EEE 312 – Lab 5

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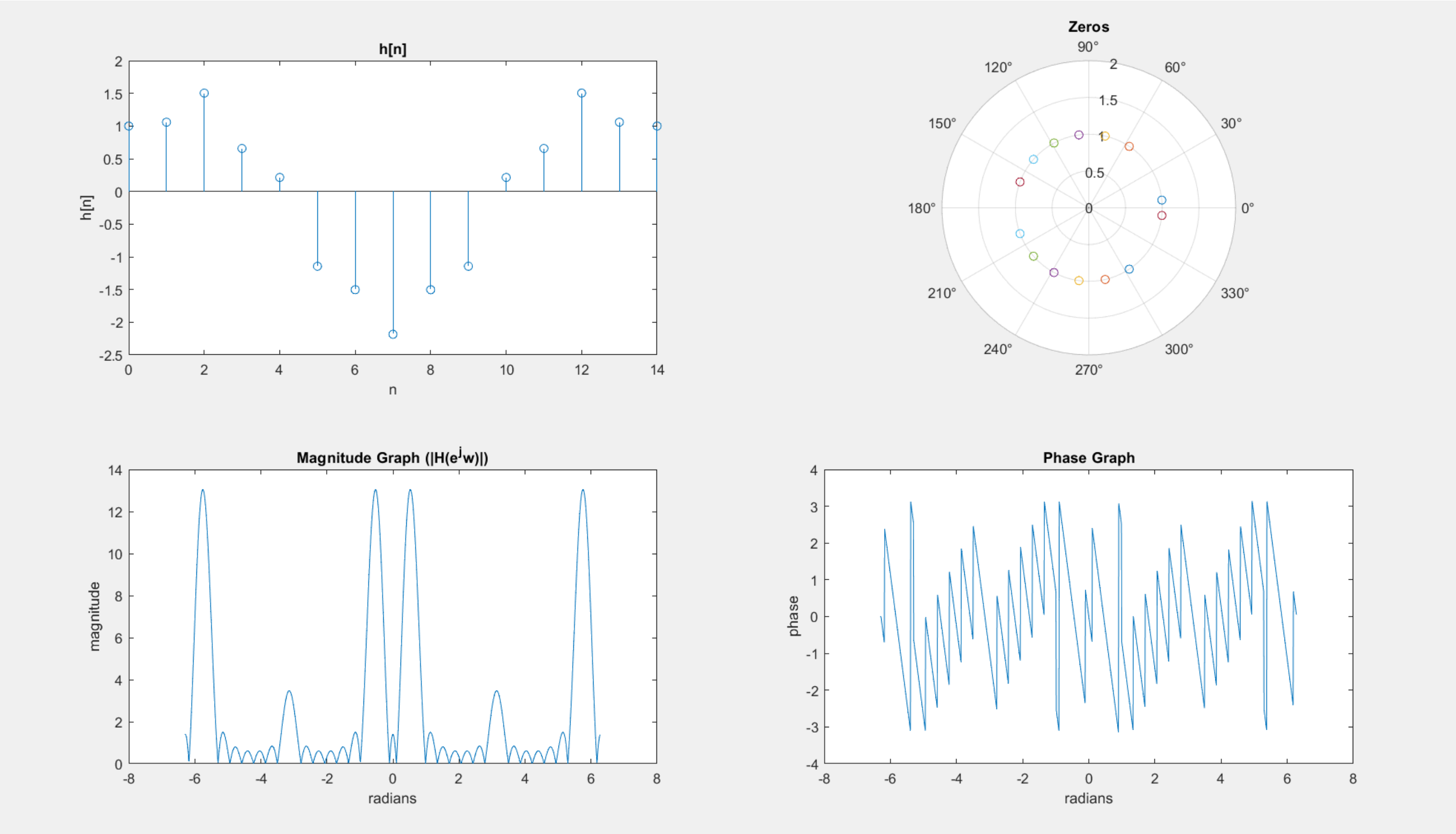

Figure 1: Graphs for Q1

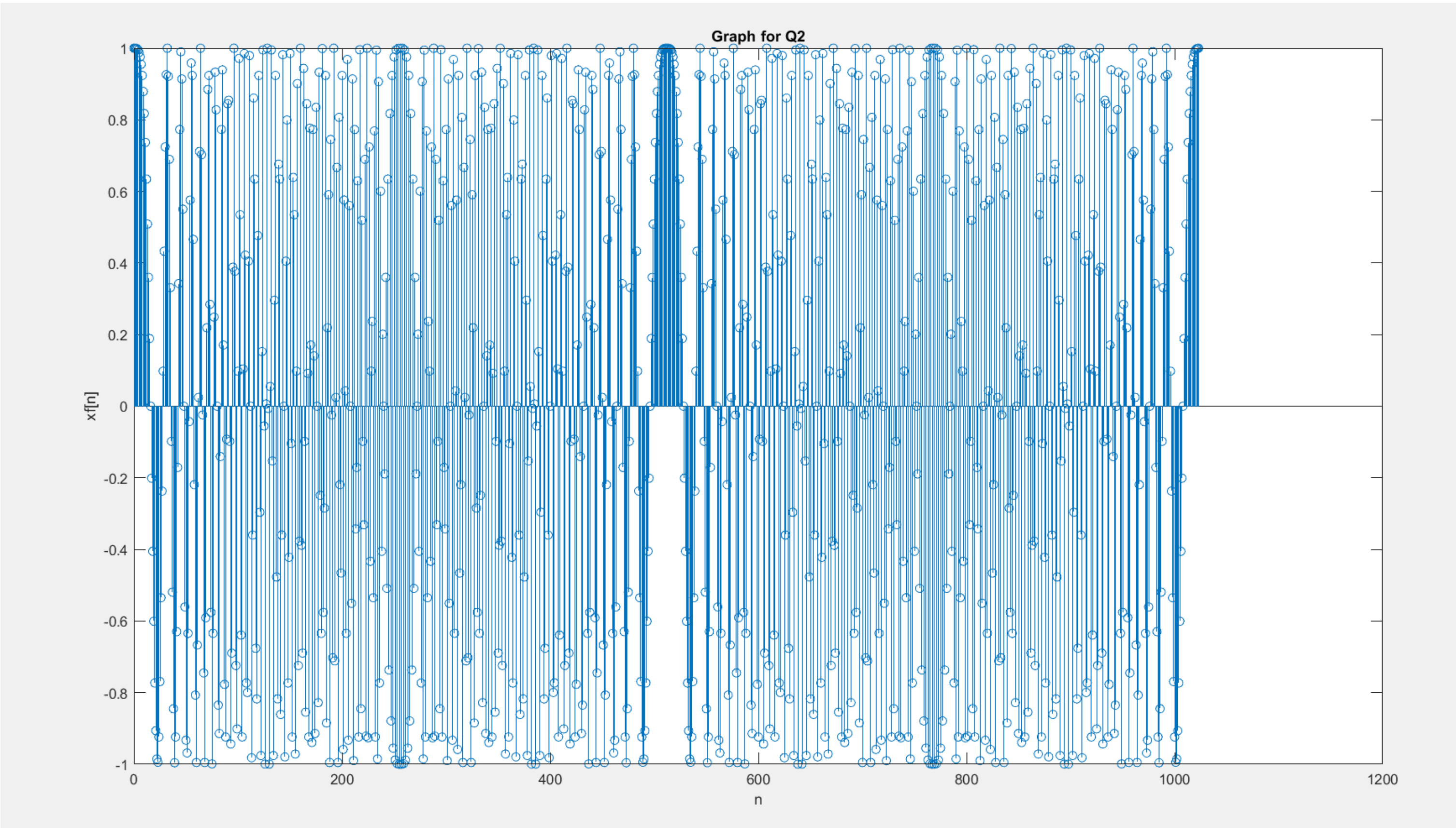

Figure 2: Graph for Q2

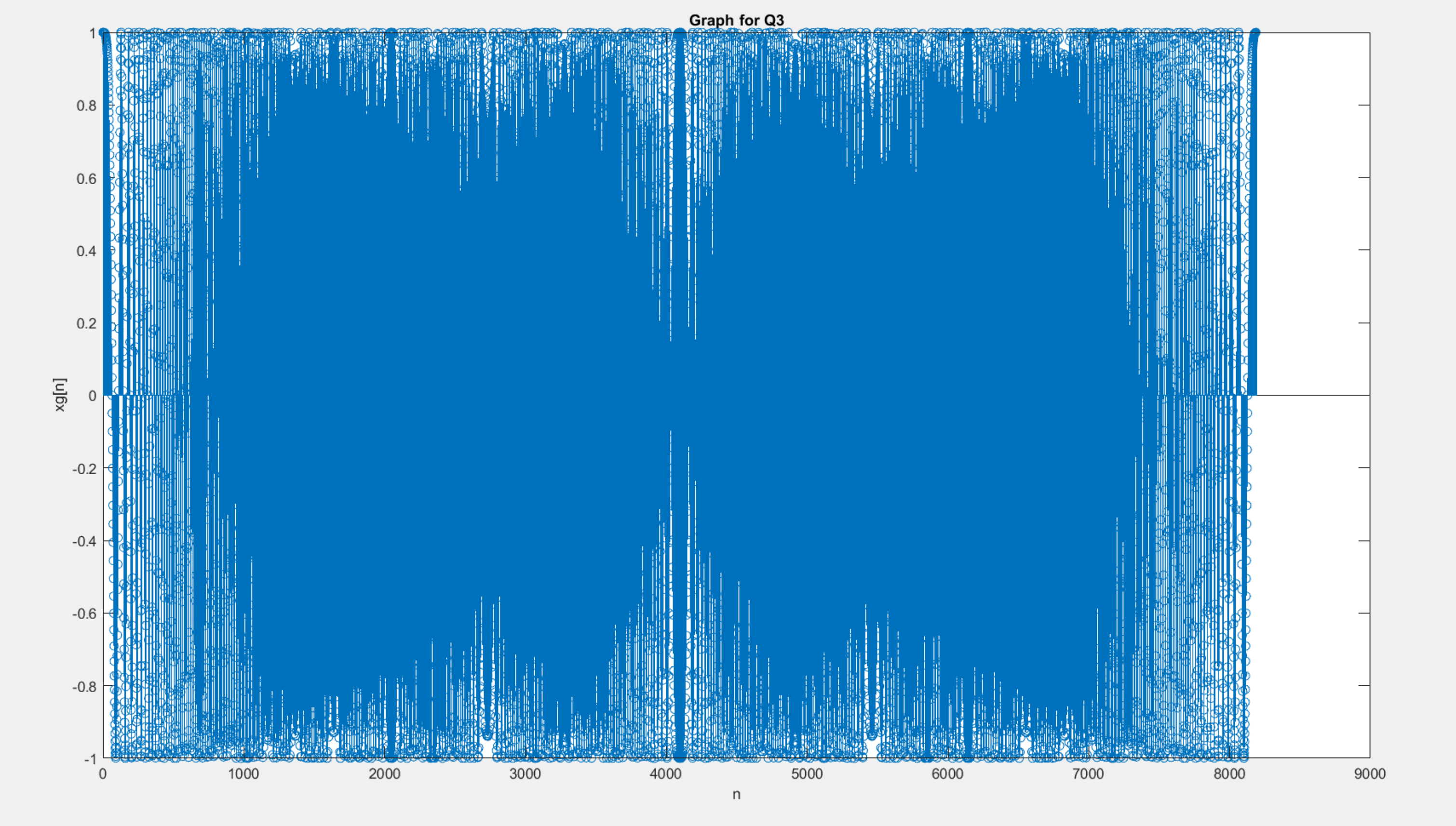

Figure 3: Graph for Q3

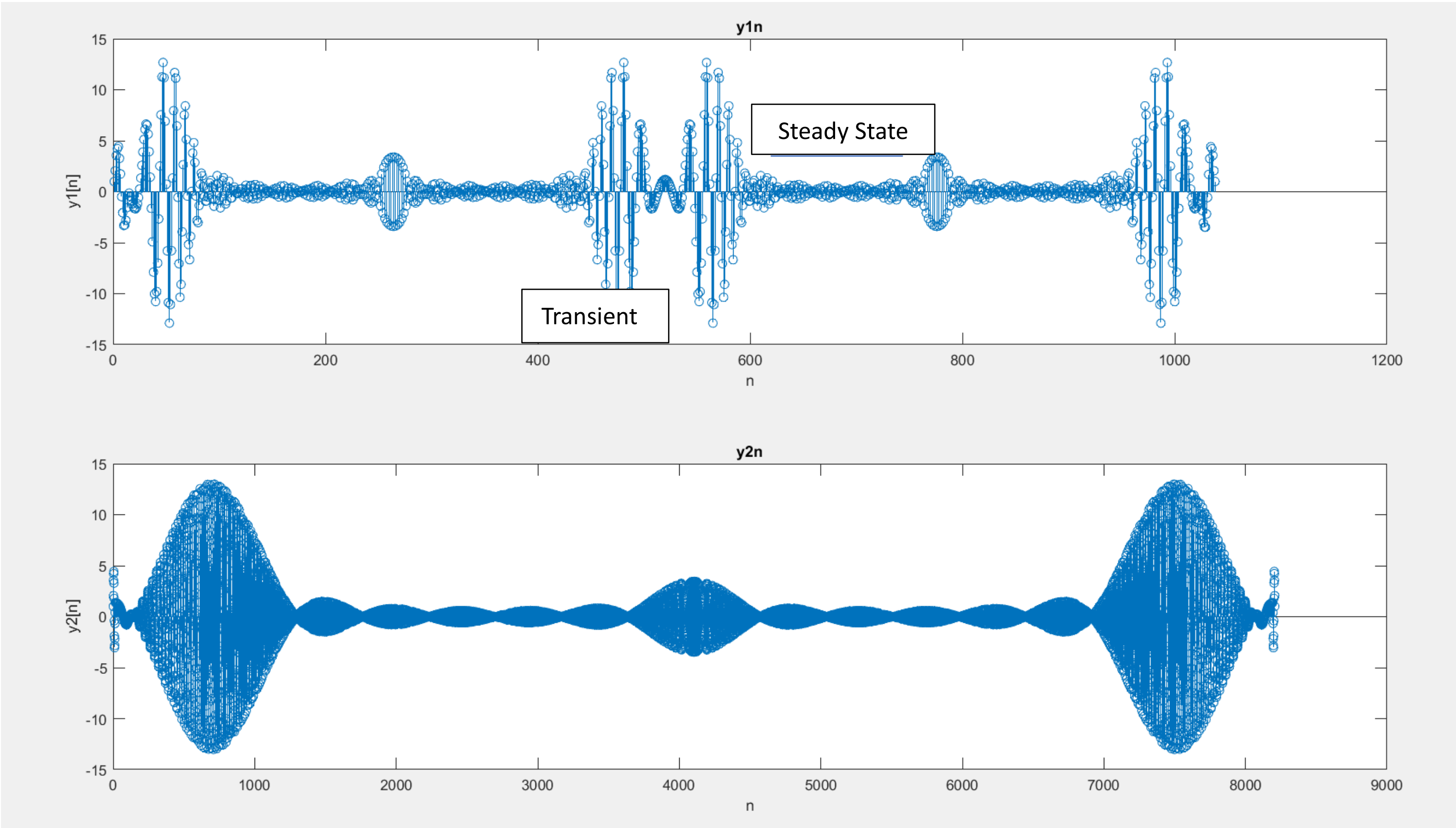

Figure 4: Graphs for Q4

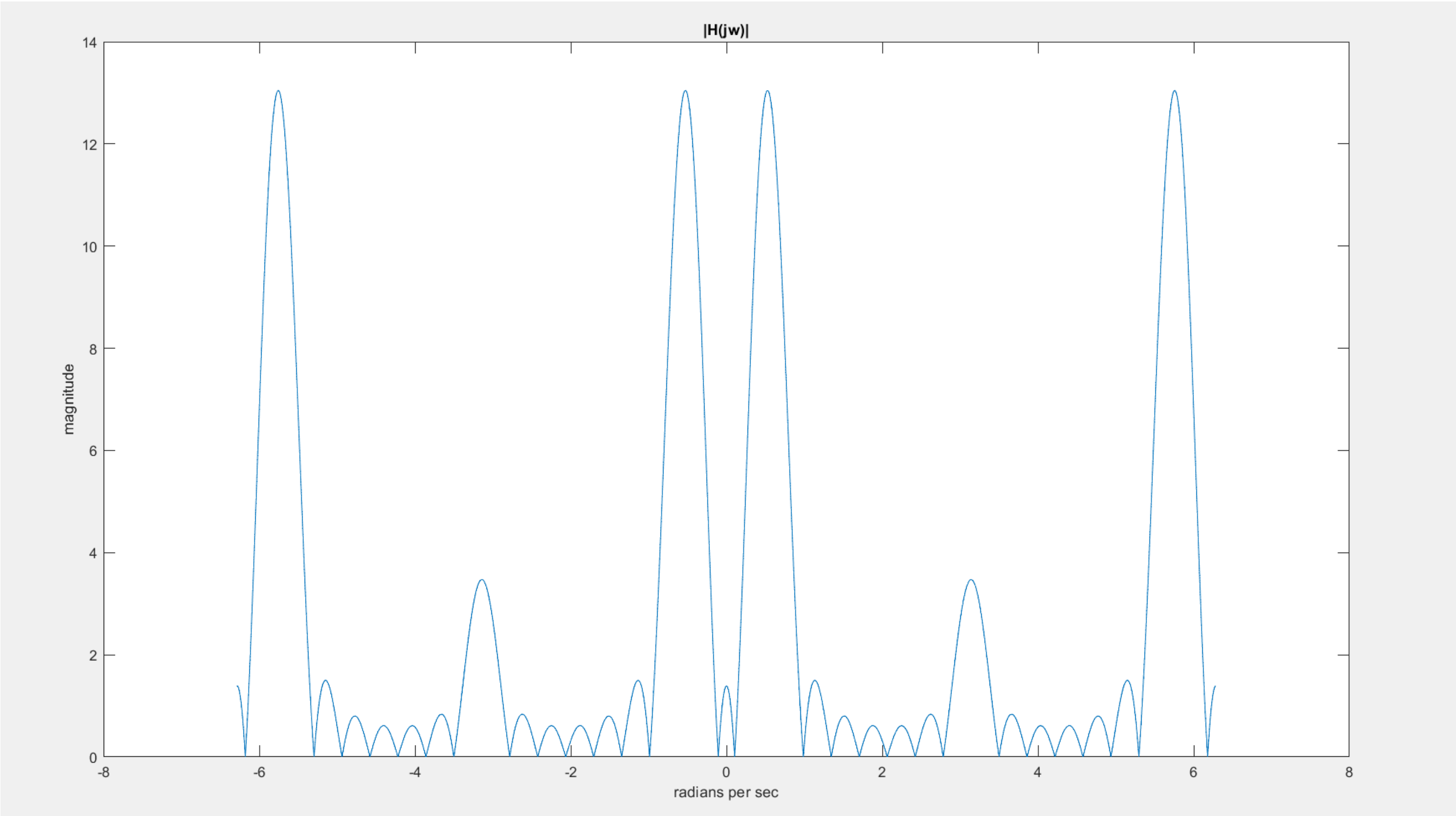

Figure 5: Magnitude Graph for Q4

Impulse Response Array

Columns 1 through 11

1.0000 1.0589 1.5041 0.6571 0.2139 -1.1443 -1.5027 -2.1848 -1.5027 -1.1443 0.2139

Columns 12 through 15

0.6571 1.5041 1.0589 1.0000

Written Explanation for Solutions

In this lab, the students designed FIR filters regarding certain specifications as below.

The filter specifications are as follows:

* Bandpass filter

* Real valued $h[n]$.

* Length of the filter is $L = 11 + N_1$.

* Cutoff frequencies are $\min\{\frac{\pi}{M_1}, \frac{\pi}{M_2}\}$ and $\max\{\frac{\pi}{M_1}, \frac{\pi}{M_2}\}$ radians.

* A stopband and a passband, which are as flat as possible, are desirable.

* Causal; $h[n] = 0$ for $n < 0$. Also, $h[n] = 0$ for $n \geq L$.

Figure 6: Manual Specifications

I found my values to be N1=3, M1=5, N2=8, M2=10, L=14; and found my cutoff frequencies as π/10 and π/5. Due to the conjugate symmetry of the frequency response due to z-transform, my passbands are (π/10, π/5) and (-π/10, -π/5). Furthermore, the length implies that my filter has 14 zeros distributed symmetrically to the x axis. The evaluation of zeros using symmetry helps in finding the impulse response of the filter, since $\prod_{k=0}^{14} \frac{z-b_k}{z^{14}} = \sum_{k=0}^{14} h[k]z^{-k}$. The found coefficients help in obtaining the impulse response as in Figure 1, which is purely real. The initially found zeros, the impulse response and the magnitude with the phase of the frequency response are as in Figure 1.

For Q2, taking α as 1 revealed that we need to evaluate = cos($(t+T)^2$) for checking periodicity, and we found that t^2 needs to equal t^2 + 2tT + T^2 for the signal to be periodic. Observing that 2tT + T^2 can be written as 2πk, we finalize that T = -t ±$\sqrt{t^2 - 2\pi k}$, and the derivative of t^2 with respect to time (instantaneous frequency) is 2t. This implies the signal cannot be finalized to be periodic or not, since T is a continuous function. Sampling this signal with period $\sqrt{\frac{\pi}{512}}$, we find that x1[n] is cos[$\frac{n^2\pi}{512}$] and it is required for cos[$\frac{n^2\pi}{512}$ +2πk] to be equal to cos[$\frac{(n+N)^2\pi}{512}$]. We can finalize here that n=(1024k/2N)-N/2, and n=k-256. This implies that there exists an integer k that satisfies this equation, and the signal is periodic. This periodic signal is plotted in Figure 2. Then for Q3, we found that changing alpha to 1000 rds^−2 reveal cos(1000* $\frac{n^2\pi}{1000*8192}$) = cos($\frac{n^2\pi}{8192}$), as in Figure 3.

Regarding Q4, convolution was used between xf[n] and the previously found coefficients of the z-transform's output impulse response, revealing the plotted signal in Figure 4. The transient part is demonstrated as the time intervals where the signal approaches a constant value, and the steady state displays the time intervals when the signal is oscillating near this constant. The output is the response to frequency placed into the filter. For this reason, cos(αt^2), with frequency 2αt demonstrates which time point the signal is evaluated, as the instantaneous frequency would be the itself with α being 0.5. The chirp signal thereby helps in representing the frequency response when applied as the input. Furthermore, it can be stated that the resolution can be found by the sampling rate of the chirp signal, and value with high magnitude is necessary to properly observe the frequency response of a filter. It can also be stated that the chirp signal is not stable, and convolution with a chirp signal could diverge.

Applying xg[n] as the input reveals the output y2[n] in Figure 4. For Q5, it can be stated that sampling with periodic approach forms the periodic output xr(t), even though xα(t) is not periodic. This implies that there is a change of output, one periodic one aperiodic. The chirp signal has the behavior of increasing frequency, while the output has periodically changing frequency, implying that the input reaches higher notes while the output oscillates.

It can be dictated that yr(t) and y2[n] are both periodic, one being discrete and one being continuous. yr(t) is formed by sampling its discrete signal. Every time point in the discrete signal can be sampled by lasting them for sample rate time amounts, finally creating a continuous signal. Another name for this process is interpolation.

The system's cut off frequencies are constant for every signal, and they are symmetric to the y-axis in its plot as in Figure 5. The system impulse response's cut off frequencies are as in the filter. This filter suppresses the frequencies in the stopband and allow the frequencies in the bandpass region. This implies only certain frequencies within a signal passes the filter, just as in smartphone microphones that suppress air/wind noise frequencies. We can conclude that such filters change sound signals in desired ways, making them clearer to the human ear, removing certain undesired parts of the input signal.

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HANDWRITTEN OBSERVATIONS/SOLUTIONS

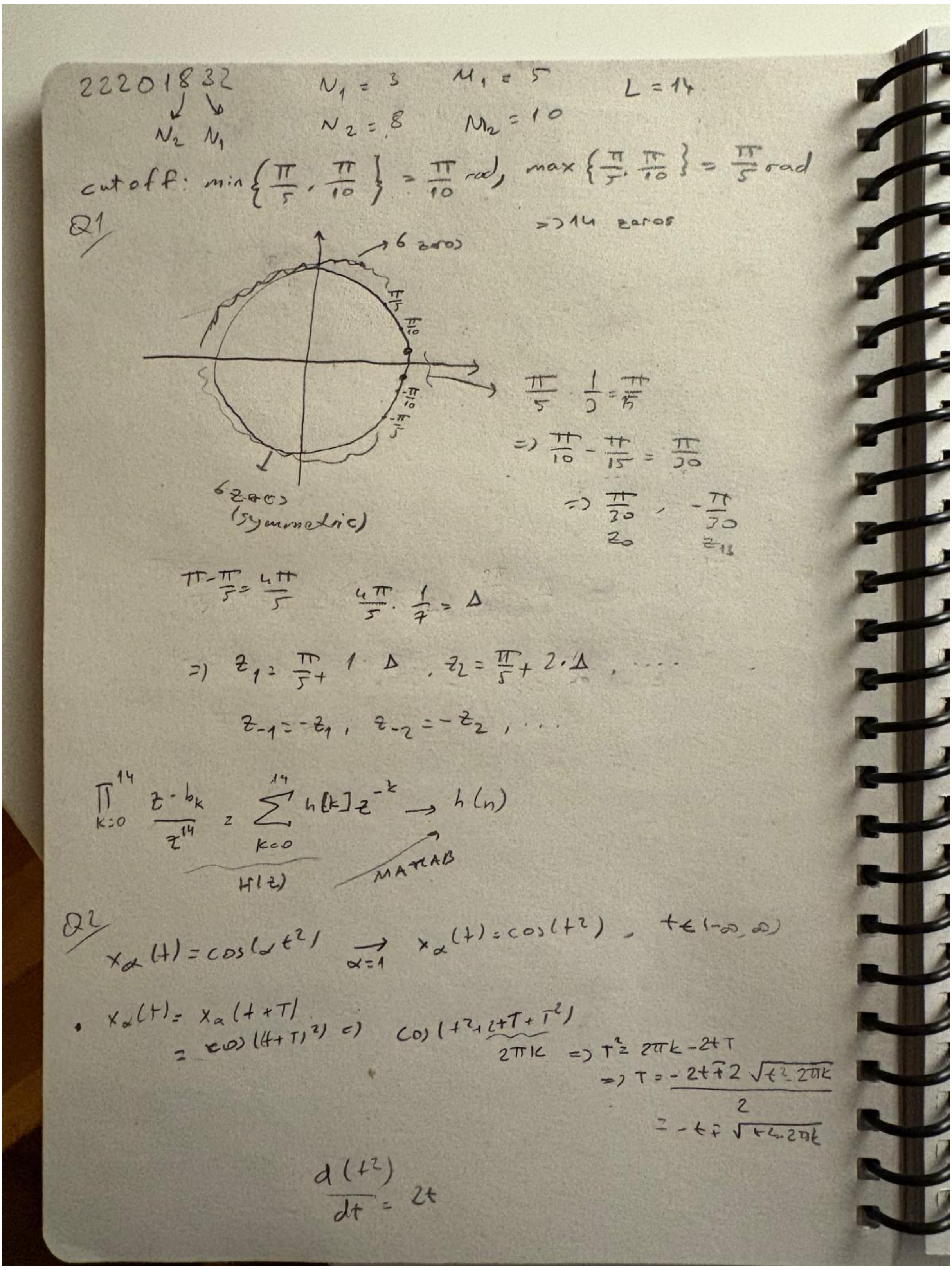

Figure 7: Handwritten Solutions Page 1

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$T_s = \sqrt{\frac{\pi}{512}}$, $x_1[n] = \cos\left[\frac{n^2\pi}{512}\right]$ $(\alpha = 1)$

$\Rightarrow \cos\left[\frac{n^2\pi}{512} + 2\pi k\right] = \cos\left[\frac{(n+N)^2\pi}{512}\right]$ $(k \in \mathbb{Z})$

$$n^2\pi + 1024\pi k = n^2\pi + N^2\pi + 2nN\pi$$

$$2nN + N^2 = 1024k$$

$$n = \frac{1024k}{2N} - \frac{N}{2}$$

$$\Rightarrow N = 512, \quad n = k - 256$$

There exists an integer k that satisfies $n = k - 256$.

$\Rightarrow N = 512$, periodic.

Q2)

$x_g[n] = \cos\left(1000 \cdot \frac{\pi n^2}{\underbrace{1000 \cdot 8192}_{T_s^2}}\right)$ $\rightarrow x_g[n] = \cos(\alpha \cdot (nT_s)^2)$

$= \cos\left(\frac{n^2\pi}{8192}\right)$

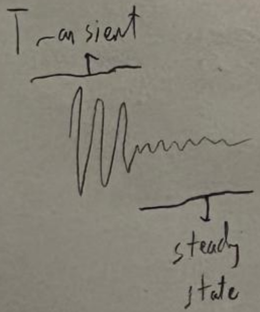

Figure 8: Handwritten Solutions Page 2

Appendix

MATLAB CODE

File 1

```
% n1 = 3;
% n2 = 8;
% m1 = 5;
% m2 = 10;
% L = 14;

clc
clear
close all
% 22201832 Emir A. Bayer EEE321 Lab5p1
syms z

%% Q1
delta = (4 * pi/5) / 7;

z0 = pi / 30;
z1 = pi / 5 + 1 * delta;
z2 = pi / 5 + 2 * delta;
z3 = pi / 5 + 3 * delta;
z4 = pi / 5 + 4 * delta;
z5 = pi / 5 + 5 * delta;
z6 = pi / 5 + 6 * delta;
z7 = -z1;
z8 = -z2;
z9 = -z3;
z10 = -z4;
z11 = -z5;
z12 = -z6;
z13 = -z0;

zeros_list = [z0, z1, z2, z3, z4, z5, z6, z7, z8, z9, z10, z11, z12, z13];

fres = (z-exp(1i*zeros_list(1))) * (z-exp(1i*zeros_list(2))) * (z-exp(1i*zeros_list(3))) *...
+(z-exp(1i*zeros_list(4))) * (z-exp(1i*zeros_list(5))) * (z-exp(1i*zeros_list(6))) * (z-exp(1i*zeros_list(7))) *...
+(z-exp(1i*zeros_list(8))) * (z-exp(1i*zeros_list(9))) * (z-exp(1i*zeros_list(10))) * (z-exp(1i*zeros_list(11))) *...
+(z-exp(1i*zeros_list(12))) * (z-exp(1i*zeros_list(13))) * (z-exp(1i*zeros_list(14)));

deg = -2*pi : 0.01 : 2*pi;
q1(z) = fres/(z^14);
```

```
figure(1);
subplot(2, 2, 1);
ires = round(double(coeffs(fres)), 8);
disp(ires);
stem(0:14, ires);
title('h[n]');
xlabel('n');
ylabel('h[n]');

subplot(2, 2, 2);
polarscatter(zeros_list, 1);
title('Zeros');

subplot(2, 2, 3);
plot(deg, abs(q1(exp(1j.*deg))));
xlabel('radians');
ylabel('magnitude');
title('Magnitude Graph (|H(e^jw)|)');

subplot(2, 2, 4);
plot(deg, angle(q1(exp(1j.*deg))));
xlabel('radians');
ylabel('phase');
title('Phase Graph');



%% Q2

n2range = 0:1023;
xfn = cos(pi * (n2range.^2/512));
figure(2);
stem(n2range, xfn);
xlabel('n');
ylabel('xf[n]');
title('Graph for Q2');
save('xfn.mat','xfn')

```

```
%% Q3

n3range = 0:8192;
xgn = cos(pi * (n3range.^2/8192));
figure(3);
stem(n3range, xgn);
xlabel('n');
ylabel('xg[n]');
title('Graph for Q3');
save('xgn.mat','xgn');


%% Q4

y1n = conv(ires, xfn);
save('y1n.mat', 'y1n');
figure(4);
subplot(2, 1, 1);
%1036
stem(1:1038, y1n);
xlabel('n');
ylabel('y1[n]');
title('y1n');

y2n = conv(ires, xgn);
save('y2n.mat', 'y2n');
subplot(2, 1, 2);
%8204
stem(1:8207, y2n);
xlabel('n');
ylabel('y2[n]');
title('y2n');

ts = sqrt(pi/(1000*8207));
rangeq7 = -2 * pi : ts : 2 * pi;
figure(5);
plot(rangeq7, abs(q1(exp(1j.*rangeq7))));
xlabel('radians per sec');
ylabel('magnitude');
title('|H(jw)|');

save('ires.mat', 'ires')
```

File 2

```
clc
clear
close all
% 22201832 Emir A. Bayer EEE321 Lab5p2

load('y2n.mat', 'y2n');

ts = sqrt(pi/(1000*8207));
rate = 1/ts;
player = audioplayer(y2n, rate);
T = ts .* length(y2n);

continuee = 1;
while continuee
    play(player);
    pause(T);
    stop(player);
end
```

File 3

```
clc
clear
close all
% 22201832 Emir A. Bayer EEE321 Lab5p3

load('ires.mat', 'ires');


[internstellar, samplerate1] = audioread('internstellar30sec.m4a');
out1 = conv(ires, internstellar(:,1));
audiowrite('internstellarout.m4a', out1, samplerate1);

[me, samplerate2] = audioread('lab5recording.mp3');
out2 = conv(ires, me);
audiowrite('meout.m4a', out2, samplerate2);

audiowrite('interns.m4a', internstellar(:,1), samplerate1);
```