and implementation gains for IIR structures.
4. Use Bode Analyzer instrument to view system responses.

Prerequisites

You should have completed Lab 1 and Lab 2 and be familiar with the equipment, its use and the handling precautions for the equipment.

Required Tools and Technology

Platform: ADS MAX

Instruments used in this lab:

- Oscilloscope-Time
- Oscilloscope-FFT
- Function Generator

Hardware: Emona Signals & Systems Board

Components used in this lab:

- Six BNC to 2mm banana-plug leads
- Assorted 2mm banana-plug patch leads

Software: LabVIEW 2025 Run Time Engine

Emona Signals & Systems Board SFP

EMONA Signals & Systems ADSMAX*.zip

Expected Deliverables

In this lab, you will collect the following deliverables:

- ✓ Calculations
- ✓ Data from measurements
- ✓ Observations

Your instructor may expect you complete a lab report. Refer to your instructor for specific requirements or templates.

Section 1: IIR structures

1.1 Theory & background

In Lab 11 we discovered how poles and zeros can be used as an intuitive tool for analyzing and designing continuous-time (CT) filters. Next, in Lab 14 we examined discrete-time (DT) FIR filters and found the same ideas could be applied there. The complex "s" plane was replaced with the complex "z" plane, and the unit circle used instead of the j axis for the representation of frequency. Because zeros only are involved in FIR filter work, this provided a convenient gateway to getting started with z-plane ideas.

In this Lab we will investigate more general DT filters that are characterized with both poles and zeros. These filters are known as *recursive* since they use feedback, and also as *Infinite Impulse Response (IIR)*. With feedback we will be able to realize much higher selectivity than possible with a comparable complexity FIR implementation. The most conspicuous example is the second-order resonator, which will open the way to achieving realistic bandpass responses. As we proceed, we will find many parallels with the CT (continuous time) filter experiments in Lab 11.

Section 1: we examine the behaviour of the basic second-order resonator implemented without zeros.

Section 2: zeros are introduced to generate lowpass, bandpass, highpass and allpass responses using the Direct Form 2 structure.

1.2 Preliminary work:

This preparation extends the theory covered in Lab 14 to include poles.

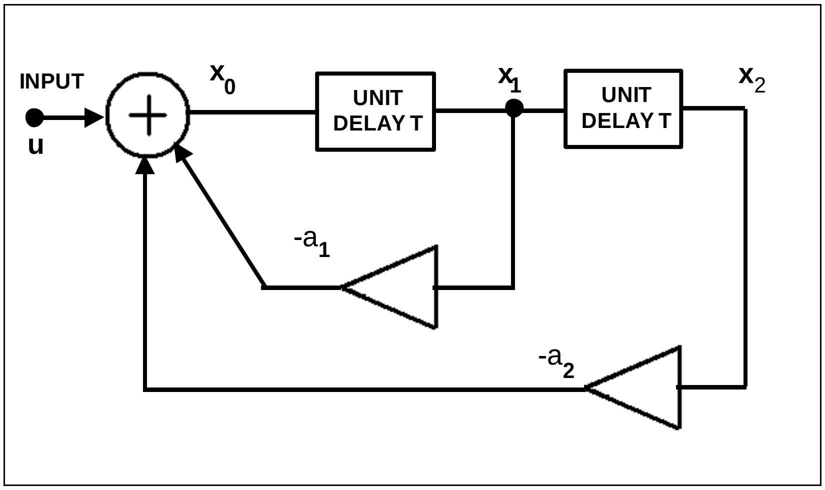


Figure 1: schematic of 2nd-order feedback structure without feedforward.

1-1 Consider the feedback system in Figure 1.

Show that the difference equation relating the adder output x0(nT) and the input u(nT) is

$$x_0(nT) = u[nT] - a_1.x_0[(n - 1)T] - a_2.x_0[(n - 2)T] \qquad \text{(Eqn 1)},$$

where nT are the discrete time points, T sec denoting the unit delay, i.e. the time between samples.

Show by substitution that e^{jnTw} is a solution, i.e. show that when $x_0(nT)$ is of the form e^{jnTw}, the input u(nT) is e^{jnTw}, multiplied by a constant (complex-valued); w is the frequency of the input in radians/sec; (the use of complex exponentials for the representation of sinusoidal signals is discussed in Lab 8, 10 and 13.

From the above, with input u(nT) = e^{jnTw} obtain

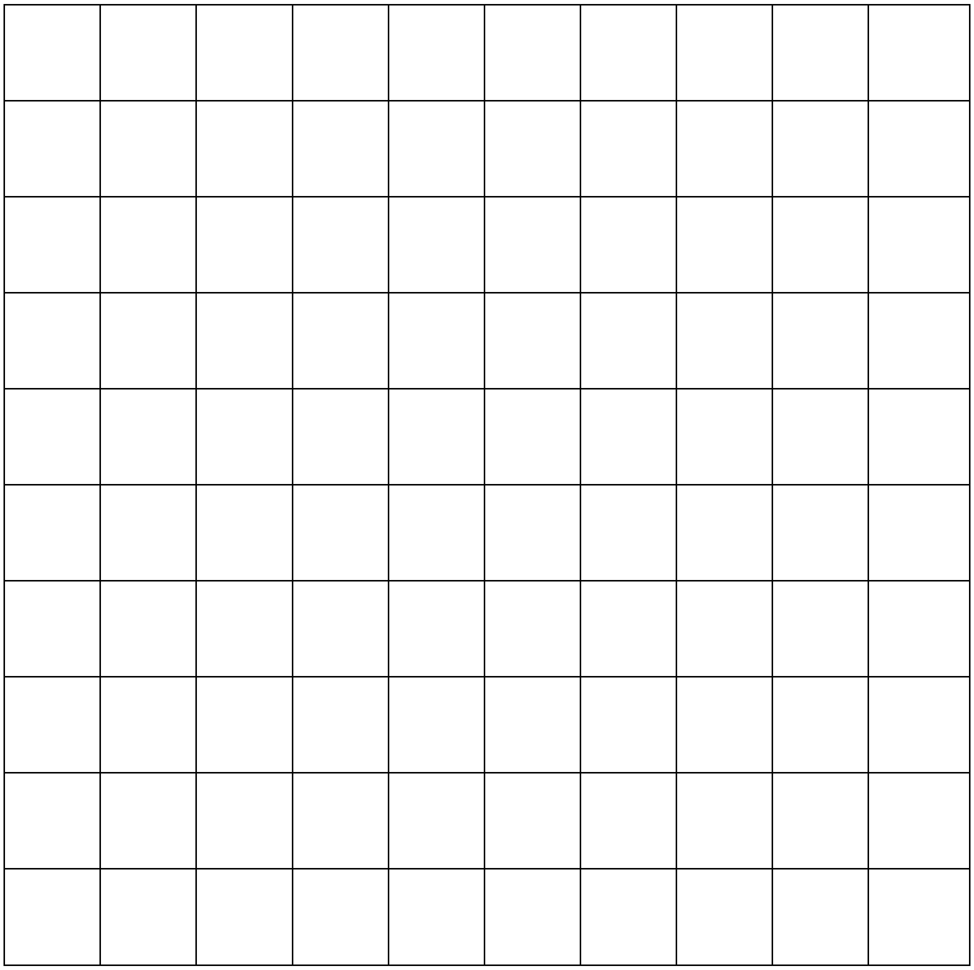
$$x_0/u = 1/[\, 1 + a_1.\, e^{-jTw} + a_2.\, e^{-j2Tw}] \qquad \text{(Eqn 2).}$$

Note that x_0/u is not a function of the time index n.

1-2 Use this result to obtain a general expression for $|x_0/u|$ as a function of w.

Tip: to simplify the math, operate on u/x_0 instead of x_0/u, expressing the result in polar notation.

Set T = 1 sec for the time being, and plot the result for the case $a_1 = -1.6$, $a_2 = 0.902$ over the range w = 0 to π rad/sec. Label the frequency axis in Hz and in rad/sec. You should find there is a peak in the response near 0.09Hz.



Graph 1:response plot

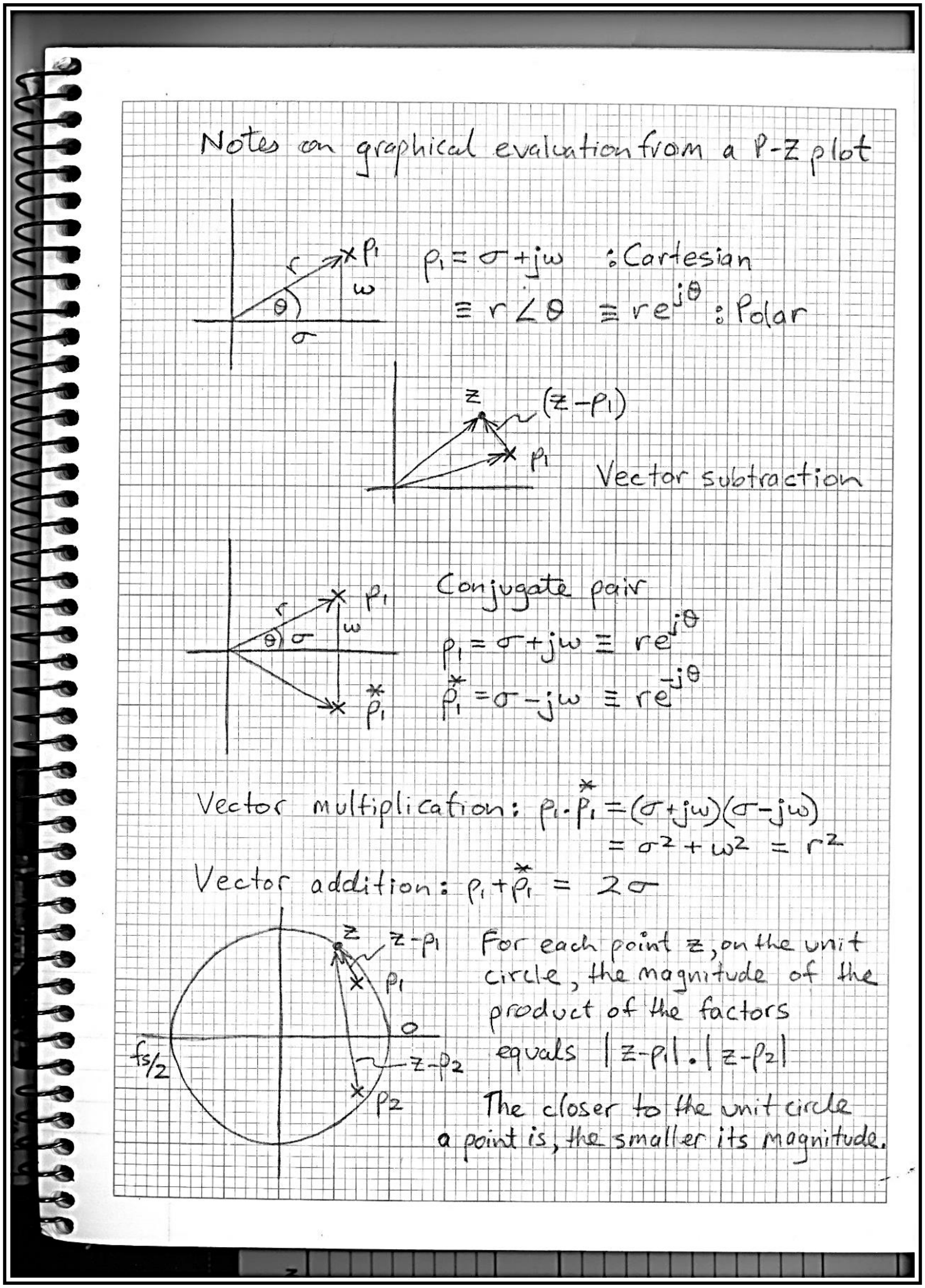


Figure 2: Notes on the graphical interpretation of pole-zero plots

Reviewing the finding of roots of the quadratic polynomial

From equation 1 above
$x_0(nT) = u[nT] - a_1.x_1(nT) - a_2.x_2(nT)$ substituting $x_1 = x_0.z^{-1}$ and $x_2 = x_0.z^{-1}.z^{-1}$ we arrive at
$x_0 = u - a_1x_0/z^1 - a_2x_0/z^2$
Grouping x0 terms:
$x_0(1 + a_1/z^1 + a_2/z^2) = u$

At this point we can see that although we started with negative gains in the circuit model, we now have positive values as coefficients in the quadratic equation.

Further, we arrive at:
$x_0/u = z^2/(z^2 + a_1z^1 + a_2)$ which we earlier named Eqn 3.
We now have the general quadratic form with positive coefficients.
<u>INSIGHT: positive coefficients result in negative gains in the actual implementation</u>

This quadratic $(z^2 + a_1z + a_2)$ can be expressed in factored form, as (z-p1)(z-p2)
Remember that z, p1 & p2 are complex numbers. You can think of these as vectors: from the origin of the z plane to a 2 dimensional point on that plane.

Each factor ie: (z-p1) and (z-p2) is a difference vector between a general point z, who's locus we restrain to the unit circle, and the 2 specific roots p1 & p2. It will be a vector, having direction and magnitude, and can be expressed in polar notation as $r\angle\theta$, or $re^{j\theta}$, or in Cartesian notation as (a + ib). Both these representation are complex numbers.

If we define p1 as $(\sigma + iw)$ and its conjugate, p2 as $(\sigma - iw)$ we can express the quadratic factors as:
$(z- p1)(z- *p1) = z^2 + p.*p - pz - *pz$
Switching to polar notation for convenience, $p.*p = re^{j\theta}.re^{-j\theta} = r^2$
So that leaves $z^2 + r^2 - z.(p + *p)$, and if using Cartesian notation in this instance for convenience, ie. $p = \sigma + jw$ then $p + *p = 2\sigma$ so
$z^2 + (-2\sigma)z + r^2 = z^2 - 2\sigma z + r^2 = z^2 + a_1z + a_2$

Relating coefficients gives $\mathbf{a_1 = -2\sigma}$ **and** $\mathbf{a_2 = r^2}$

For stability the poles must always be inside the unit circle, hence $0 < a_2 < 1$

Changes in a_1 directly influence the real component of the pole position
a_2 has a square law relationship with r of the pole.
Other relationships, such as θ, w, imag part, can be derived from these easily with trigonometry.

The general solution for the roots of the quadratic polynomial $x_2 + a_1x + a_2$ is:
$\mathbf{x = -a_1/_2 +/- i\sqrt{[a_2-(a_1/_2)^2]}}$

With these equations in mind consider how changes in the coefficients from the math will move the poles or zeros about the unit circle, and influence the response of the system.

This lab aims to make you more familiar of the interrelationships between these parameters.

1-3 Replace "exp(jTw)" by the symbol "z" in Eqn 2. The result is

$H\_x_0(z) = x_0/u = 1/(1 + a_1.z^{-1} + a_2.z^{-2}) = z^2 / (z^2 + a_1.z + a_2)$
(Eqn 3).

The quadratic ($z^2 + a_1.z + a_2$) can be expressed in the factored form $(z - p_1)(z - p_2)$.

Using the values of a_1 and a_2 given in Question 2 above, find the roots p_1 and p_2 (express the result in polar notation). Mark the position of p_1 and p_2 on the complex z plane with an "x" to indicate that they represent poles. The distance between these points and the unit circle is of key importance.

This is a parallel process to that in Lab 11 where we plotted zeros. A similar procedure was carried out in Lab 11 for a CT transfer function in the complex variable s.

Write down a formula for p_1 in terms of a_1 and a_2. Note that p_1 may be real or complex depending on a_1 and a_2. Determine the conditions for p_1 to be complex valued. For this case, express p_1 in polar notation. Take note of the fact that $|p_1|$ does not depend on a_1 (this will be useful later). Obtain p_2 from p_1.

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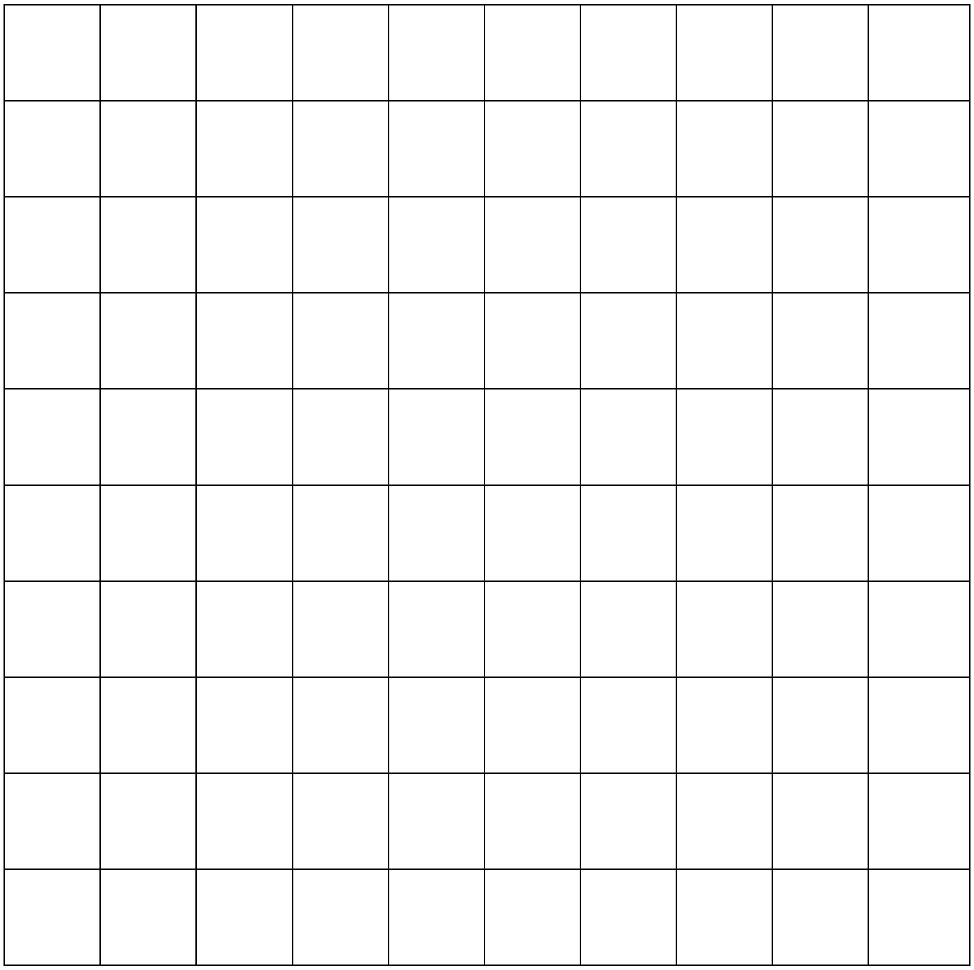
1-4 Satisfy yourself that the magnitude response of H_x_0 is given by

$|H\_x_0(w)| = 1/[|(e^{jTw} - p_1)|.|(exp^{jTw} - p_2)|]$ (Eqn 4).

This provides the key for the graphical method described in Lab 13 to obtain an estimate of the magnitude response. Again, we will use T = 1 .

Plot the magnitude of the denominator for selected values of w over the range 0 to π. The quantity $|(e^{jTw} - p_1)|$ becomes quite small and changes rapidly as the point on the unit circle is moved near p_1. Plot additional points there as needed. Invert to get $|H\_x_0(w)|$ and compare this with the result you obtained in (b).



Graph 2:response plot

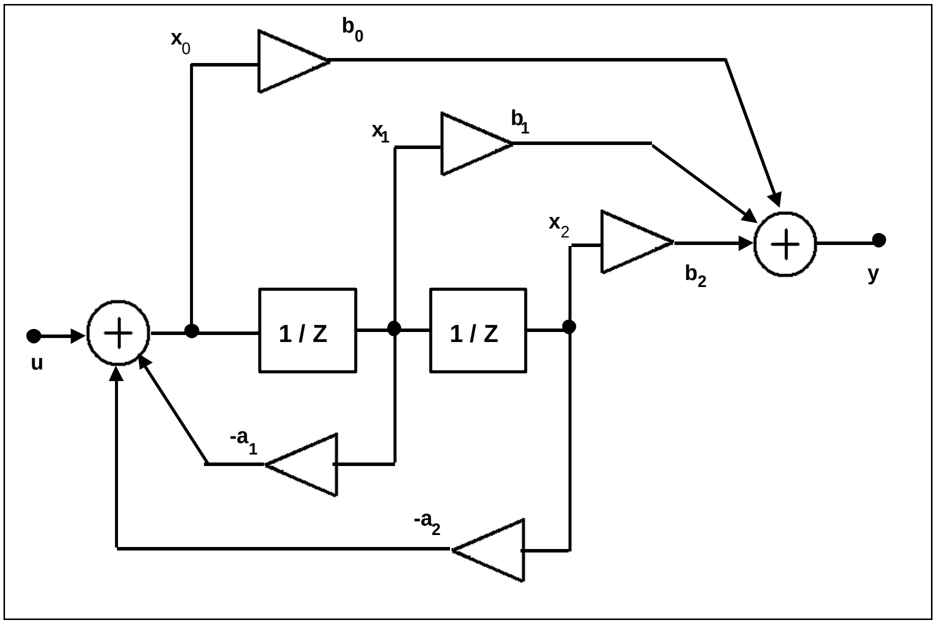


Figure 3: block diagram of 2nd-order Direct-form 2 structure with feedforward.

1-5 Modify Fig 1 by replacing the unit delays with a gain of 1/z and show that Eqn 3 follows by inspection using simple algebra.

1-6 Apply this idea to show that the transfer function for the system in Fig. 3 is

$$H\_y(z) = y/u = (b_0 + b_1.z^{-1} + b_2.z^{-2}) / (1 + a_1.z^{-1} + a_2.z^{-2}) \qquad \text{(Eqn5)}$$

1-7 Use the graphical pole-zero method (covered in Experiment 14) to obtain estimates of the magnitude responses for the following cases (0 to Nyquist freq):

(i) $b_0 = b_2 = 1$, $b_1 = 2$, a_1 and a_2 as in Question 2.
(ii) $b_0 = b_2 = 1$, $b_1 = -2$, a_1 and a_2 as in Question 2
(iii) $b_0 = 1$, $b_1 = 0$, $b_2 = -1$, a_1 and a_2 as in Question 2

Which of these is lowpass, highpass, bandpass?

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1-8 Consider a DT system with sampling rate 20kHz. Obtain estimates of the poles and zeros that realize a lowpass filter with cut-off near 3kHz. Obtain a highpass filter using the same poles.

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1-9 For the same sampling rate as in Question 8 obtain estimates of the poles and zeros that realize a bandpass filter centered near 3.1kHz, with 3dB bandwidth 500Hz. HINT: review Question 7

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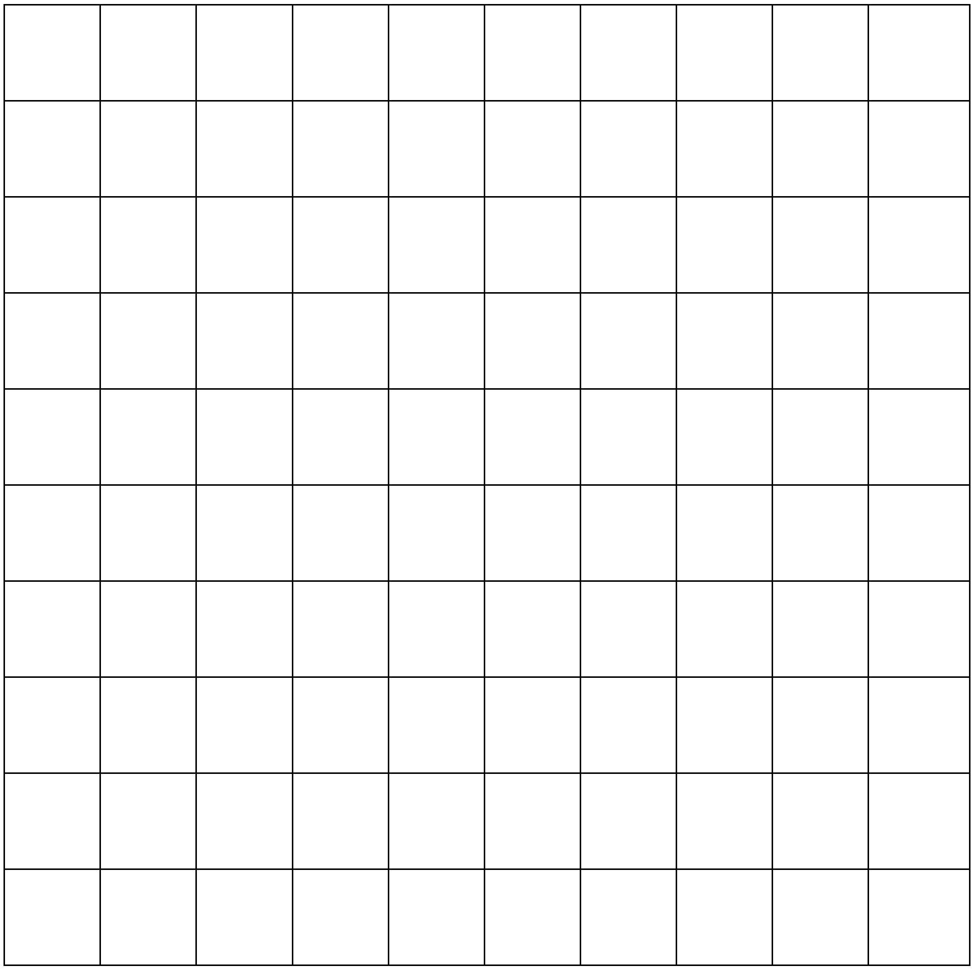


Graph 3:pole-zero method

Powering up the ADS MAX + EMONA Signals & Systems Board

1. Ensure that the ADS MAX Application Board power button at the top left corner of the unit is OFF (not illuminated).

2. Carefully plug the Emona Signals & Systems board (ESSB) into the ADS MAX ensuring that it is fully engaged both front and back.

3. Ensure that you have connected the ADS MAX to the PC using the USB cable and that the PC is turned on.

4. Turn on the Application Board *Power* button by pressing it once and confirm that it is illuminated. The LEDs on the ESSB should also be illuminated. If they are not, then switch the unit off immediately and check for connection or insertion errors.

5. Launch the Signals & Systems Explorer Soft Front Panel (SFP).

6. You're now ready to work with the ADS MAX/Signals & Systems bundle.

7. No need to select any particular tab at the moment.

Note: To stop the Signals & Systems SFP when you've finished the experiment, it's preferable to use the STOP button on the Signals & Systems SFP itself rather than the LabVIEW window STOP button at the top of the window. This will allow the program to conduct an orderly shutdown and close the various channels it has opened.

Section2: IIR without feedforward

2.1: Implement: Second-order resonator

In this part we implement and investigate the system in Fig 1.

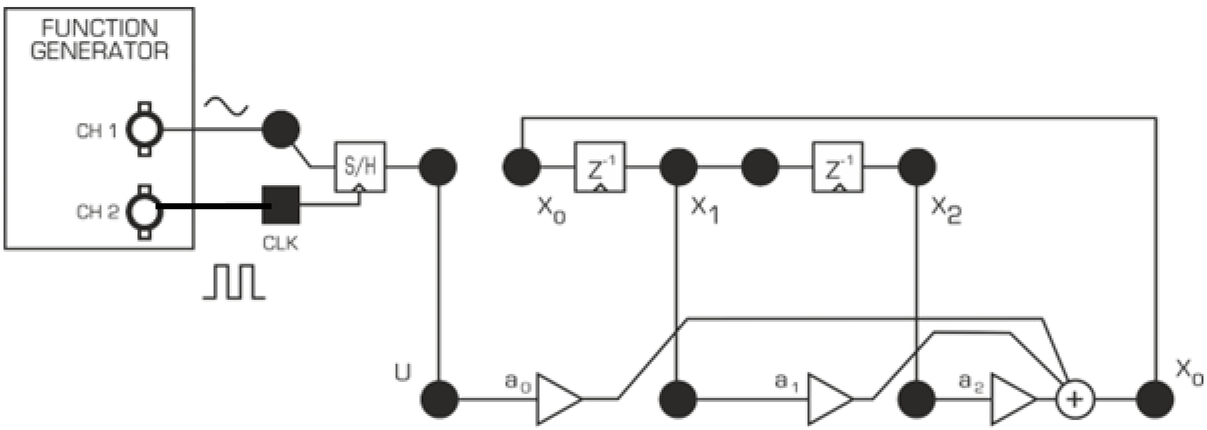


Figure 4: patching diagram for feedback structure without feedforward from Figure 1

3. Patch up a Signals & Systems board model of the system in Fig 1.

Settings are as follows:

Scope Configuration

Horizontal Timebase	500 us/div
Trigger	Source: sine signal; 0V level

Function Generator Configuration

Waveform	Sine
Frequency FG1	1 kHz
Amplitude	1Vpp
DC Offset	0V

Function Generator Configuration

Frequency FG2	**20 kHz**
Duty Cycle	0.5 (50%)

ADDER GAINS: a0=1; a1=+1.6; a2= -0.902

2-1 Calculate the poles corresponding to these values. Measure and plot the magnitude response at the output of the feedback adder. Note and record the resonance frequency and the bandwidth. Use the poles to graphically predict these parameters; compare with your measurements.

HINT: Do a quick sweep of frequency range then sweep the range more slowly while taking measurements. This will enable you to see the variation in gain of the output.

2-2 Decrease $|a_1|$ by a small amount (around 5-10%, say) and measure the effect on the resonance frequency and bandwidth. Use this to estimate the migration of the poles. Does this agree with your expectations?

2-3 Repeat step 3 for a 5% decrease of a_2. Compare the effects of varying a_1 and a_2. Which of these controls would you use to tune the resonance frequency? Use the formulas you obtained in the preparation to explain this.

NOTE about TAB " PZ plot" on the Signals & Systems board SFP.

This panel calculates the poles and zeros relating to the currently set ADDER gains which relate to the coefficients of the transfer function. Use this visualization tool to confirm your understanding . You may wish to move back and forth between the experiment TAB and the "PZ plot" TAB as required during the experiment.

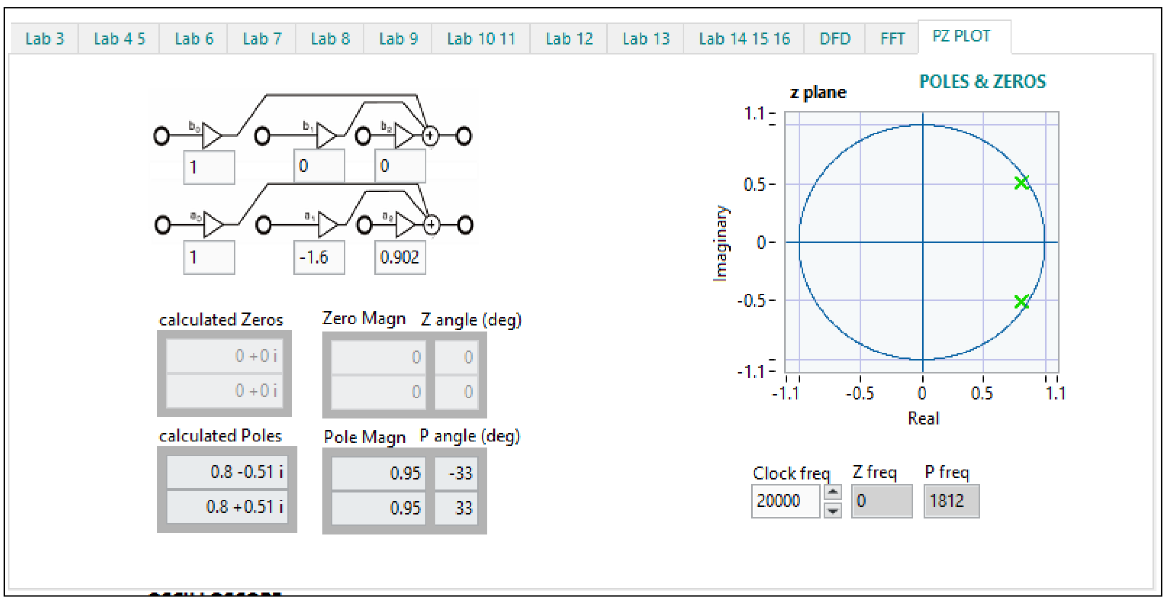


Figure 5: "PZ plot" TAB from Signals & Systems board SFP

2-4 With a1 unchanged, gradually increase a_2 and observe the narrowing of the resonance. Continue until you see indications of unstable behaviour. At that point, remove the input signal and observe the output (if needed, increase a_2 a little more). Is it sinusoidal? Measure and record its frequency. Measure a_2. Calculate and plot the pole positions. Note especially whether they are inside or outside the unit circle.

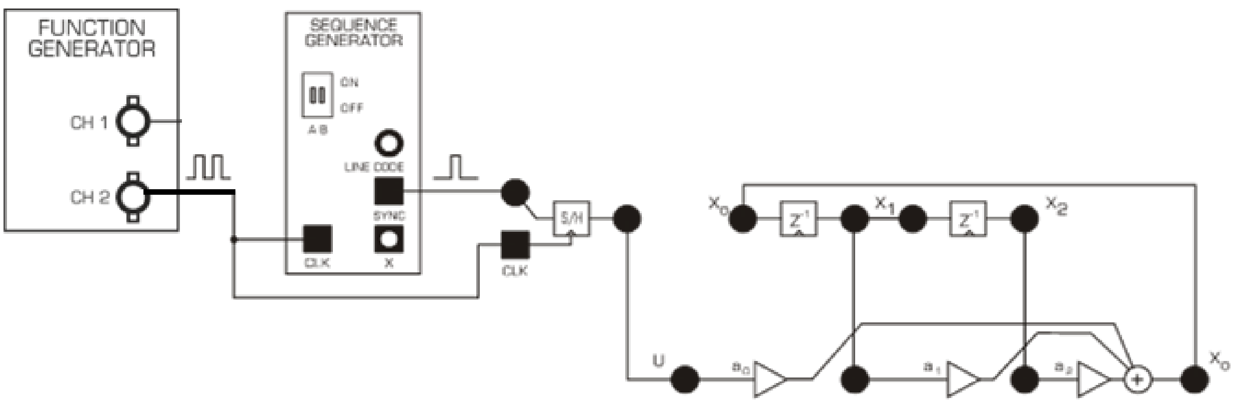


Figure 6: patching diagram for step response

4. We will now repeat the step in the time domain. Use FUNCTION GENERATOR FG2 as a clock source and SEQUENCE GENERATOR to set up a unit pulse input. The SYNC signal from the SEQUENCE GENERATOR will act as a repetitive unit pulse source. What matters is that the unit pulses are far enough apart that each pulse is a unique event to the system under investigation.

Setting are as follows:
SEQUENCE GENERATOR: DIPS set to UP:UP (short sequence)
ADDER gains: a0=1.0; a1=1.6; a2=-0.902; b0=1.0

2-5 Begin with a_2 around -0.9. Describe the effect on the response as the magnitude of a_2 reduces. Measure the frequency of the oscillatory tail of the response and compare with your observations in step 5.

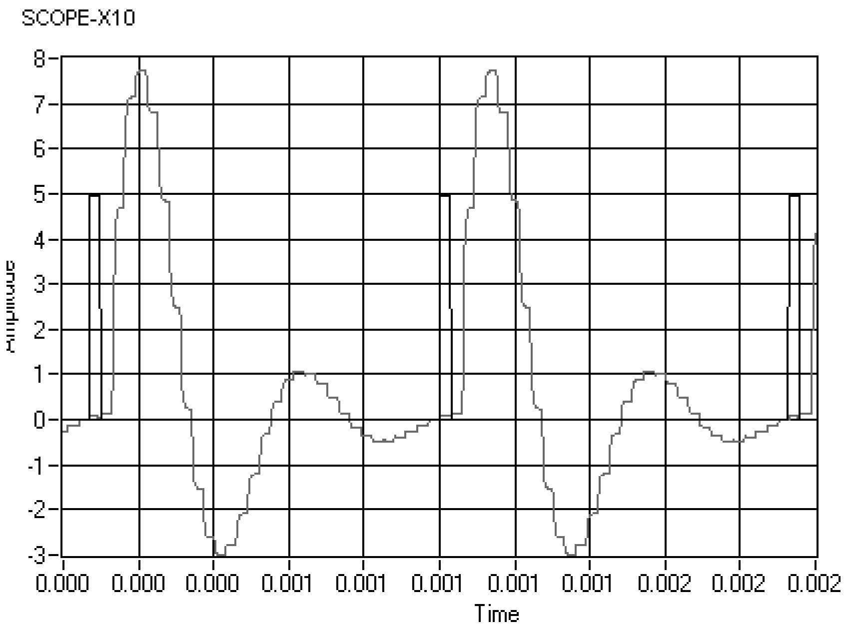
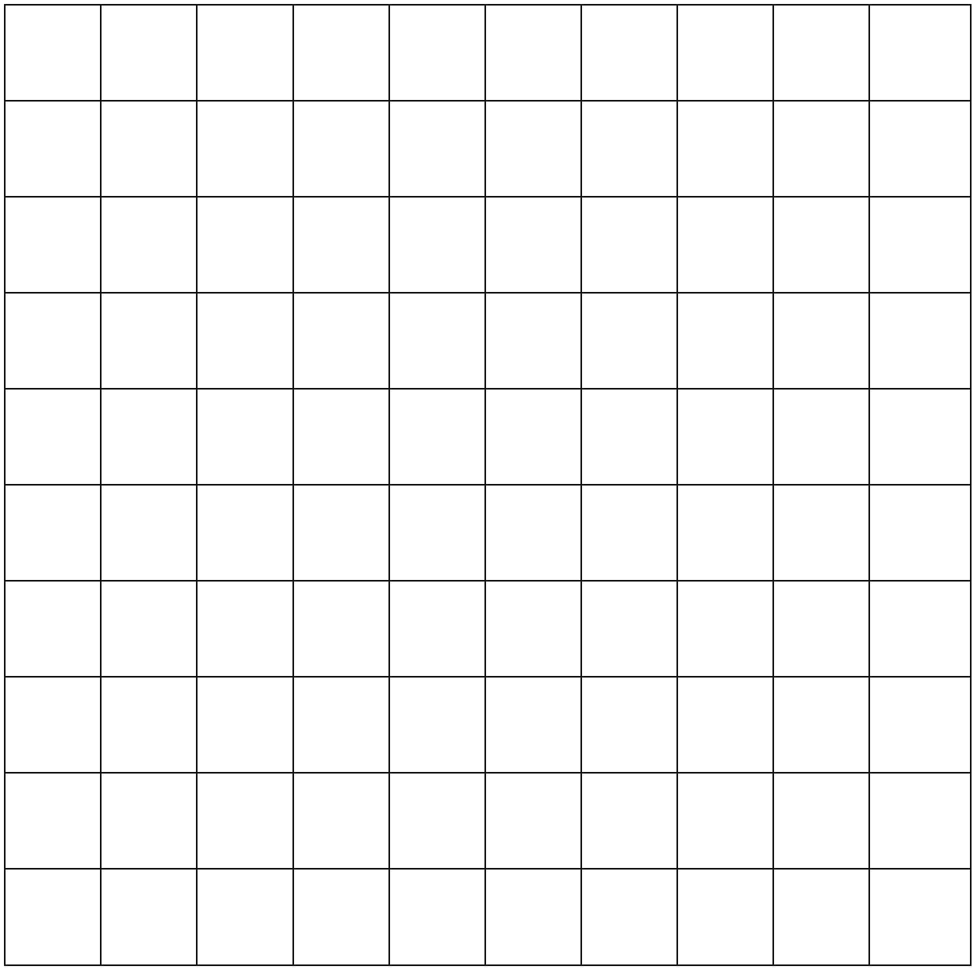


Figure 7: typical pulse response

3. Take a look at the complete response with the unit pulses further apart by changing the SEQUENCE GENERATOR DIPs to the UP:DOWN and DOWN:DOWN positions. Confirm for yourself that the oscillations reduce to zero.



Graph 4**:pole only response plot**

Section3 - IIR with feedforward

3.1 Implement: Second-order filters

In this part we implement and investigate the system in Fig 8. Note that the system with feedforward simply builds upon the previous system with feedback only. It also provides a new output point. The system response x0 for the feedback only, all-pole system is still available as a subset within this new arrangement and is unchanged by the additional feedforward elements. The feedforward elements simply add numerator terms to the overall transfer function which becomes y/u.

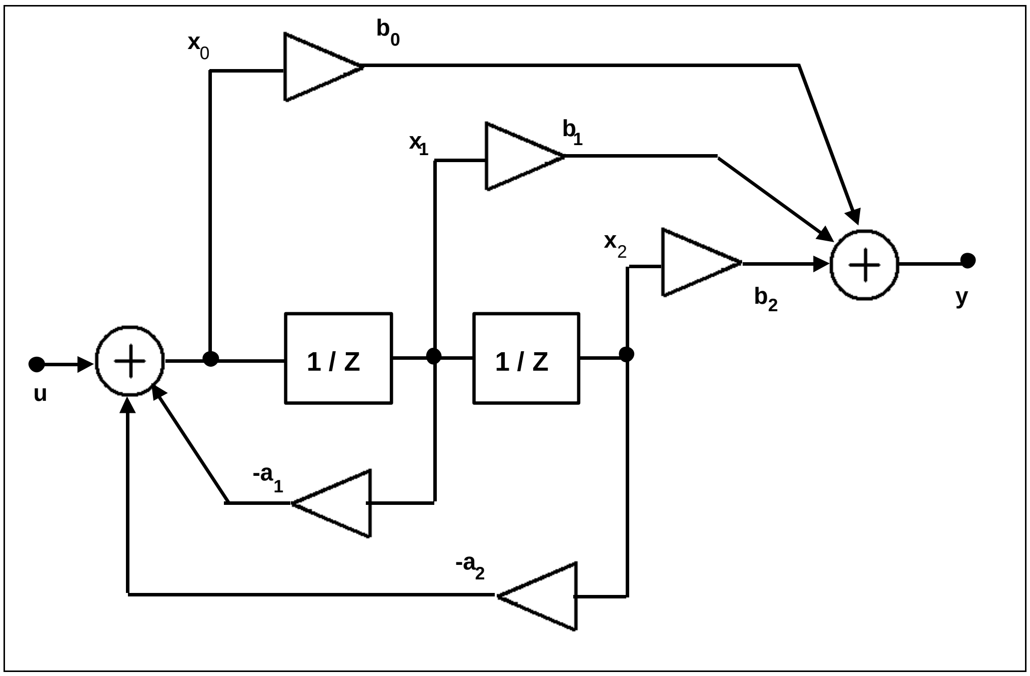


Figure 8: block diagram of 2nd-order Direct-form 2 structure with feedforward.

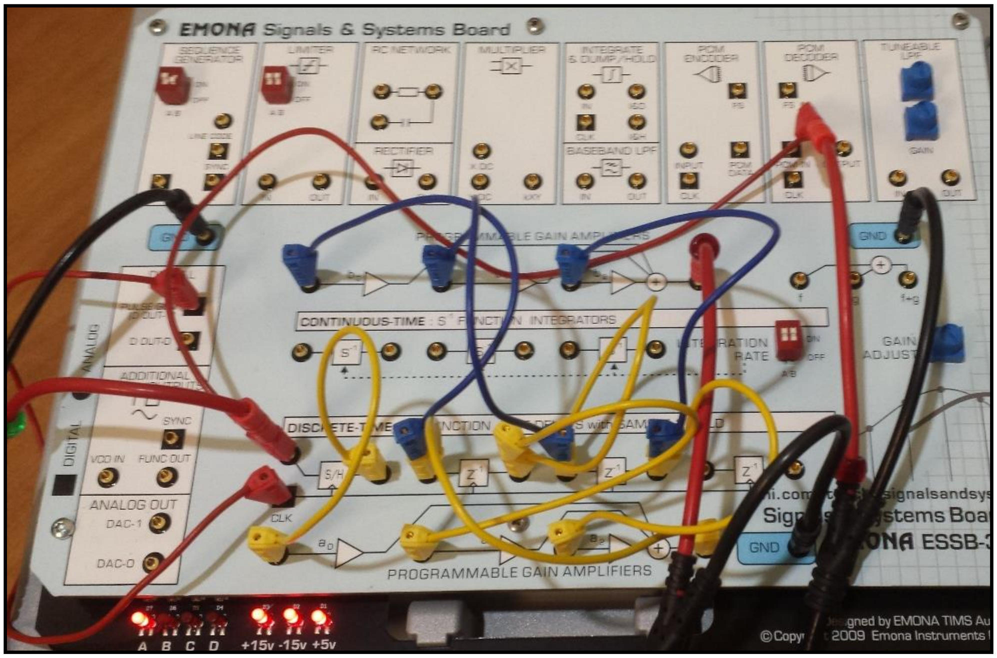


Figure 9: Setup with feedforward and feedback sections implemented, as per Figure 10 below

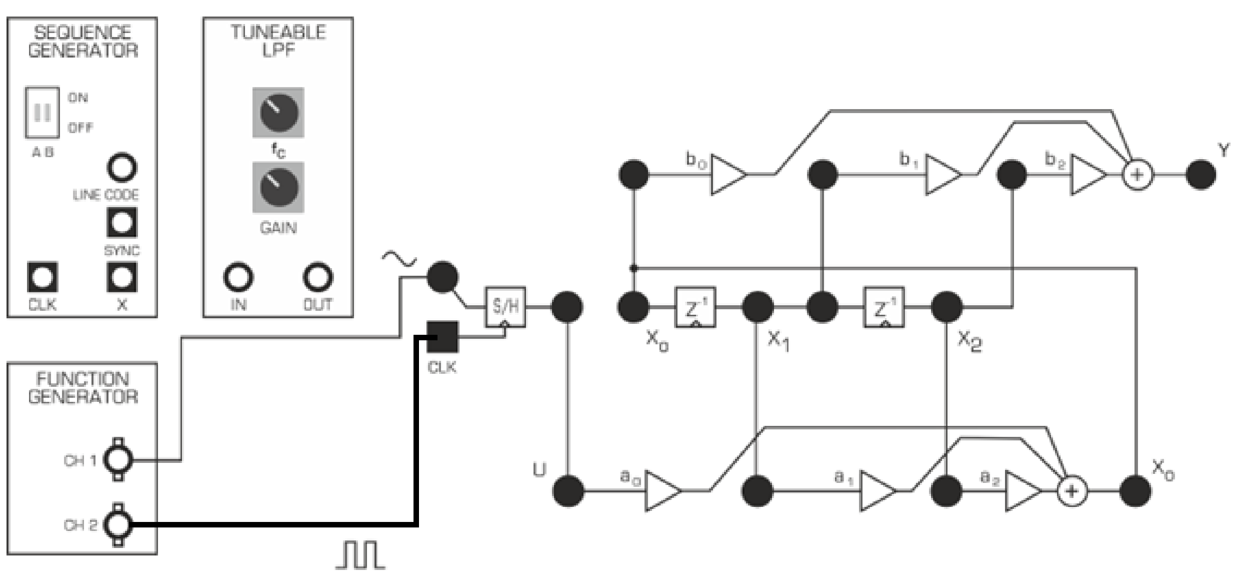


Figure 10: patching diagram for block diagram above

5. Use variable gain ADDER section B in a z-TRANSFORM module to convert the Signals & Systems board model of Section 1 to the system in Fig 8.

6. Implement case (i) in Prep (Q7).

Settings are as follows:

Scope Configuration

Horizontal Timebase	500 us/div
Trigger	Type: Analog edge, Source: sine signal; 0V level

Function Generator Configuration

Waveform	Sine
Frequency FG1	1 kHz
Amplitude	**0.4Vpp**
DC Offset	0V

Function Generator Configuration

Frequency FG2	**20 kHz**
Duty Cycle	0.5 (50%)

ADDER GAINS: b0=1; b1=2; b2=1;   a0=1; a1=+1.6; a2= -0.902

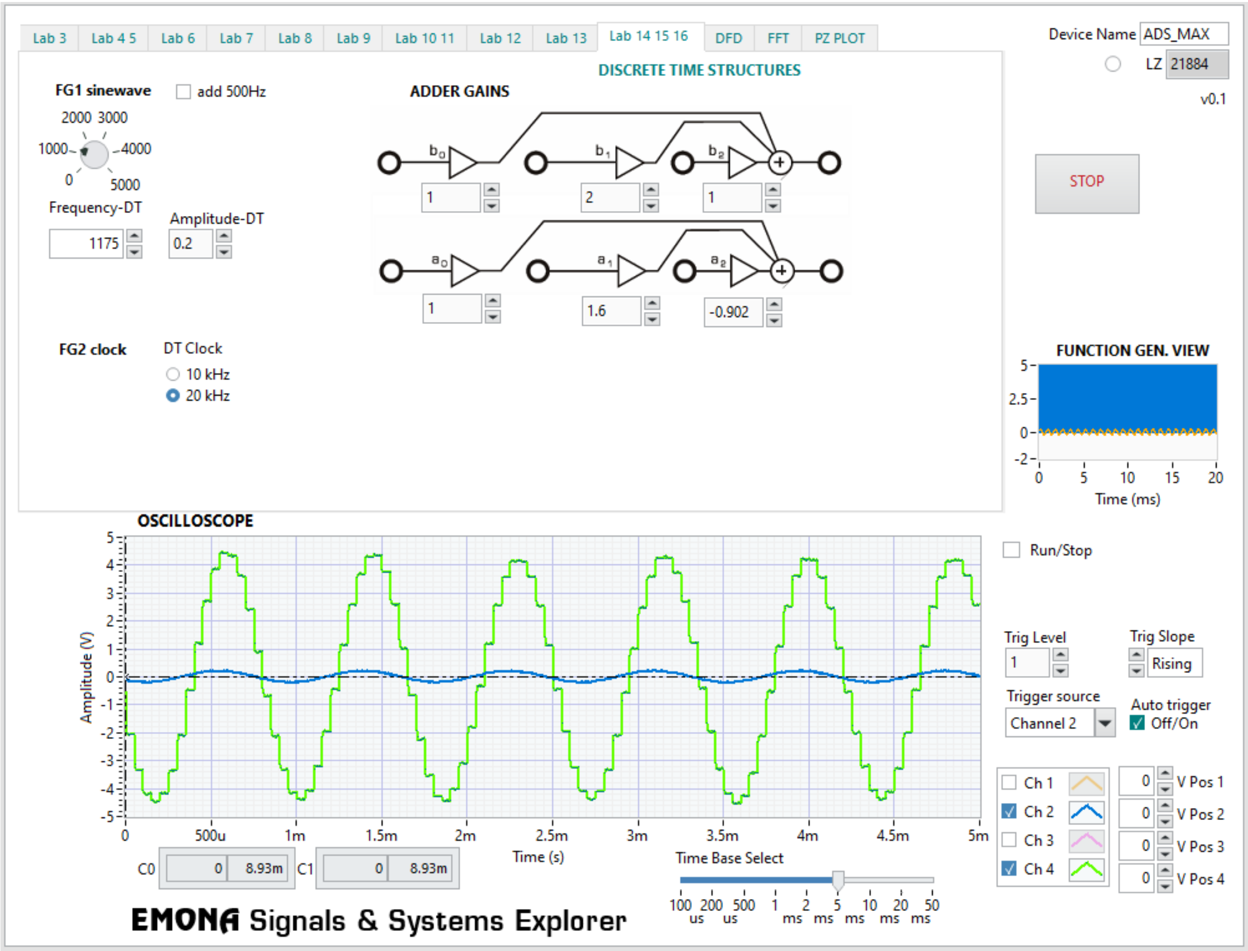


Figure 11: Using the Function Generator to sweep a sinusoid across the spectrum of interest.

Observation: the high pass band gain due to the selection of coefficients resulting in poles very close to the unit circle, as shown in the figure below.

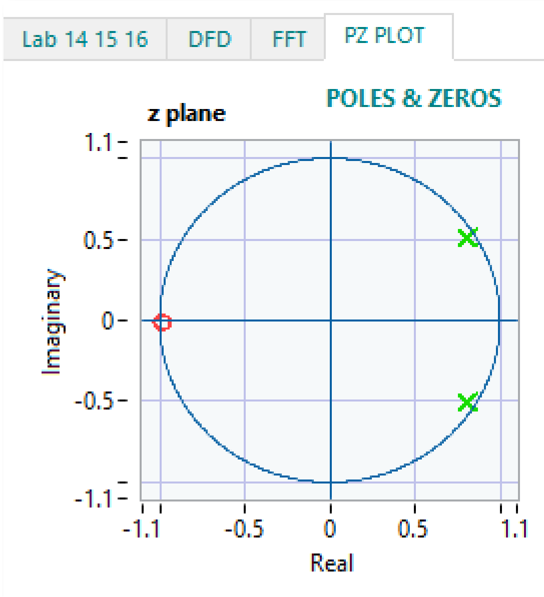
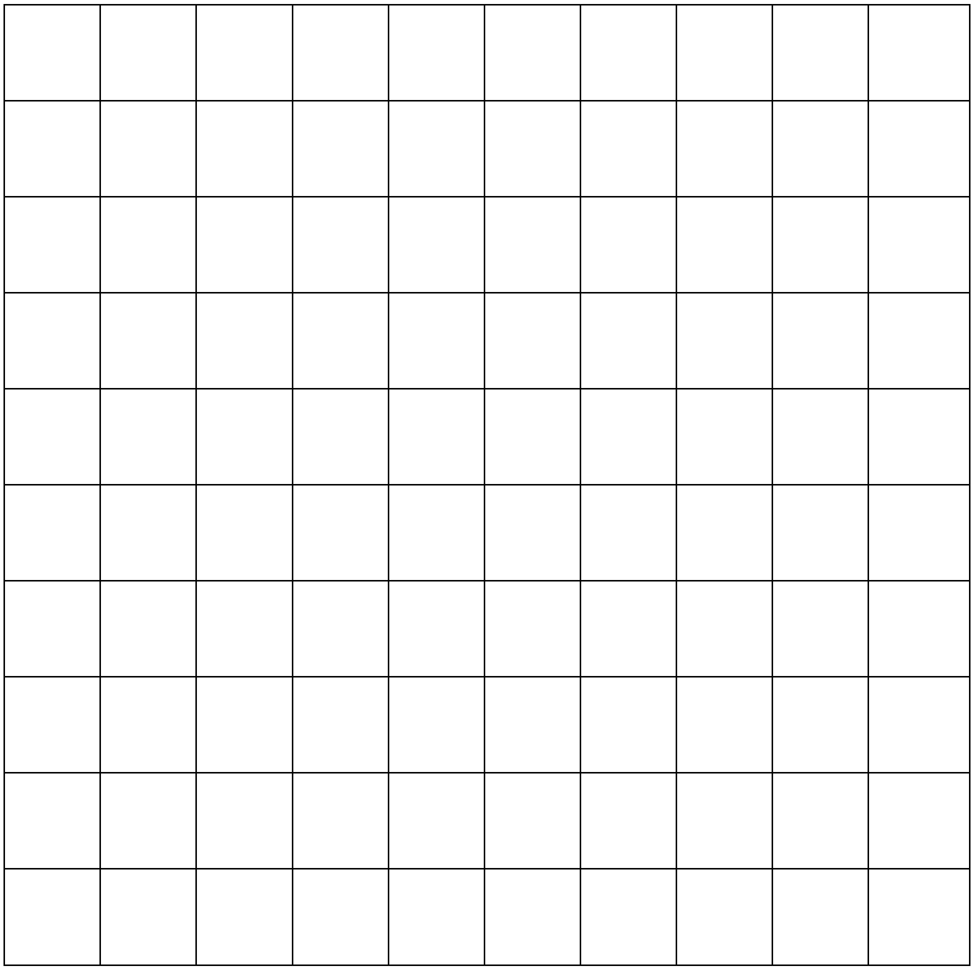


Figure 12: close-up of the PZ PLOT TAB for the current settings

NOTE: The poles are very close to the unit circle. In fact, the pole radius is 0.95. Hence the gain close to the poles is very large. You can use the PZ PLOT to visualize the poles and zeros for any "live" coefficient settings. Zeros are also present at z = -1. (NB: **2 zeros** at present at (-1 +/- 0i))

Measure the magnitude response |y/u| and confirm that it is a lowpass filter.



Graph 5**: plot of responses**

3-1 In the model of Figure 10, adjust a_2 to reduce the peaking to a minimum. As well you will need to reduce the amplitude of the input signal to 0.4Vpp to reduce saturation. Confirm this for yourself. Plot the resulting response and measure the new value of a_2. Calculate and plot the new poles. Obtain an estimate of the theoretical magnitude response with these poles and compare this with the measured curve. Why was a_2 used for this rather than a_1?

3.2 Once you have manually evaluated the response use the Bode TAB to give you an overall frequency response from say 100 Hz to 3kHz and compare this with your own measurements. In particular note the linear scale to appreciate how much amplification placing the poles so close to the unit circle creates in this system.

Remember to use the Bode Analyzer as a convenient and precise instrument in future tasks.

Settings are as follows:

Bode Analyzer Configuration

# Steps	100
Step frequency seperation	50
Frequency Range	100 – 5000 Hz
Peak Amplitude	**0.2Vpk**

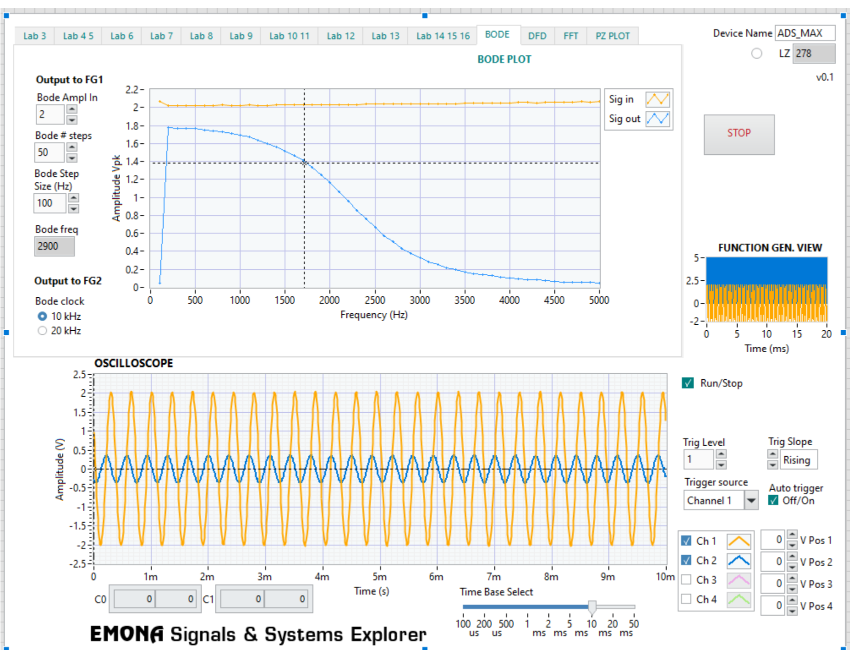


Figure 13: example of Bode plot TAB

3-3 Change the polarity of b_1 in the lowpass of step 19 and show that this produces a highpass. Compare with your findings in Question 7.

NOTE: the following three questions refer to the transfer function coefficient values. Remember to negate the a1 and a2 values when settings them up as ADDER GAINS.

3-4 Repeat for case (iii) in Question 7, that is: $b_0 = 1$, $b_1 = 0$; $b_2 = -1$; $a_0 = 1$; $a_1 =-1.6$; $a_2 =0.902$; Confirm this is a bandpass filter. Tune a_1 and a_2 to obtain a peak at 3.1 kHz and 3dB bandwidth 500Hz. Measure the resulting a_1 and a_2 and plot the new poles. Compare this with your findings in Question 7.

3-5 Implement the following case: $a_0 = 1$, $a_1 = 0$, $a_2 = 0.8$, $b_0 = 0.8$, $b_1 = 0$, $b_2 = 1$. Note that $b_0=a_2$ and $b_1=a_1$. Measure the magnitude response. Confirm it is allpass. Locate the positions of the poles and zeros. Plot them below for your records.

3-6 Change a_1 and b_1 to - 1.6 and confirm the response is still allpass. Examine the behaviour of the phase response. Look for the frequency of most rapid phase variation, and confirm this occurs near a pole.
Plot the poles and zeros below for your records.

3.2 Implement: Viewing spectrum of system with broadband noise input & FFT

As well as sweeping a single frequency signal from the FUNCTION GENERATOR across the spectrum of interest it is also convenient to input a broad range of frequencies at once and view the overall output frequency response of the system. Creating a broadband analog noise signal was covered in Lab 9. That methodology is shown in the figures below. You can revisit this experiment with this setup in place and see the relationships of poles and zeros to system response in real time.

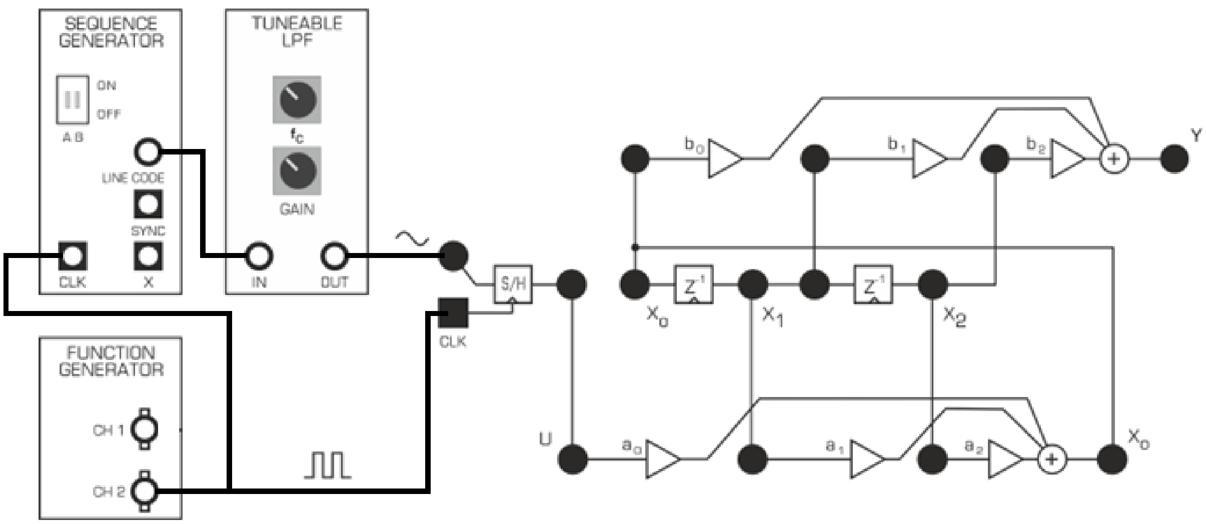


Figure 14: IIR filter with flat noise input

7. Ensure that SEQUENCE GENERATOR DIP switches are set to positions DOWN:DOWN for the long sequence. Set TUNEABLE LPF knobs to fully clockwise for now. Switch on the FFT display to view time and frequency domains simultaneously. Change scope timebase to 1ms/div

View the input to the TUNEABLE LPF with scope CH1, the output of the TUNEABLE LPF on Ch2 and the output Y with scope CH 3.

Set up the coefficients as per Figure 10.

Setting up the input noise signal:

8. i) Reduce the TUNEABLE LPF GAIN by rotating counter clockwise until the output Ch3 signal (the output Y) is no longer saturated ie: less than 12V peak. Your input signal will be quite small, as you know from previous parts of this experiment.

HINT: turn off the AutoScale Y of the scope (right click on the scope) so you can see the signal reduce as you adjust more easily)

ii) View the output of TUNEABLE LPF Set the scope timebase to 2 ms/div so as to have higher resolution on the FFT display. You will observe the noise signals to have a flat spectrum reaching to approximately 20kHz.

Reduce the noise bandwidth to around 4khz by rotating the TUNEABLE LPF block's "Fc" control-counter clockwise. View the noise signal on the SCOPE & FFT windows.

The objective is to prepare a noise signal with a bandwidth of approx. 4kHz and which does not saturate at the power rails out of the filter.

The signals displayed should be similar to Figure 15 showing the bandlimited noise signal. Notice how tiny the input noise signal to the filter is, as shown in the Scope display.

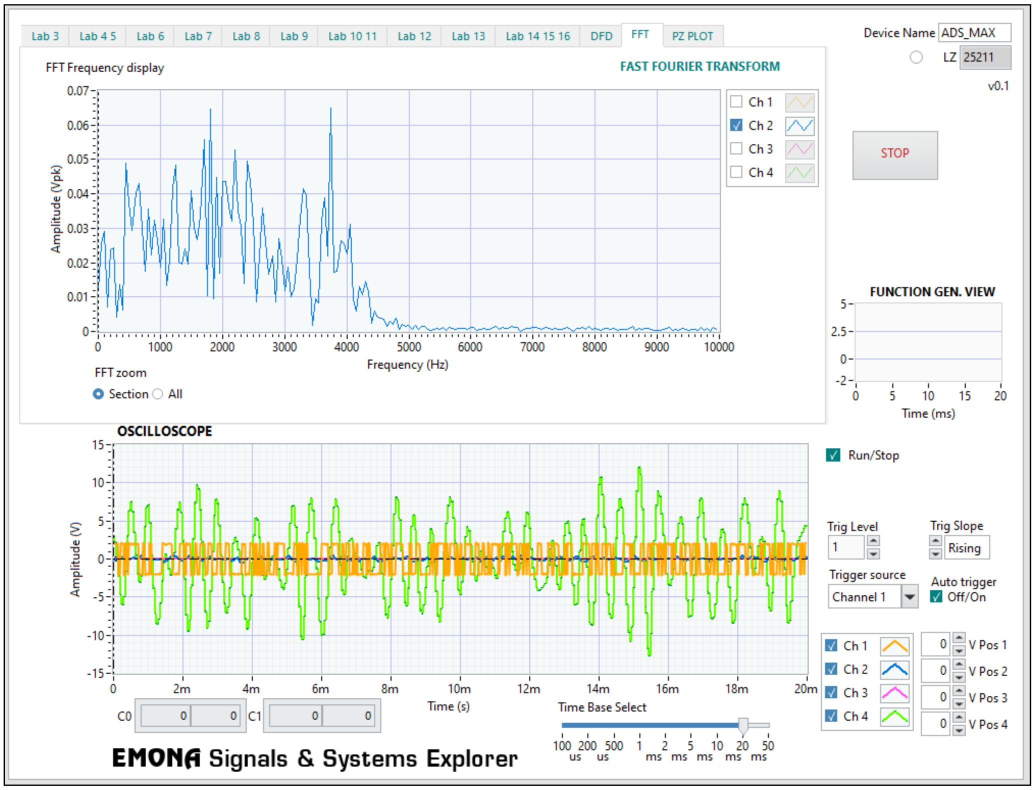


Figure 15: FFT display used to view experiment setup with flat noise spectrum as input signal

Due to high gain the input noise signal level is very small. Note the limited bandwidth of the input noise to maintain a flat input response. (SEQUENCE GENERATOR must be set to long sequence.)

At this point we can see and explore the issues relating to :
-controlling our input signal level and bandwidth
-viewing the response in both time and frequency domains
-setting up a transfer function with appropriate internal gains

We can also confirm that the peak of the response is correct according to the position of the poles. ie: PZ PLOT tells us that poles are at 32 degrees, hence we expect a peak close to 32/360*20,000 = 1777 Hz. You can use the cursors in ZOOM FFT window to confirm this.

3-6 Show your calculation of the where you expect the peak frequency to be using the pole position and sampling frequency.

3.3 Implement: Dynamically varying the poles and zeros to adjust response

9. View the frequency response while slowly varying the value of a_1. You will find that the peak frequency changes.

10. Find a range of a_1 settings that work well and then view the PZ PLOT TAB whilst varying across that range. You will see the poles moving and reflecting the changing $\mathbf{a_1}$ coefficient. (Theory states that $a_1=-2\sigma$, which is the real part of the pole and its conjugate.)

3-7 Confirm this relationship from values displayed on PZ PLOT and show your working here:

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11. Set a_1 back to +1.6. For more resolution vary the setup parameters as required.

3-8 Vary a_2 to vary the gain or peak level of the filter. Notice what happens in the time domain when $a_2 = -1.0$. The filter breaks into oscillation. View the poles again using PZ PLOT while varying a_2. (Theory states that $a_2 = r^2$).

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12. Set a_2 back to -0.9.

3.4 Implement: Using Digital Filter Design toolkit to implement Highpass filters

13. Switch to the DFD TAB. This will allow you to automatically load the computed coefficients for the selected filter onto the Signals & Systems board.

NOTE: Maintain the order of your filter structure <= 2, to match the structure you have built. You will receive a Warning if you exceed 2nd order designs. You are only able to implement second order designs.

As well, you need to ensure that the Frequency Specifications of your filter are compatible ie: for LPF, the stopband frequency is higher than the Passband frequency etc. If you do not adhere to this the program will stop. Simply adjust these values and then restart the SFP program.

14. Connect CH1 to " $\mathbf{x_0}$", and CH2 to "Y" and view the internal signal levels at "x_0" as well as the Y output. Set the TUNEABLE LPF GAIN higher but avoid saturating.These filters have lower internal gains than the previous ones.

APPLY the default highpass filter by pressing the APPLY button, which will load the coefficients to the hardware on the Explorer board.

Vary the filter design type (at the DFD tab by clicking on "DESIGN METHOD" to select) and view the output responses using the FFT display window.

NOTE: Press the APPLY button to transfer the coefficients into the hardware ADDER gains when you are ready to do so.

Set timebase to 2ms/div.
Again you can use the cursors to compare the actual performance to theory and design.

You can expect to see a display like so:

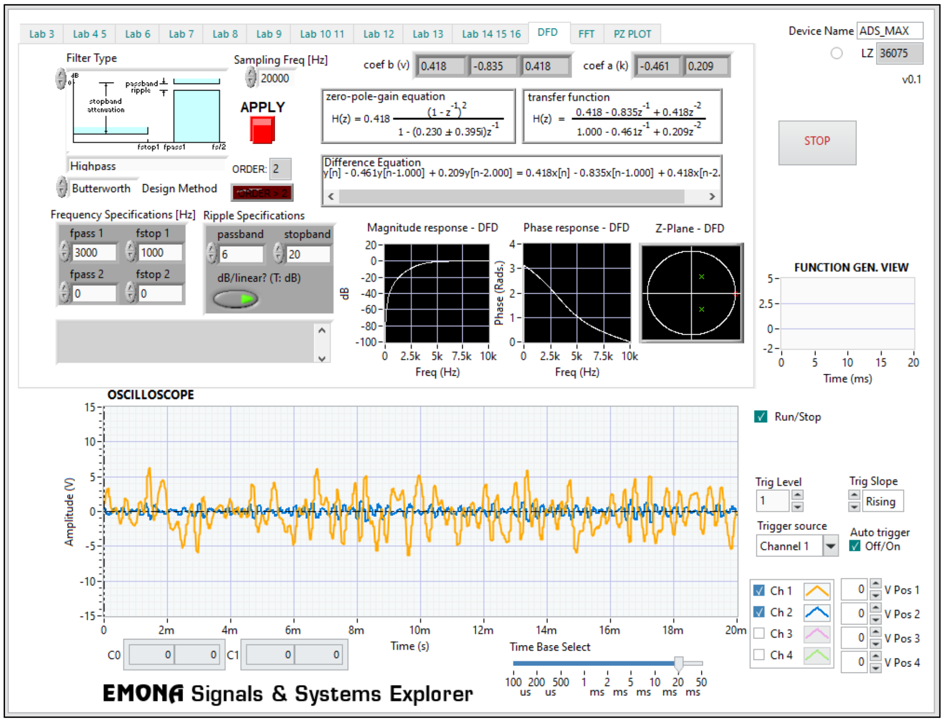

Figure 16: DFD TAB used to design filters and setup coefficients to the patched Signals & Systems board

Default values are chosen to enable students to see textbook like responses which they can easily measure.

3-9 Confirm that the Signals & Systems board hardware performs as designed by theory in terms of notch positions etc. You will have to use the zero positions mostly in these cases. Why ?

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3-10 Try varying design values and take note of the ORDER of the filter designed. NOTE that the Signals & Systems board experiment we have implemented can only support a 2nd order structure.

HINT: Stop the running code, make your changes, check that the settings are consistent with your Design chosen, etc and then restart to implement.
The Filter Type will specify what needs to happen. Remember to press APPLY to load the hardware.

Note your observations.

3-11 Explain what is happening around the 20kHz region of the spectrum of the output ?

Lab Manual:
Signals & Systems

Using the EMONA Signals & Systems Explorer board for ADS MAX

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