

Reverberation Devices

1.0 Aim

In this work you will explore reverberation devices.

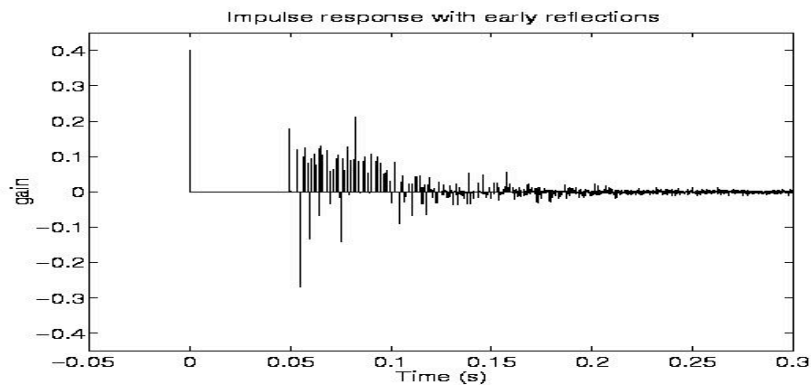
2.0 Learning Outcomes

You will be able to:

- Implement algorithms for reverberation in MATLAB.
- Investigate how commercial reverberation devices work through text books, journals and web searches.
- Describe the audible effects of the processors and compare to commercially available processors.

3.0 REVERBERATION MODELING

Creating an artificial room impulse response.



Impulse response typically modelled as

1. Direct sound,
2. early reflections (FIR filter) and
3. exponentially decaying late reverberation (with an IIR filter to reduce computation):

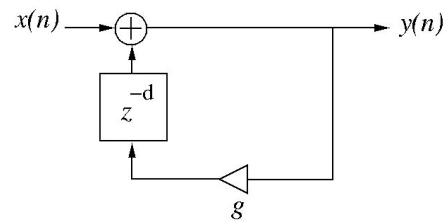
Early reflections often derived from a room model, e.g., as reflections caused by an image source.

Late reverberation implemented often with recursive (IIR) filter structures, because a longer response can be produced with less computation. Important aims are:

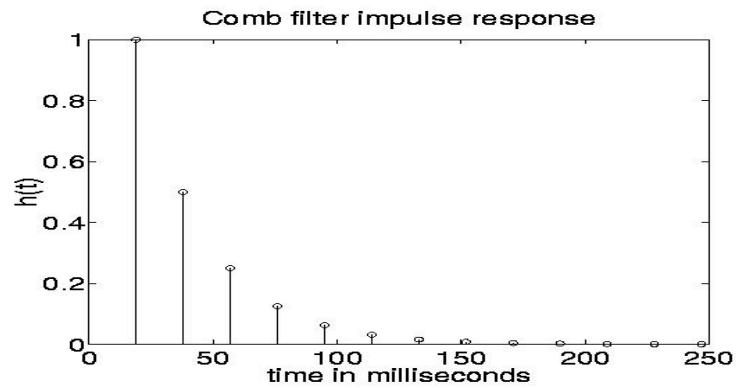
- exponentially decaying response (with adjustable reverberation time)
- sufficient reflections density (to avoid flutter echoes)
- sufficient modal density = modes (resonances) per herz (to avoid coloration, ringin tones, metallic sound)
- incoherence between different channels (monophonic reverberation does not give a good spatial impression)

LATE REVERBERATION MODELING:

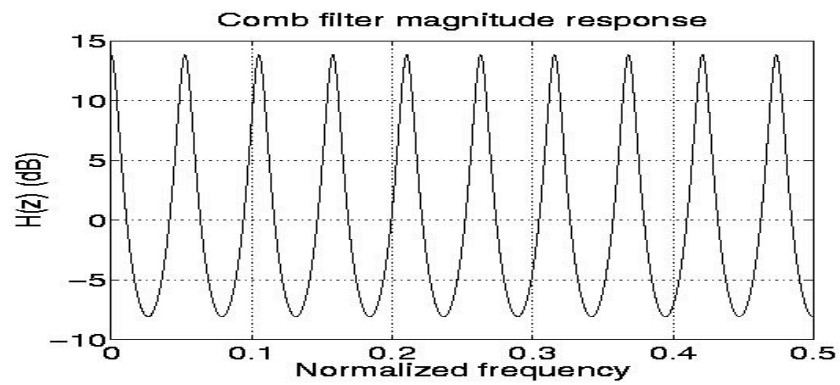
Comb filter: Commonly used basic component:



Comb filter response: Provides exponentially decaying impulse sequence (can be thought of as simulating a single room mode). The advantage is that the decay time ("reverberation time") can be used to define the gain of the feedback loop. The reflection density depends on the length of the delay line:



Comb filter frequency response: Evenly spaced modes -> sounds very bad (metallic)



Example filter structure of an early reverberator (for producing late reverberation):

Digital Allpass Filter

The allpass filter has the unique property that it passes all frequencies equally. This property makes it useful in reverberator design because it increases echo density while reducing signal coloration (an allpass filter is NOT transparent to transient signals, as it does have a phase response). The basic transfer function of an allpass filter is

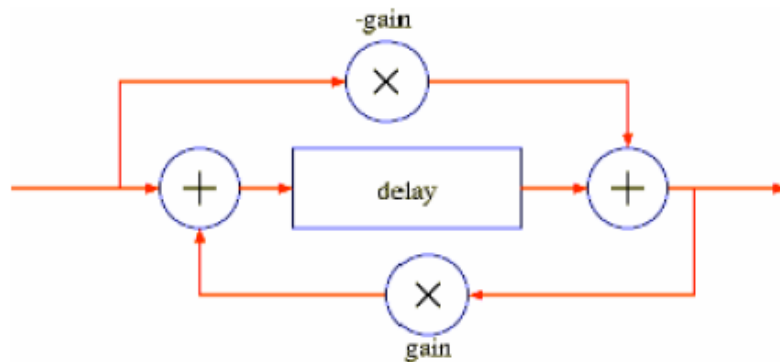
$$H(z) = \frac{z^{-1} - a^*}{1 - az^{-1}} = \frac{1 - a^*z}{z - a}$$

In a pole-zero plot, the poles and zeros occupy conjugate reciprocal locations. The proof of the flat frequency response is as follows:

$$H(\omega) = \frac{e^{-j\omega} - a^*}{1 - ae^{-j\omega}} = \frac{1 - a^*e^{j\omega}}{e^{j\omega} - a}, \quad H^*(\omega) = \frac{e^{j\omega} - a}{1 - a^*e^{j\omega}}, \quad |H(\omega)|^2 = H(\omega) \cdot H^*(\omega) = \frac{1 - a^*e^{j\omega}}{e^{j\omega} - a} \times \frac{e^{j\omega} - a}{1 - a^*e^{j\omega}} = 1$$

The structure of an allpass filter is similar to the comb filter, but it contains an additional feedforward path. The structure is shown in Figure 1, and the impulse response is shown in Figure 2. The delay can be any positive number of samples and still feature a flat frequency response.

Flow diagram of an allpass filter.



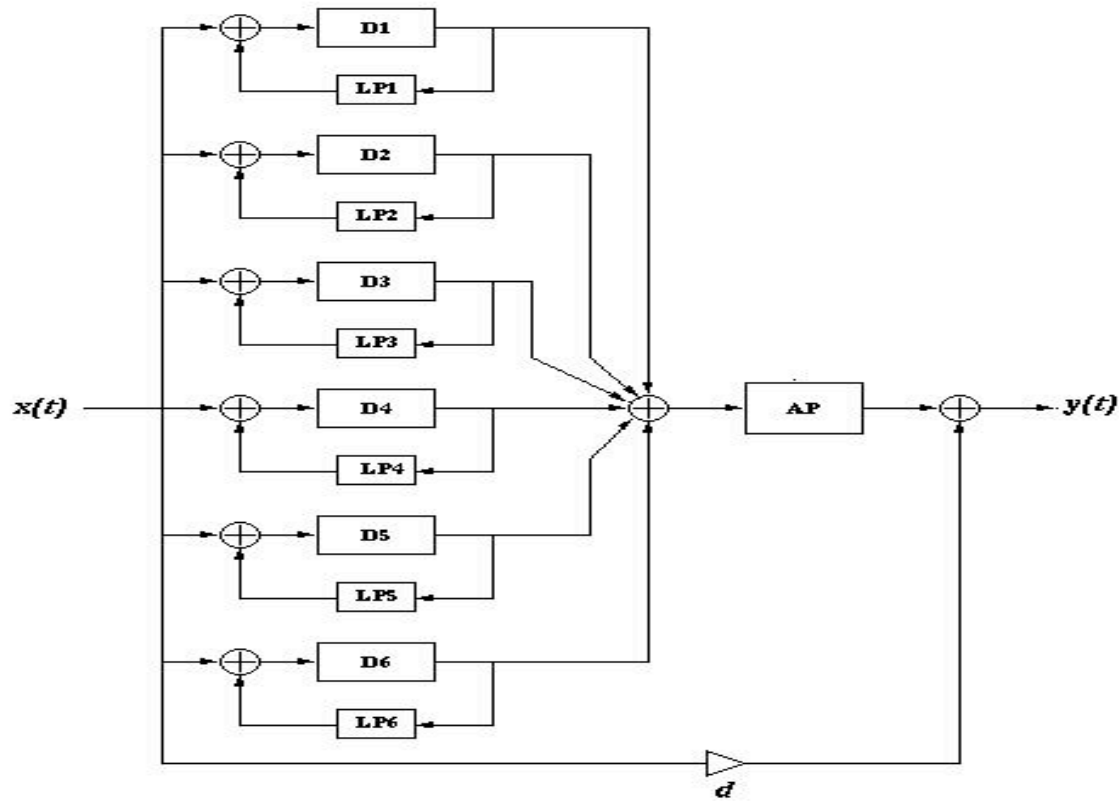
In reverb applications, the delay used in an allpass filter is generally measured in milliseconds. This keeps the 'reflections' spaced apart in a similar fashion to a room's impulse response.

Examples:

Shroeder's reverberator:

Parallel comb filters with different delay lengths (to simulate modes of a room, and sound reflecting between parallel walls) + and allpass filter to increase the reflection density (diffusion). Lowpass filters inserted in the feedback loops to alter the reverberation time as a function of frequency (shorter reverberation time at higher frequencies is caused by air absorption and reflectivity characteristics of walls).

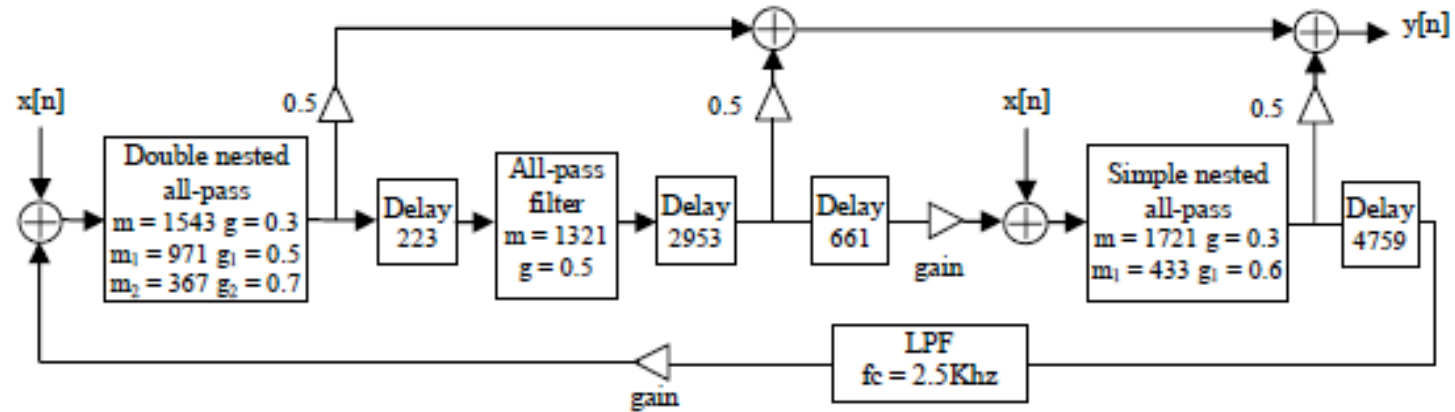
The lowpass filters implement a dc-attenuation, and a frequency dependent attenuation. They are different in each comb filter because their coefficients depend on the delay line length (and the desired reverberation time).



Comb filters still not good enough: Often have a fluttering and metallic response.

Gardner's reverberator

It is based on nested all-pass *filters*, where the delay is replaced by a series connection of a delay and another all-pass filter. Gardner suggested three structures for different size rooms. A structure of the medium room reverberator is



Convolution reverberators

These are state of the art reverberators, based on a convolution between an actual room response and the incoming audio signal. Since room responses have long reverberation time, the impulse response is very long and requires numerically heavy processing. Some of the methods involve efficient ways of FIR filtering using the FFT. Hybrid solutions that mix real impulse early reverb with synthetic late reverberators also exists.

Getting Started with Artificial Reverb

Schroeder reverberation device

In this exercise, you will implement a Schroeder reverberator and test it with both impulse and music, for example a piano signal.

Exercise 1:

Enable the diary function.

Write a MATLAB function (comb.m) as follows.

Comb Filter Function:

The functions parameters are input sound vector x , delay in milliseconds, and gain factor. Use the following template to fill in the missing lines of code marked by ?. Note that instructions and hints are given as comments in the code template.

```
function y = comb(x,d,g,fs)
```

```
%Comb filter
```

```
% x – input vector
```

```
% d – delay in milliseconds
```

```
% g – gain factor
```

```
% fs – sampling frequency
```

```
nd = round(?);
```

```
% nd is the delay in terms of number of samples. You need to translate the delay in milliseconds to number of samples.
```

```
% Hint – use sampling frequency to relate delay to number of samples.
```

```
% round is used to turn it into the closest integer.
```

```
a(1) = 1;
```

```
a(nd) = -g;
```

```
b(1) = 1;
```



```
y = filter(b,a,x);
```

Download blip.wav from <http://music.ucsd.edu/~sdubnov/Mu172/Sounds/>

Run it through this script (as the x vector above). Save the result as [yourname]blipComb.wav.

Exercise 2:

Write an All Pass function as follows:

The functions parameters are input sound vector x, delay in milliseconds, and gain factor. Use the following template to fill in the missing lines of code marked by ?.

```
function y = allpass(x,d,g,fs)
%Allpass filter
% x – input vector
% d – delay in milliseconds
% g – gain factor
% fs – sampling frequency
```

```
nd = round(?);
```

% nd is the delay in terms of number of samples. Use the nd code from combfilter.

```
a(1) = 1;
```

```
a(nd) = -g;
```

```
b(1) = ?;
```

```
b(nd) = ?;
```

% Your task is to write the appropriate values of b so that the filter will be an allpass.

% Remember – the “trick” is to place a zero in a conjugate reciprocal location to the pole in zplane.

```
y = filter(b,a,x);
```

Create [yourname]blipAllpass.wav.

Exercise 3:

Plot the frequency response (phase and amplitude) of the following comb and the allpass filters.

Use the following gains and delay:

Comb: gain = 0.9, delay = 2msec, fs = 44100;

Allpass: gain = 0.8, delay = 2msec, fs = 44100;

1. Explain what you see. How is the delay related to the location of the peaks in comb filter? How is it related to the staircase shape of the phase response in allpass?

Exercise 4:

Use zplane to plot the poles and zeros of an allpass filter with gain = 0.8, delay = 2msec.

2. Explain what you see. What is the gain? Why? Try to zoom in to see the pole and zero locations. Why are they such? How many poles and zeros are there? Explain how it is related to the order of the filter.

Exercise 5:

Create an impulse response of the allpass filter. Try different delays and gains.

3. How do different gains and delays affect the decay of the impulse (qualitative answer).

Exercise 6:

Write a function Schroeder.m that implements the following signal chain:

Filter a signal x in parallel through 4 comb filters with following parameters

Comb: gain = 0.9, delay = 29msec, 37msec, 41msec, 44msec

Sum the four outputs into one signal and divide by 4 (to avoid extra gain).

Run the result in cascade through two allpass filters with parameters

Allpass: gain = 0.8, delay = 5msec, 1.7msec

Hint: in cascade means the output of the first allpass should become the input of the second allpass.

4. Does it matter which one is first and which is second? Why?

Exercise 7:

Experimental work:

The following terms are used to describe resonators

* Room resonance or coloration – mostly determined by the comb filter.

* Reverberation time - Usually divided into early decay and overall reverberation time. These are determined mostly by allpass parameters.

* Clarity – ratio between impulse response energy in the first 80ms and after that. (in speech it is set to 50ms).

5. Using freqz to plot the impulse response of the filter from Exercise 6. Describe qualitatively the behavior of the filter in terms of the above factors.

Filter blip.wav using the Schroeder.m.

Perform the following three experiments:

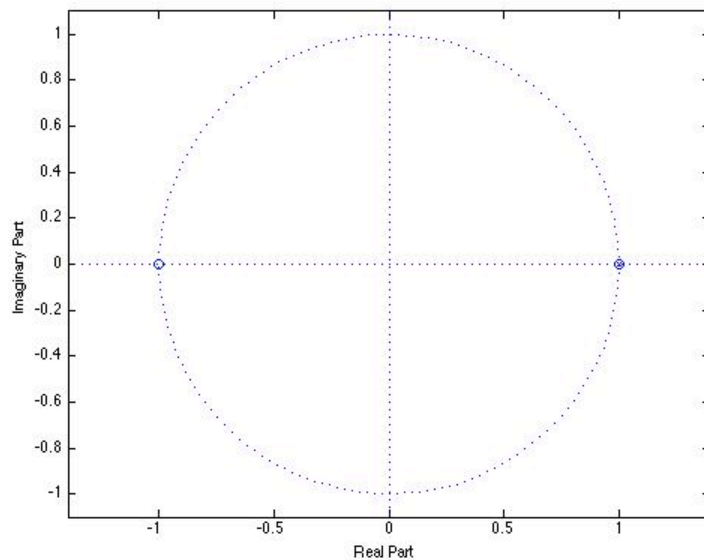
6. Investigate the comb filter operation without allpass. What effect does it create?

7. Investigate allpass alone (filter comb filter). What effect does it create?

8. Investigate a complete filter. Can you hear the different aspects of the processing?

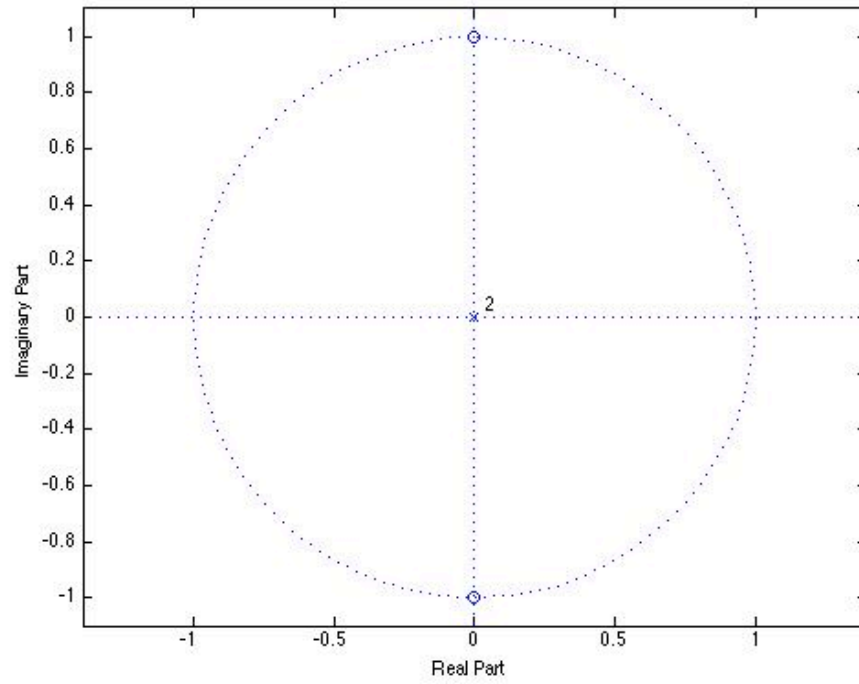
Further Questions:

9. How many roots (zeros) will a polynomial of order 5 have? Order 3? Order 1? Order 1000?
10. For quadratic equation $f(x)=x^2+x$, find its roots. Use the formula $x = [-b \pm \sqrt{b^2-4ac}]/2a$ to solve the problem.
11. Consider the fourth-order filter $y(n)=x(n)+0.7x(n-1)+0.7x(n-2)+0.2x(n-3)+0.2x(n-4)$ and tell if it produce notches or resonances or both? How many?
12. For the following filter $y(n)=x(n)-0.8x(n-1)+0.3x(n-2)-0.6y(n-1)+0.7y(n-2)$, determine the number of poles and zeros.
13. Express the transfer function of the filter above using $e^{j\omega}$. Find and write down the filter coefficients B (forward) and A (recursive) that correspond to this function.
14. Given a zplane graph shown in the following figure, which is the correct behavior?

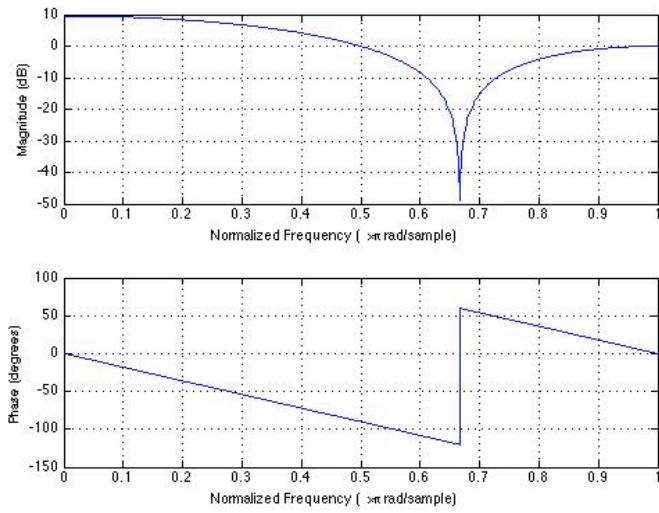


- A. the frequency response has a notch at DC and Nyquist.
- B. the freq. response has a peak (resonance) at 440Hz.
- C. the freq. response has peaks at DC and 300Hz and notches at 500 and 900Hz.

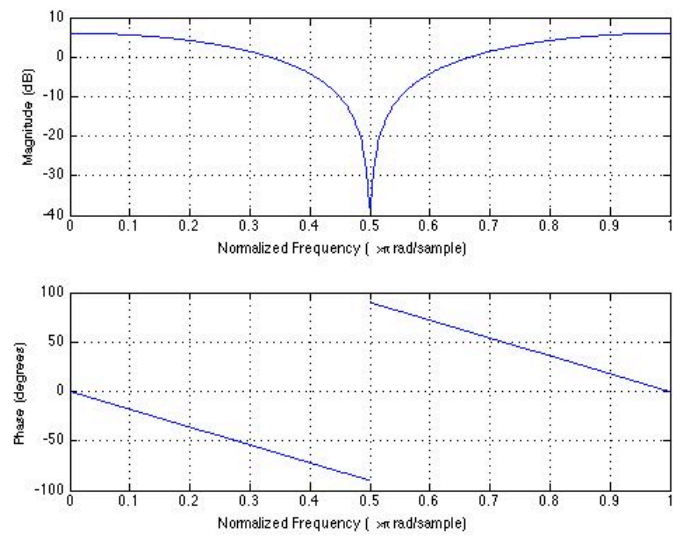
15. Which of the following frequency responses corresponds best to the following zplane graph?



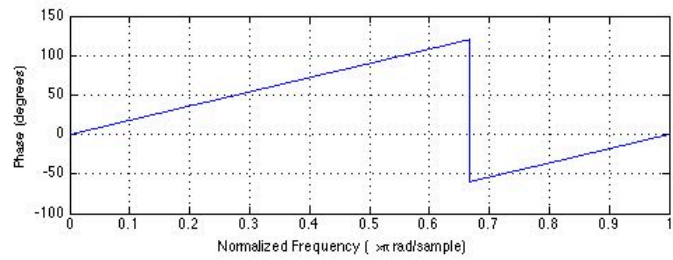
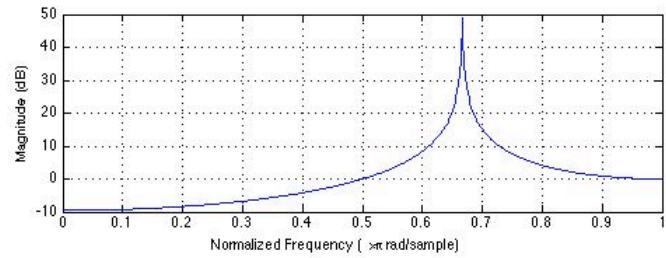
A.



B.

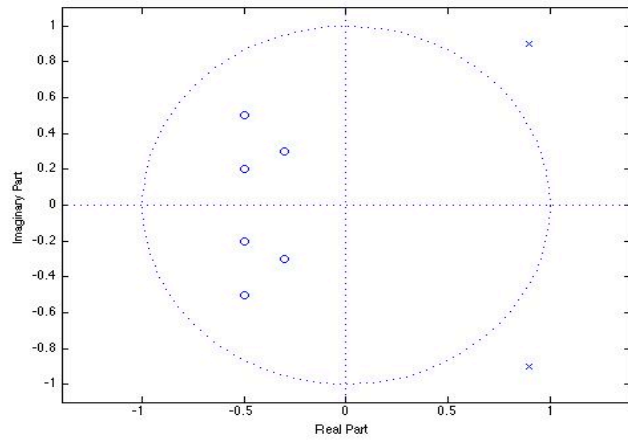


C.

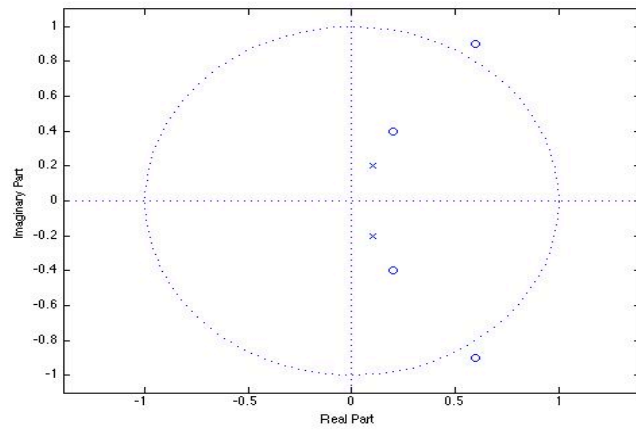


16. Which of the following filters are stable?

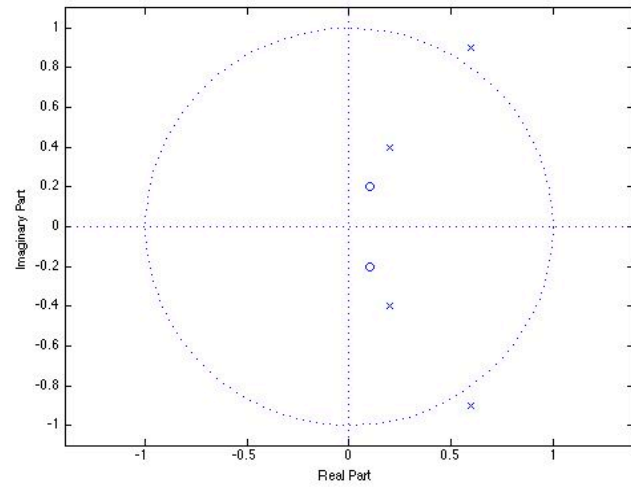
A.



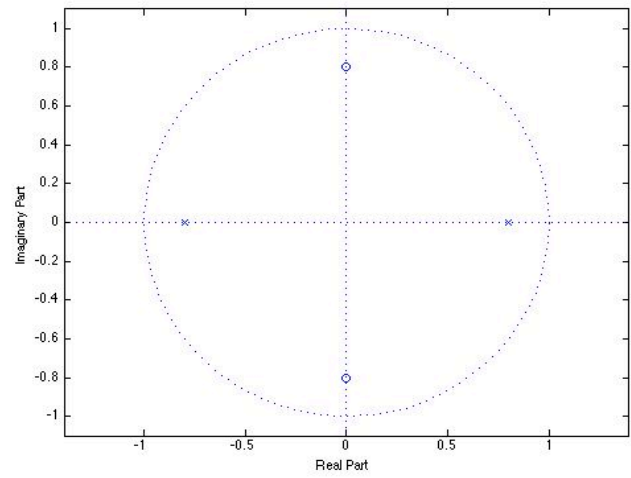
B



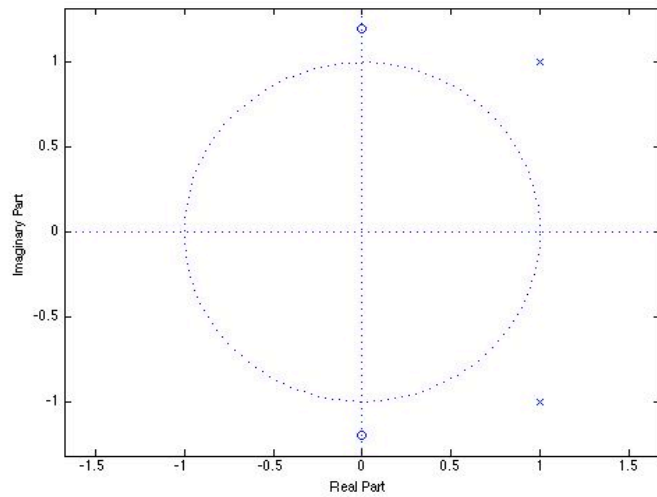
C



D



E



Please zip and email the following to cdwarren@ucsd.edu by Monday May 4th at midnight.

[yourname]diary
[yourname]A3Q1-16
[yourname]comb.m
[yourname]blipComb.wav
[yourname]allpass.m
[yourname]blipAllpass.wav
[yourname]Schroeder.m