Module 1: DSP Introduction & Z-Transforms

Please load J-DSP by pressing the 

button of the quick tour of J-DSP web page. You need to establish and connect all blocks mentioned in this text manually. If you are running from the web page, using J-DSP scripts, everything is loaded automatically.

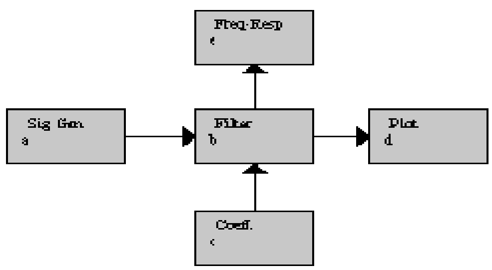
Digital filters Overview:

Digital filters are programmable filters whose purpose is to allow the desirable portion of the input signal to pass and cut off the part of the signal that is unwanted

DEMO Overview:

This DEMO is intended to familiarize the students or participants with some basic concepts in digital filters. It is divided into two parts: part A covers some basic concepts in digital filters and part B goes through a speech filtering example.

The student or participant will make use of the J-DSP that simulates a source-filter configuration. In J-DSP, a simple simulation of digital filtering consists of 5 blocks, as shown below.



1. The source (***Sig Gen***) block: A signal generator that generates the input signal to be filtered. The user can choose from a variety of input signals (step function, sinusoid, triangular, exponential, etc...).

2. The filter (***Filter***) and the filter coefficient (***Coeff***) blocks: By changing the filter coefficients we can change the frequency response of the filter. Figures 1 and 2 show the ideal frequency response of the low-pass and high-pass filters, respectively.

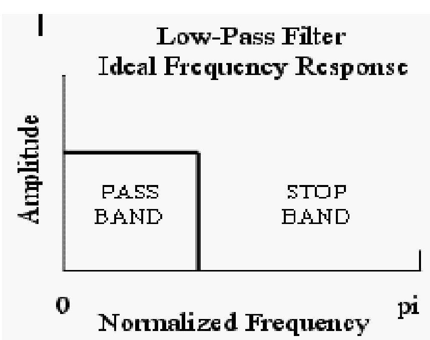


Figure 1

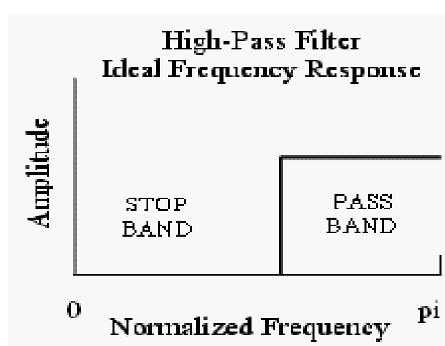


Figure 2

3. The frequency response (***Freq Resp***) block plots the response of the filter depending on the filter coefficients. It plots the normalized frequency versus the amplitude of the filter response.

The sampling frequency is set at 8 kHz the frequency that most telephony signals are sampled at.

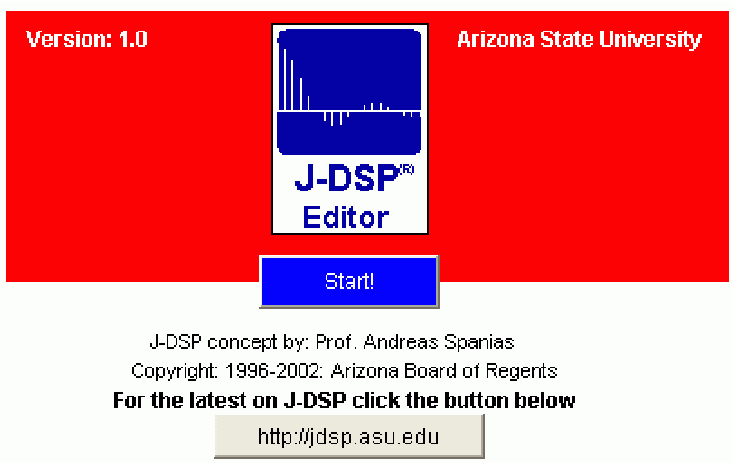
In J-DSP, all frequencies are referenced to the normalized frequency. A simple formula that shows the relationship between the normalized frequency and the actual frequency is: $\mathbf{\Omega = 2\pi f / fs}$ where $\mathbf{\Omega}$ is the normalized frequency, **f** the actual frequency, and **fs** is the sampling frequency.

For example in J-DSP when $\mathbf{\Omega}$ = pi then the actual frequency is equal to **fs/2.** (f= fs/2).

4. The plot (***Plot***) block basically shows the output signal. In other words it plots the filtered input signal.

<u>Note that in order for J-DSP to execute any parameter changes made on the blocks the user must press the UPDATE button located on the bottom of each block window.</u>

PART A: Basics on digital filters



Press <u>Start</u> on the J-DSP Editor and follow the instructions below

STEP 1: The signal generator is feeding the filter with a **low frequency** sinusoid with amplitude equal to one. Observe the *filter coefficients* and the *frequency response* of the filter.

- What is the normalized frequency of the input signal? Ω = ____

Remember the general transfer function of a digital filter:

$$H(z) = \frac{\sum_{i=0}^{10} b_i z^{-1}}{1 + \sum_{i=1}^{10} a_i z^{-1}}$$

In our case the general transfer function simplifies to:

$$H(z) = \frac{b_0 + b_1 z^{-1}}{1 + a_1 z^{-1}}$$

- Take a note of the coefficients: a0 = _____, a1 = _____, b0 = _____, and b1 = _____ .
- Write the transfer function:

$$H(z) = \frac{}{}$$

- Observe the frequency response plot and state the kind of digital filter that is realized? (All-pass? Low-pass? High-pass?) _____________
- Observe the output. Has the input sinusoid being altered by this filter? (Check the output plot window and take a note of the amplitude of the signal. You may choose to view the continuous or discrete output signal by using the menu options of the ***Plot*** block)

STEP 2: Change the following filter coefficient. Set **b1 = 1.0**, and observe the frequency response of the filter.

- Take a note of the new coefficients: a0 = _____, a1 = _____, b0 = _____, and b1 = _____ .
- Write the new transfer function:

$$H(z) = \frac{}{}$$

- What kind of digital filter is implemented? (Low-pass? High-pass?) _____________
- Observe the output. Did the amplitude of the output signal increase or decrease with respect to the input signal? Write the amplitude of the output signal, | y[n] | = ____.

STEP 3: Change the following filter coefficient. Insert a minus sign in front of the **b1** coefficient. (therefore **b1 = -1.0**)

- Observe the frequency response plot of the filter.
- Write the new transfer function:

$$H(z) = \frac{}{}$$

- What kind of digital filter is realized now? (Low-pass? High-pass?) _____________
- Observe the output. Did the amplitude of the output signal increase or decrease with respect to the input signal? Write the amplitude of the output signal, | y[n] | = ____.
- What can you conclude about the effect of a high-pass filter on a low frequency input signal?

STEP 4: Make the following changes to the filter coefficients. Set **b0 = 1.0, b1 = 0.0 and a1 = -0.9**

- Observe the frequency response plot of the filter.
- Write the new transfer function:

$$H(z) = \frac{}{}$$

- What kind of digital filter is implemented now? (Low-pass? High-pass?) _____________
- Observe the output. Did the amplitude of the output signal increase or decrease with respect to the input signal? Write the amplitude of the output signal, | y[n] | = ____.

STEP 5: Change the following filter coefficient. Set **a1 = 0.9**.

- Observe the frequency response plot of the filter.
- Write the new transfer function:

$$H(z) = \frac{}{}$$

- What kind of digital filter is implemented now? (Low-pass? High-pass?) _____________
- Observe the output. Did the amplitude of the output signal increase or decrease with respect to the input signal? Write the amplitude of the output signal, | y[n] | = ____.

STEP 6: In order to understand the difference of the effect of filtering between the low and high frequency sinusoids set the signal generator for a **high frequency** sinusoid with a normalized frequency equal to **0.8** x pi. (**Ω = 0.8 x pi)**.
Also, set all the filter coefficients to zero except a0 and b0. (a0 = 1.0 and b0 = 1.0).

- Repeat Steps 1 through 5.
- Note the difference between the results when the input signal was a low frequency sinusoid with those when the input signal was a high frequency sinusoid.
- Compare the output of the low-pass and high-pass filters for low frequency sinusoids.
- Compare the output of the low-pass and high-pass filters for high frequency sinusoids.

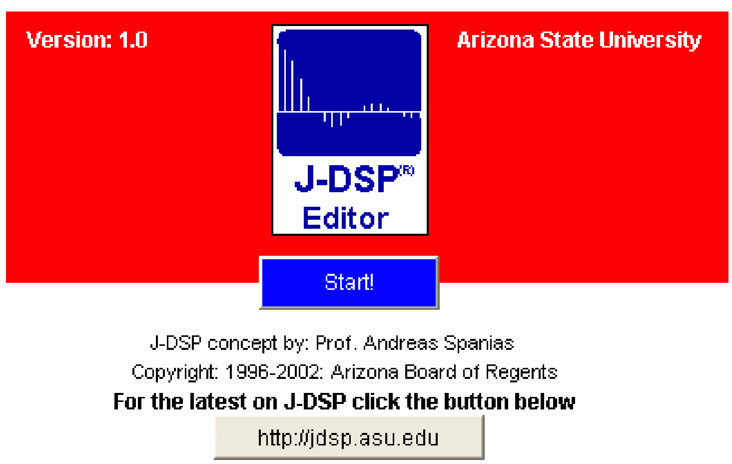
STEP 7: Set the signal generator for a rectangular input, a step function u[n], with pulsewidth = 64.

- Observe the output on the ***Plot***.
- Is there a transient response (region)? _______
- How long, measured in terms of samples, is the transient response? No. samples = _______ (Hint: Use the cursor on the plot window and choose between continuous and discrete representation of the output signal).

*THIS IS THE END OF PART A. PLEASE **CLOSE** THE J-DSP EDITOR WINDOW.*

PART B: Speech Example

****For this part of the DEMO you will need a pair of speakers properly installed on your computer****



Press <u>Start</u> on the J-DSP Editor and follow the instructions below

STEP 1: Press the *<u>Rerun</u>* button of the long signal generator block (***Sig. Gen (L)***) then, press the green button *<u>Play</u>* of the sound player (***Snd Plyr***) block.

- You have heard the original audio sample without being subjected to any filtering.
- Verify that the transfer function defined by the given filter coefficients is an all-pass filter.
- Write the transfer function:

$$H(z) = \underline{\qquad\qquad}$$

STEP 2: Change the following filter coefficients: Set **a1 = -0.9**.
Repeat the instructions given in STEP 1.

- What range of the audio spectrum has survived filtering? (Low frequencies? High frequencies?)
- What kind of digital filter is implemented? (Low-pass? High-pass?) ______________
- Write the transfer function:

$$H(z) = \frac{\qquad}{\qquad}$$

STEP 3: Change the following filter coefficients. Set **a1 = 0.9.** Repeat the instructions given in STEP 1.

- What range of the audio spectrum has survived filtering? (Low frequencies? High frequencies?)
- What kind of digital filter is implemented? (Low-pass? High-pass?) ______________

$$H(z) = \frac{\qquad}{\qquad}$$

Questions:

- What have you learned from this DEMO?

- Did you understand the effect of digital filtering on sinusoids? ◉ YES ○ NO

- Did you understand the difference between low and high pass filtering? ◉ YES ○ NO
- Did you understand the significance of digital filters on the speech example? ◉ YES ○ NO

- Did you find this DEMO helpful? ◉ YES ○ NO

<u>**THIS IS THE END OF THE EXERCISE**</u>