

DSP and Communications

**Compile a report detailing signal processing analysis
using Matlab software.**

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Abstract

The aim of this report is to demonstrate a detailed knowledge on the concept of digital signal processing and communications systems. It is also to understand the use of MATLAB simulation software. With computational tools such as MATLAB, this has become an integral part of routine analytic practices.

The report has been split into four separate tasks outlined in the table of contents.

The report is to be at least 2,500 words and should include a high level of information regarding the subject matter. It also includes diagrams, drawings, a good level of referencing material.

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Introduction

I have been tasked with carrying out some research into the use of digital signal processing and communications systems used regularly within the engineering and science industry. It hosts the ability to solve equations immediately using mathematics. The start of the report will focus on research into the different types of communication systems and increasing capacity on the mobile network.

The report will then move on to the use of the MATLAB software and how we use this software to analyse waveforms and remove any unwanted noise. Finally, using MATLAB, I will provide evidence on how to use cut off frequency filters to improve signal quality.

The report will finish with supported a short conclusion followed by references.

Task 1

With a growing population and technology constantly improving, the need for an increased network capacity has never been more vital. According to an Ofcom discussion paper published February (9th) 2022, it stated ‘In recent years we have seen an average 40% year-on-year growth in demand for mobile services provided over public mobile networks.’

[www.ofcom.org.uk]

Mobile application use is far more frequent as everything can now be done online, there’s a big drive to go paperless and many meetings now can be done over the phone as opposed to previously face to face. Releasing further spectrum, improving spectral efficiency through improvement of technology and network densification are suggestions to improve network capacity. [www.gsma.com]

One way in which capacity can be increased is using small cells or macro sites. These can be used to expand a current network by adding extra sites. Many networks in the UK rely mainly on macro sites. It’s cost effective and it is much easier to implement. However, with the increasing population and lack of land, new sites can prevent a stumbling block in more heavily populated areas.

Another way to increase capacity would be increasing the number of macro sites. Coverage and capacity are increased, especially in rural areas, as they are spaced further apart than small cells. However, building new sites is a large expenditure to the network companies. It would also take time due to the legal constraints of building these new sites. This is not as effective in densely populated areas where spectrum and small cell reuse is more efficient.

Another way of increasing the network capacity is by using the mmWave (high frequency millimeter-wave) which deploys small cells to support high data rates. This is very effective in areas that require high demand in scalable areas. The high frequency spectrum uses high bandwidth applications and increase the capacity in localised regions. Unfortunately, this method has poor signal distribution so built-up areas would be an issue. The technology also only caters to short ranges. It can be expensive to install and maintain.

Another way capacity could be increased is by having dual/multi network providers rather than being limited to one provider’s network. This would free up much more space on the

network, but providers are financially driven. Whilst sharing network coverage would improve performance, revenue is the main priority for these private companies.

[www.ofcom.org.uk]

Finally, using Wi-Fi and other similar capabilities to reduce the data use from mobile networks, is another way to increase network capacity. This again would free up more availability on the network. BT having been researching indoor femtocell solutions, this will ‘allow access to a mobile network through a standard internet connection’.

[www.btwholesale.com]

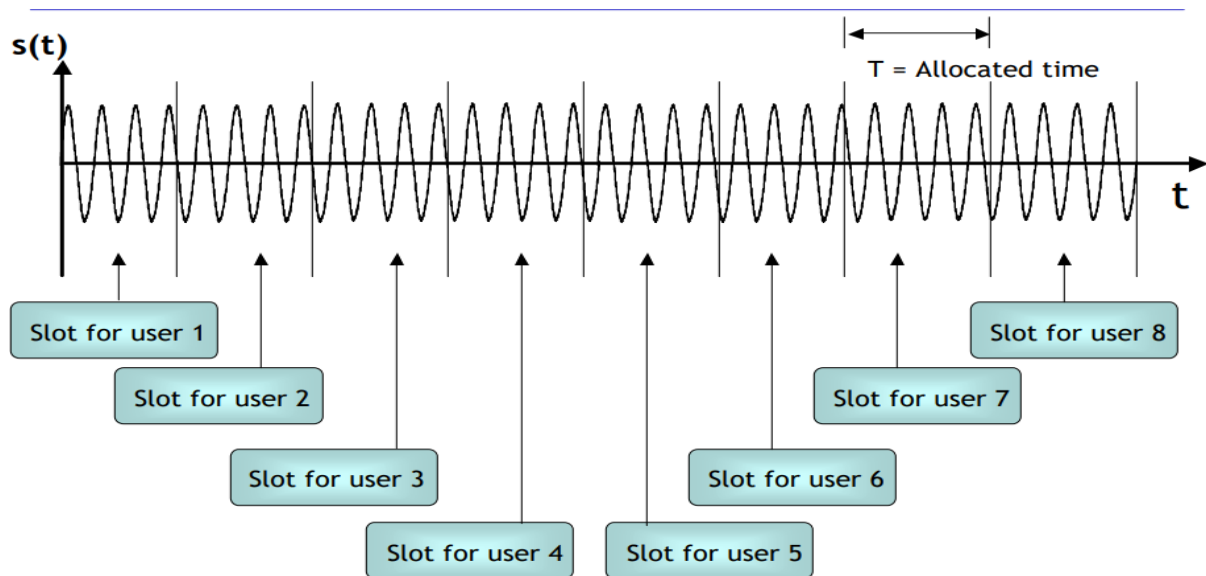
TDMA in GSM

Time division multiple access (TDMA) allows multiple users access to a single radio-frequency channel without interference by using digital transmission technology. This is done by allocating specific time slots on the channel to each user. A channel will be divided into six separate time slots and each signal will use two slots. Each caller will be assigned time slot when transmission occurs. Time division multiple access provides a 3 to 1 capacity gain over AMPS (advanced mobile-phone service).

[cdn.preterhuman.net]

In Global System for Mobile Communications (GSM), 8 slots are condensed into one frame. Continuously, this allowed multiple slots to be transmitted whilst providing data communication and high-capacity voice on one solo frequency channel. The image below shows how time division multiple access works within this. It uses 4 different frequencies 850MHz, 900MHz, 1800MHz and 1900MHz.

TDMA principle



[Image taken from web.cs.ucla.edu]

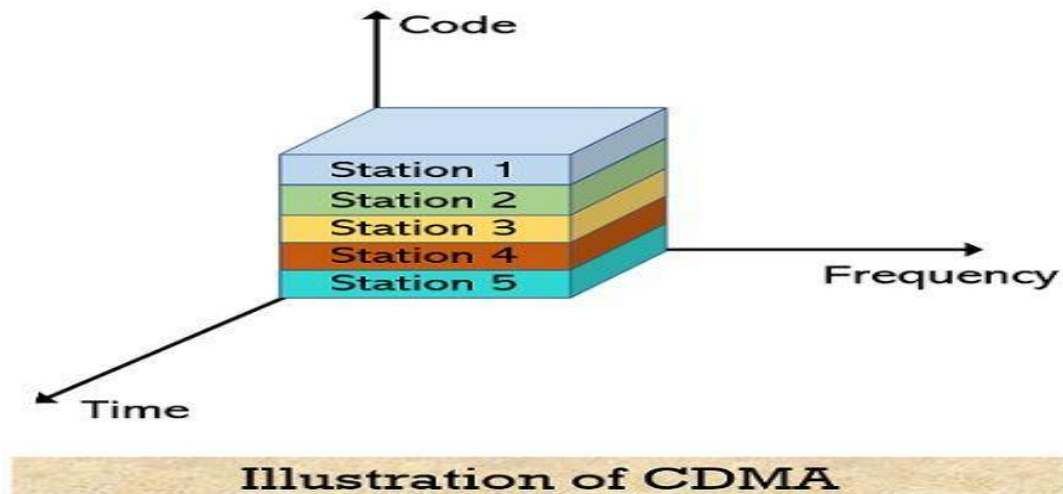
By allocating individual slots for each user that are not overlapping, this removes the potential for any interference. Interference can be caused by different signals interfering with each other. As a result, the signal can become distorted due to the interference. The end resultant communication reliability will decline. This allows the lack of guard banding needed between adjacent channels. However, guard intervals are added between slots to stop transmissions overlapping and reduce the interference. These guard intervals are typically between 30-50 microseconds. Due to the burst of the transmitted data, synchronisation is required. Another disadvantage of TDMA is it only offers moderate system flexibility as well as medium data rates. In TDMA, spectrum efficiency is improved when compared to analogue systems. Since the transmission is non-continuous, transmitters can be switched off when they are not needed. This allows the ability to reduce power consumption and save battery life on the application. Finally, by using different time slots and allowing users to share the same frequency bands, more people can use the applications simultaneously.

(Awati 2021)

WCDMA in UMTS

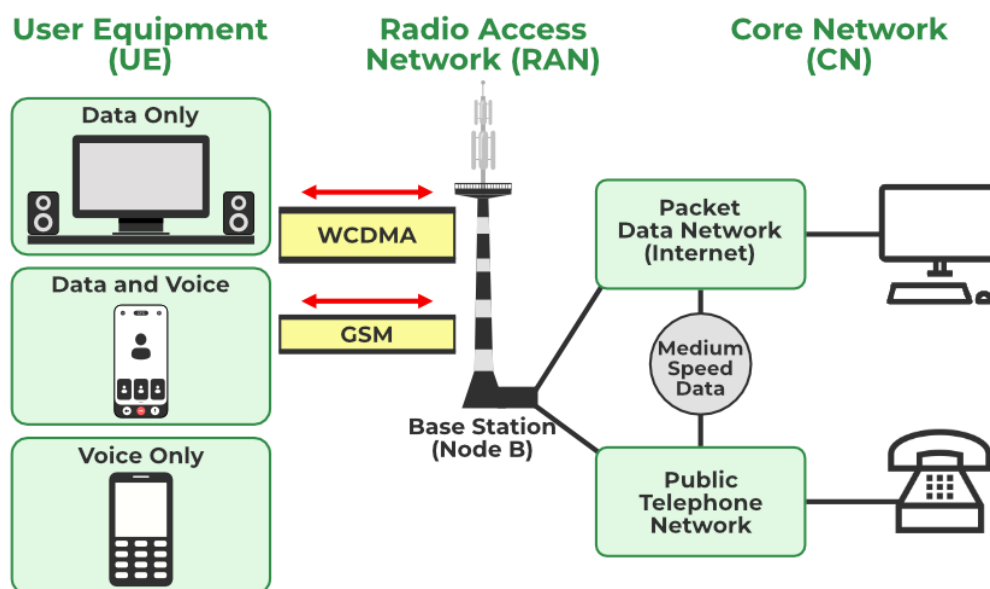
Wideband Code Division Multiple Access (WCDMA) originates from Code Division Multiple Access (CDMA). CDMA is a process which allows multiple users to transmit signals at the same time over one common channel. Each individual user's data has its own

unique code to allow for the signals at the receiver to be separated. Within the same bandwidth, this allows more users simultaneously. The image below shows how code division multiple access works within this.



(Y, 2021)

However, WCDMA provides higher data rates than CDMA by using a wider frequency band. Below is an image shows a standard setup of how WCDMA functions compared to GSM.



(Khandelwal 2023)

WCDMA network operate through Asia and Europe, which is also known as the Universal Mobile Telecommunications System (UMTS). It uses a frequency of around 2100MHz and is

one of the main technologies for third generation (3G). Due to the dominant 3G technology, it was adopted in North America offering better benefits than the old second generation (2G). Here it uses 1900MHz and 850MHz. This is because it offered better coverage, improved capacity as well as the higher data rates. However, it was overtaken by fourth generation (4G). It is still used in many areas where there is a lack of 4G networks, or the technology is not available.

One of its biggest advantages is its capability to handle large volumes of data. It's also very reliable, can transmit signals over long distances and has good coverage. This makes it a preferred use in applications such as watching videos or internet browsing on mobile phones.

[Ayush Khandelwal Jan (18th) 2023]

Interference is minimised by ensuring that each transmission only uses the power required. This power control technique ensures the integrity of the connection. The system also uses advanced signal processing to remove any interference or expected interference from known sources.

(Ufoaroh, 2018)

Task 2

```

% wav_DSPAssignmentsound.m% Generate and save sound files for five frequency
inputs

clear all;
close all;
clc;
newfilename = 'output_Alex.wav'; % write your chosen output file name
d=4.0; % wating time in s
[s,fs]=audioread('wav_Alex.wav'); % write your given sound file name
fs % display
t = (0:length(s)-1)/fs;
figure(1)
w=length(t);
plot(t(1000:w-1),s(1000:w-1),'r')
grid
xlabel('time-secs')
ylabel('signal value - volts')
title('A Plot of a voice')
figure (2)
n = 2000; % your choice to plot part of the voice signal
plot(t(1:n),s(1:n))
xlabel('time')
ylabel('s[n]')
title('A Plot of first 2000 samples')
figure (3)
plot(t,s,'b')
grid
xlabel('time-secs')
ylabel('signal value - volts')
title('A Plot of a Simple of voice Signal')
sound(s, fs); % Generate sound
pause(d + 0.5); % waiting for sound end
p=fft(s);
L=length(s);
p2=abs(p/L);
p1=p2(1:L/2+1);
k=fs*(0:(L/2))/L;
figure(4)
plot(k,p1)
plot(k(1:15000),p1(1:15000),'g')
grid
xlabel('frequency (Hz)')
ylabel('magnititude')
title ('FFT of the Voice')
% fir1 filter B =fir1(N, Wn, 'filter type')designs an N'th order lowpass, high
pass, band pass and band stop FIR digital filter
% and returns the filter coefficients in length N+1 vector B.Wn is cutoff
frequency
% B = fir1(N,Wn,'high') designs an N'th order highpass filter
% If Wn is a two-element vector, Wn = [W1 W2], FIR1 returns an order N bandpass
filter with passband W1 < W < W2.
% By default FIR1 uses a Hamming window
% eleminate the first frequency (f3) from the signal
LCoff1 =1250/(fs/2); % low cutoff frequency
HCoff1 =1450/(fs/2); %high cutoff frequency
ORD1=500; % order of FIR bandstop filter.
filtcoeff1= fir1(ORD1,[LCoff1, HCoff1],'stop');% returns the filter coefficients
in length ORD1+1 vector filtcoeff1

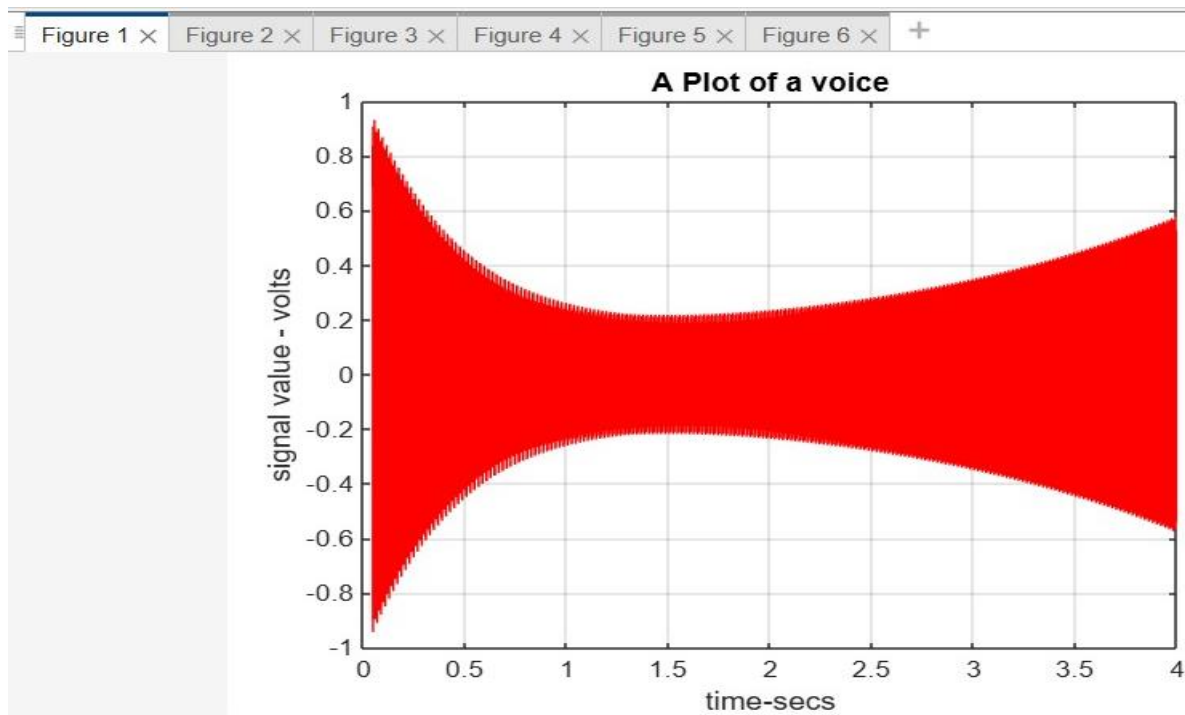
```

```

%y = filter( b, a, x);
% filters the data in vector 'x' {in this case s} with the filter described by
coefficient vectors 'a' and 'b' (filtcoeff1 vectors b, a=1 FIR filter]
% to create the filtered data vector 'y' [in this case voic1].
voic1=filter(filtcoeff1,1,s); % generate a new voice with only four frequencies
sound(voic1,fs);
pause(d + 0.5); % waiting for sound end
% eliminate the second frequency (f4) from the voice signal
LCoff2 = 2000/(fs/2); % low cutoff frequency
HCoff2 =2200/(fs/2); %high cutoff frequency
ORD2=500; % order of FIR bandstop filter.
filtcoeff2=fir1(ORD2,[LCoff2, HCoff2],'stop'); % returns the filter coefficients
in length ORD2+1 vector filtcoeff2
% filters the data in vector 'x' {in this case voic1] with the filter described by
coefficient vectors 'a' and 'b' (filtcoeff1 vectors b]
% to create the filtered data vector 'y' [in this case voic2] .
voic2=filter(filtcoeff2,1,voic1);
sound(voic2,fs); % generate a new voice with only three frequencies
figure(5)
plot(t(1000:w-1),voic2 (1000:w-1),'y')
grid
xlabel('time-secs')
ylabel('signal value - volts')
title('A Plot of the new voice')
pause(d + 0.5); % waiting for sound end
figure(6)
L1=length(voic2);
mag1=abs(fft(voic2));
kk=fs*((0:(L1-1))/L1);
plot(kk(1:15000),mag1(1:15000),'b')
grid
xlabel('frequency (Hz)')
ylabel('magnititude')
title ('FFT of the newVoice')
% Save wav file to disk
audiowrite(newfilename,voic2,fs);

```

Figure 1



Above is figure 1 which is a plot of the voice without the first 1000 samples.

Figure 2

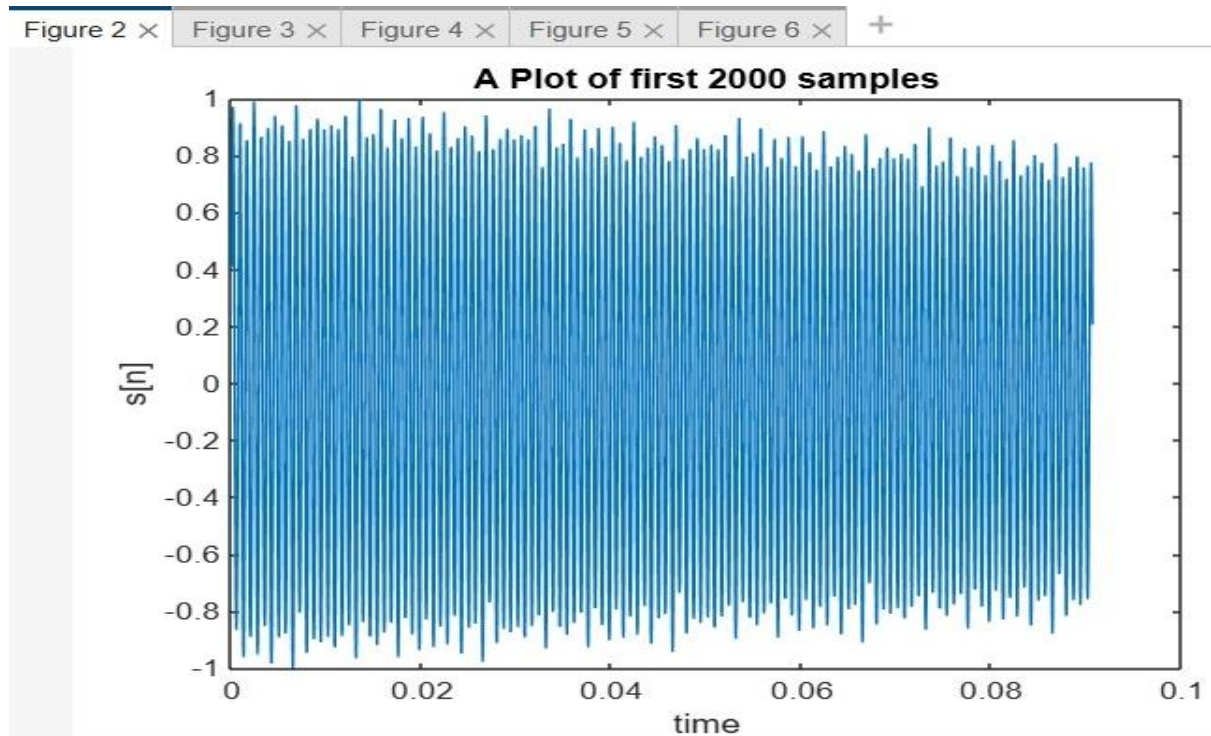


Figure 2 shows a plot of the voice where I have chosen to plot the first 2000 samples. The signal is frequently increasing and decreasing at the beginning. As the time increases, the

signal starts to fluctuate less becoming less erratic and more consistent. This signal is becoming smoother.

Figure 3

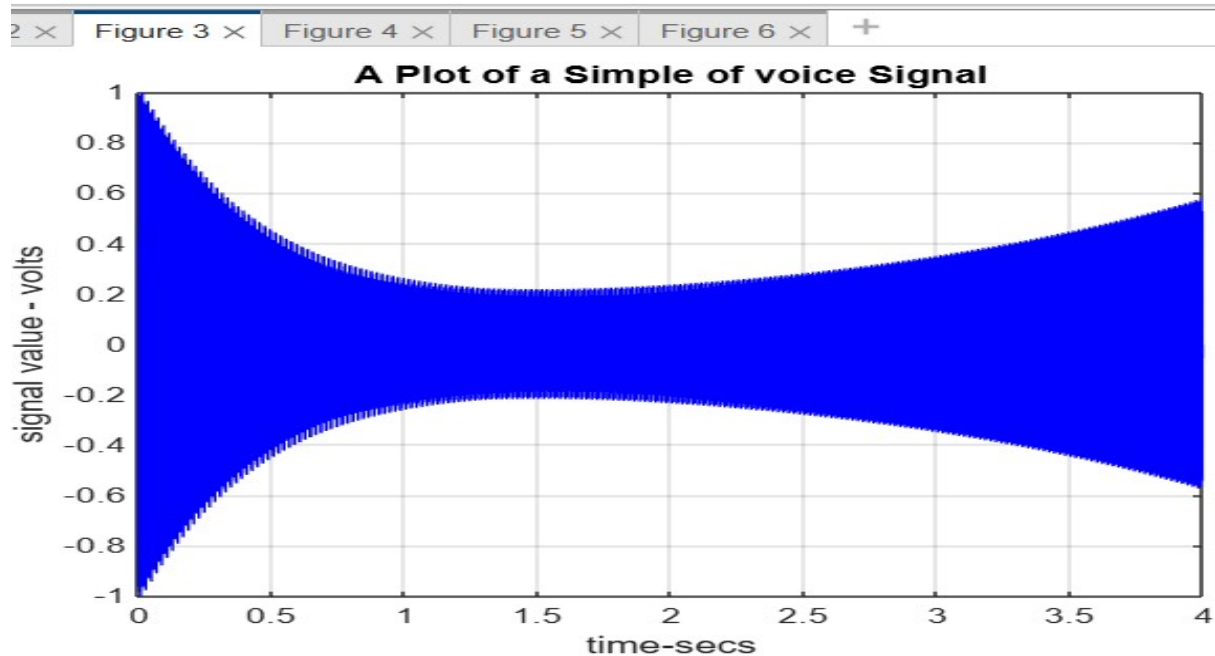


Figure 3 shows the plot of the sample of the voice, like figure 1 but it includes the start of the signal. Figures 1, 2 and 3 include part a) of the task in reading the TMPRS file provided.

Figure 4

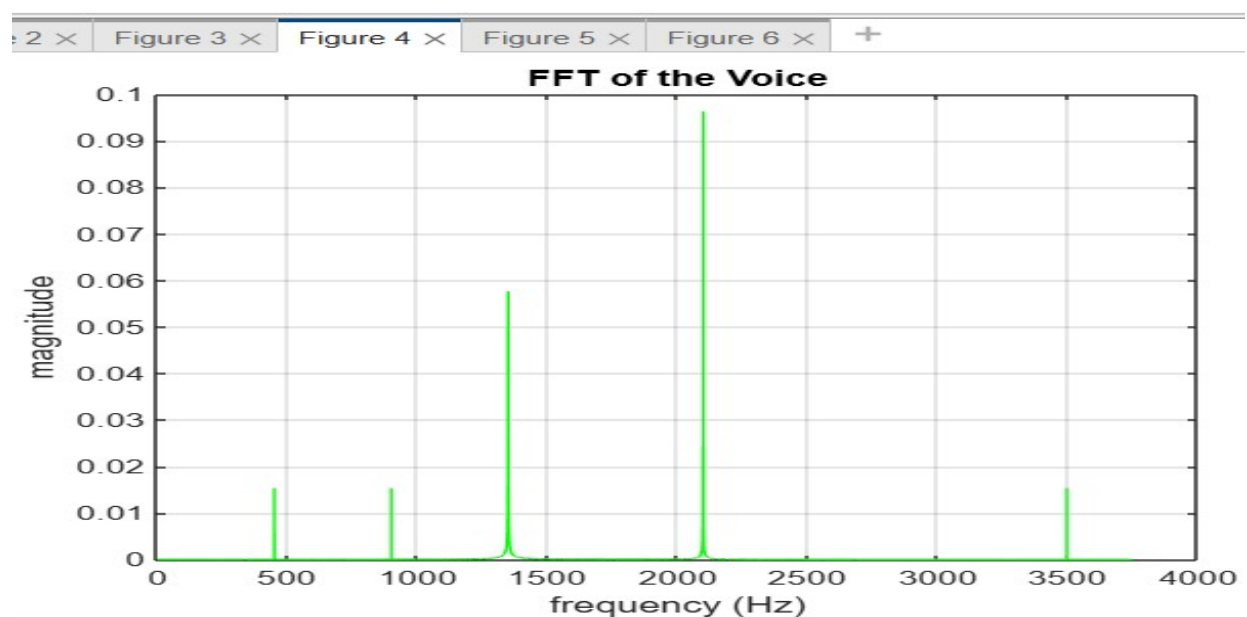


Figure 4 shows how part b) of the task is achieved by identifying the frequencies of each tone in the signal. As you can see, there are frequencies for the 5 tones.

Figure 5

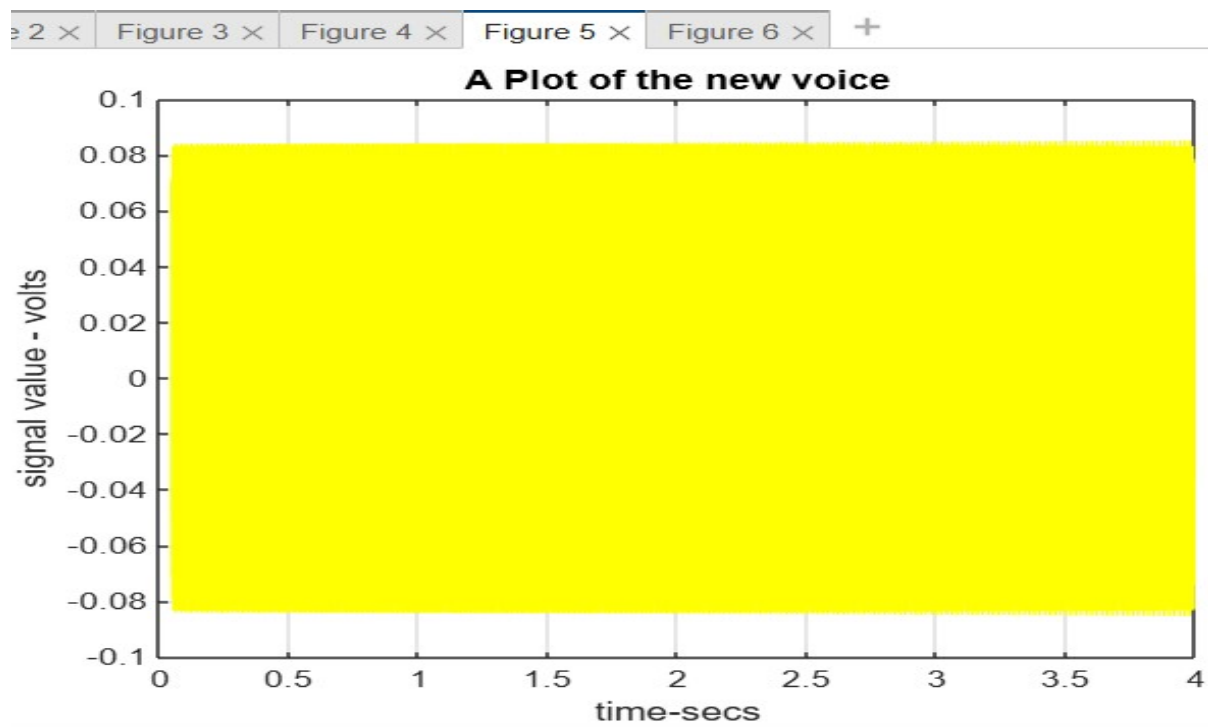


Figure 5 shows the newly generated clean version of the tone as required for part d) of the task. As you can see from the plot, there is no interference with the signal.

Figure 6

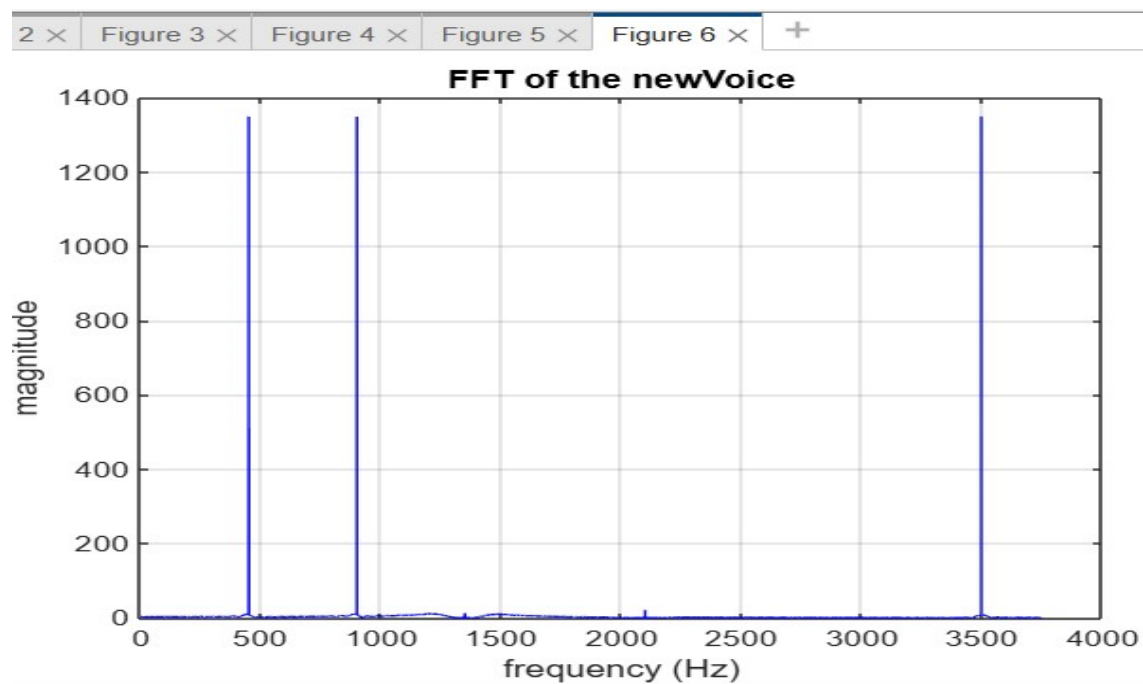
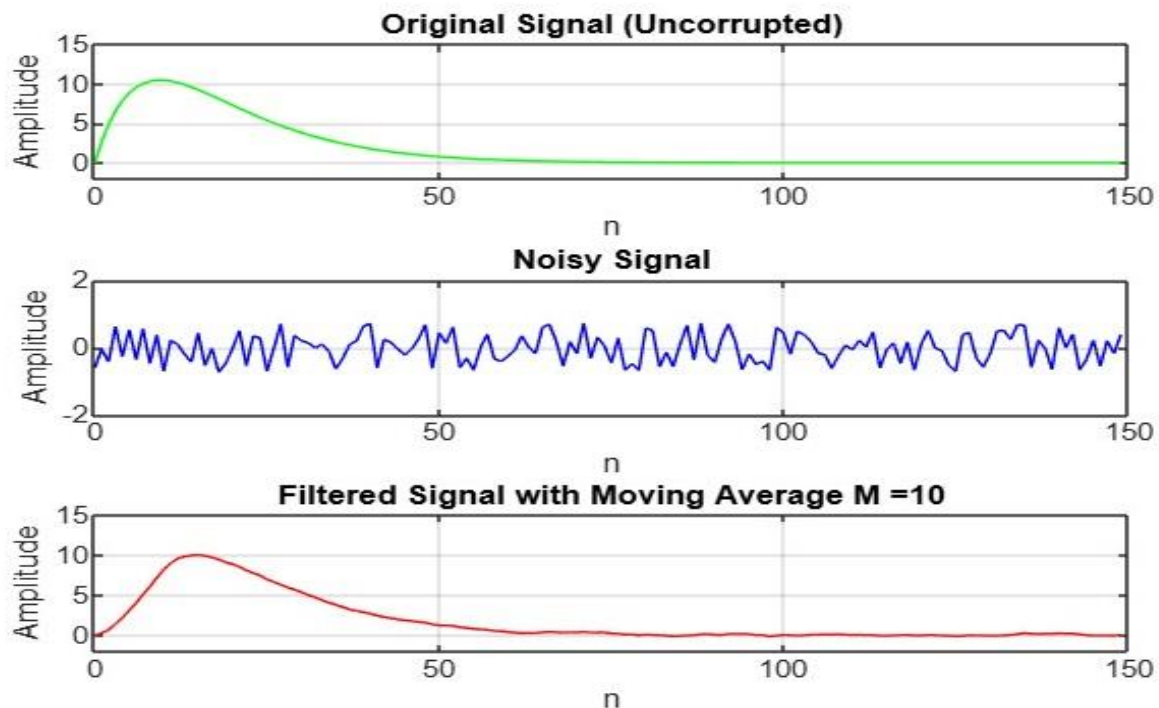


Figure 6 shows the second part of task d) which retains only the three desired frequencies.

The production of the newly created TMPRS without the corrupted tones is in the final line of the code which is now saved. This is required to complete part e) of the task.

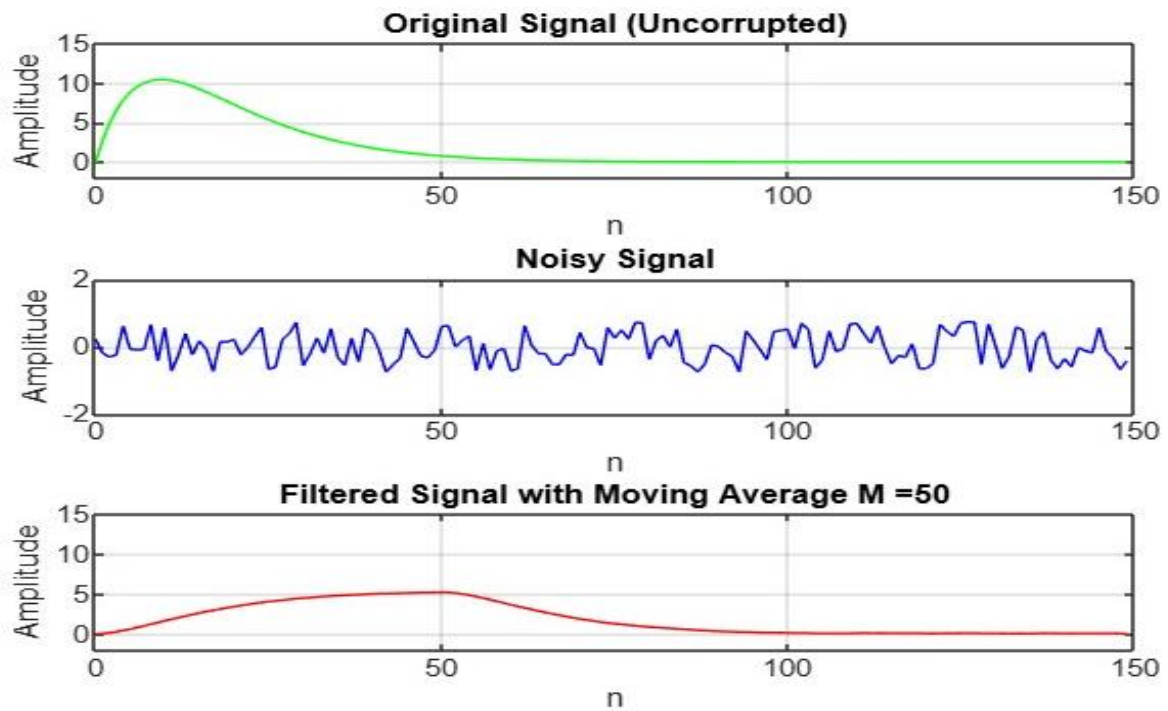
Part c) of the task requires designing a filter to remove the unpleasant sound. This part of the code below shows how this is achieved. The FIR band-stop filters are created which aims to eliminate the two unwanted frequencies. This is done by inputting high and low cut off frequencies. The new sound is generated in the final line with only the desired 3 frequencies.

```
% fir1 filter B=fir1(N, Wn, 'filter type') designs an N'th order lowpass, high pass, band pass and band stop FIR
digital filter
% and returns the filter coefficients in length N+1 vector B. Wn is cutoff frequency
% B = fir1(N,Wn,'high') designs an N'th order highpass filter
% If Wn is a two-element vector, Wn = [W1 W2], FIR1 returns an order N bandpass filter with passband W1 <
W < W2.
% By default FIR1 uses a Hamming window
% eliminate the first frequency (f3) from the signal
LCoff1=1250/(fs/2); % low cutoff frequency
HCoff1=1450/(fs/2); % high cutoff frequency
ORD1=500; % order of FIR bandstop filter.
filtcoeff1= fir1(ORD1,[LCoff1, HCoff1], 'stop'); % returns the filter coefficients in length ORD1+1 vector
filtcoeff1
% y = filter( b, a, x);
% filters the data in vector 'x' {in this case s} with the filter described by coefficient vectors 'a' and 'b'
(filtcoeff1 vectors b, a=1 FIR filter)
% to create the filtered data vector 'y' [in this case voic1].
voic1=filter(filtcoeff1,1,s); % generate a new voice with only four frequencies
sound(voic1,fs);
pause(d + 0.5); % waiting for sound end
% eliminate the second frequency (f4) from the voice signal
LCoff2 = 2000/(fs/2); % low cutoff frequency
HCoff2 =2200/(fs/2); % high cutoff frequency
ORD2=500; % order of FIR bandstop filter.
filtcoeff2=fir1(ORD2,[LCoff2, HCoff2], 'stop'); % returns the filter coefficients in length ORD2+1 vector
filtcoeff2
% filters the data in vector 'x' {in this case voic1} with the filter described by coefficient vectors 'a' and 'b'
(filtcoeff1 vectors b)
% to create the filtered data vector 'y' [in this case voic2] .
voic2=filter(filtcoeff2,1,voic1);
sound(voic2,fs); % generate a new voice with only three frequencies
```

Task 3

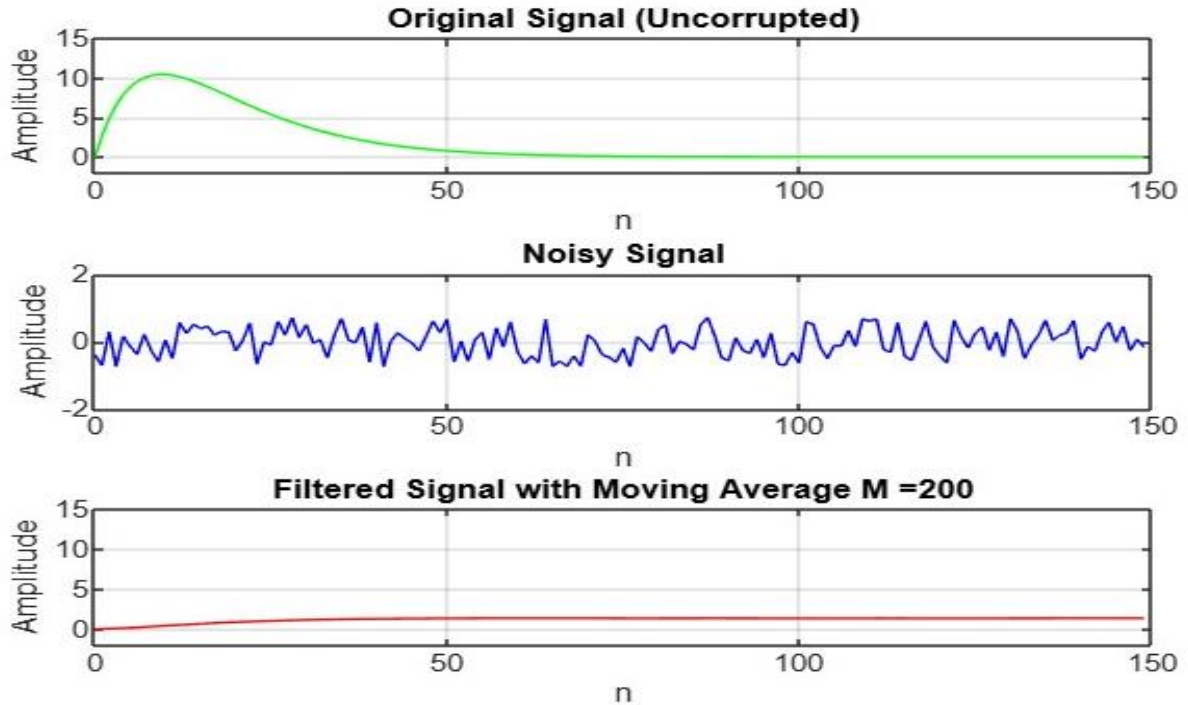
Here is the code below I have written when M=10.

```
r = 150; % Number of sampling points
n = 0:r-1; % Time vector
s = 3*n.*(0.9).^n; % Uncorrupted signal
noi = 1.5*(rand(r,1)-0.5); % Noise
x = s' + noi; % Corrupted signal
M = input('Filter length M'); % User input for filter length
num = ones(1,M); % Average coefficients
y = filter(num, 1, x)/M; % Filtered signal
subplot(3,1,1);
plot(n, s, 'g');
title('Original Signal (Uncorrupted)');
axis([0,150, -2, 15]);
xlabel('n');
ylabel('Amplitude');
grid on;
subplot(3,1,2);
plot(n, noi, 'b');
title('Noisy Signal');
axis([0,150, -2, 2]);
xlabel('n'); ylabel('Amplitude');
grid on;
subplot(3,1,3);
plot(n, y, 'r');
title('Filtered Signal with Moving Average M =10');
axis([0,150, -2, 15]);
xlabel('n'); ylabel('Amplitude');
grid on;
```



Here is the amendment to the code below I have written when M=50.

```
title('Filtered Signal with Moving Average M =50');
```



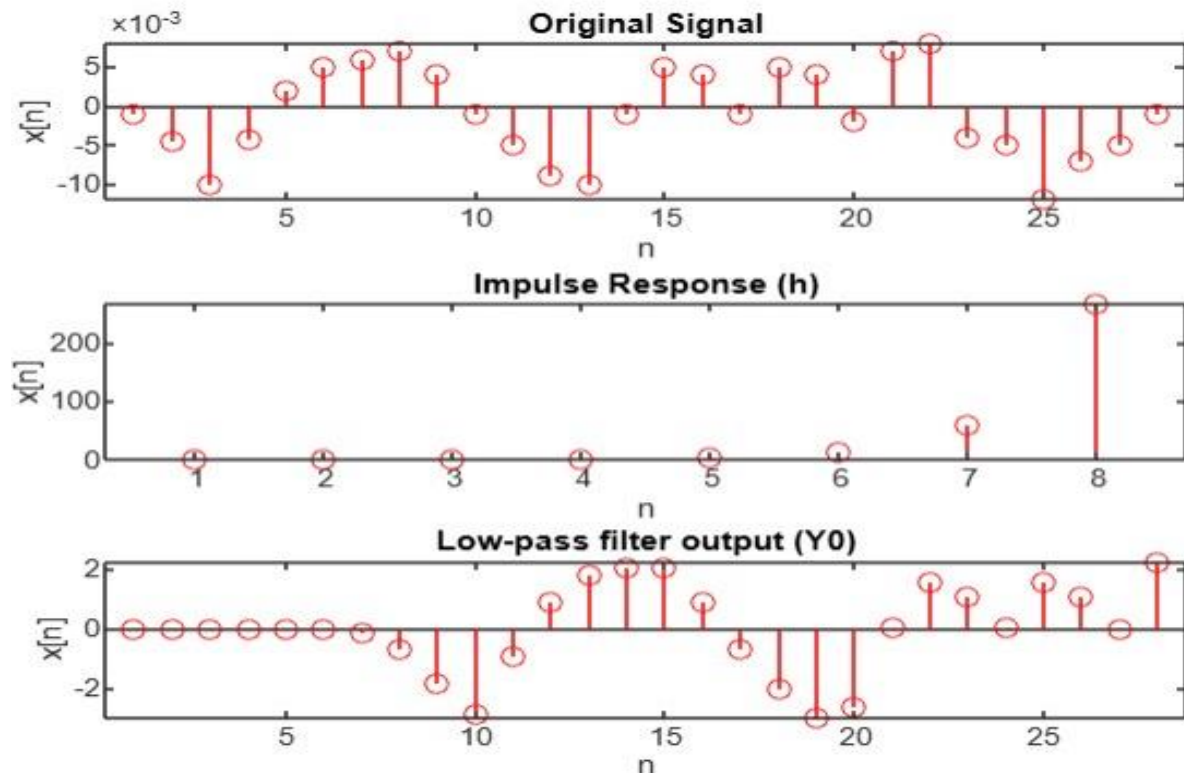
Here is the amendment to the code below I have written when M=200

```
title('Filtered Signal with Moving Average M =200');
```

As you can see from the 3 sets of results, they vary dependant on the value of the filter length. When the filter length (M) is equal to 10, there is too much distortion in the signal. When M is equal to 200, there is no information regarding the signal, the signal is just constant. When M is equal to 50, the signal is smoother, therefore this is the best value of the 3. Reflecting on the results, the value should ideally be in the middle of these values. Between 20 and 30 would have been a better value for the filter length.

Task 4

Part a/b



The first plot shows the original signal, the second plot shows the impulse response, and the third plot shows the low-pass filter output. The low-pass filter appears to smooth the beginning of the original signal, removing the high frequency noise.

Below is the code I created on MATLAB to formulate these plots.

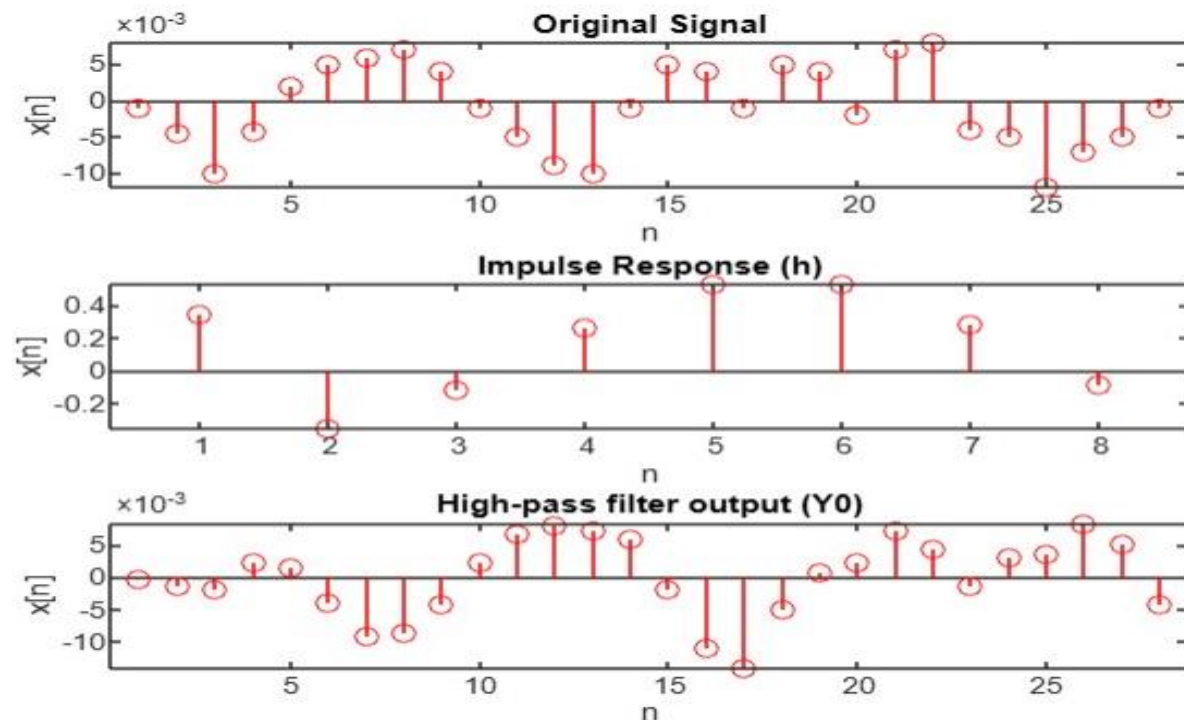
```
x=[-0.001 -0.0045 -0.01 -0.0042 0.002 0.005 0.006 0.007 0.004 -0.001 -0.005 -0.009
-0.01 -0.001 0.005 0.004 -0.001 0.005 0.004 -0.002 0.007 0.008 -0.004 -0.005 -
0.012 -0.007 -0.005 -0.001];
n=8; %number of responses to plot.
%y[n]=3.6y[n-1]+5y[n-2]-3.2y[n-3]+0.82y[n-4]+0.013x[n]-0.032x[n-1]+0.044x[n-2]-
0.033x[n-3]+0.013x[n-4]
%y[n]-3.6y[n-1]-5y[n-2]+3.2y[n-3]-0.82y[n-4]=0.013x[n]-0.032x[n-1]+0.044x[n-2]-
0.033x[n-3]+0.013x[n-4]
num=[0.013 -0.032 0.044 -0.033 0.013]; %x-coefficients
den=[1 -3.6 -5 3.2 -0.82]; %y-coefficients
h=impz(num, den, n);
subplot(3,1,1);
stem(x,'r');
title("Original Signal");
ylabel("x[n]");
xlabel("n");
subplot(3,1,2);
stem(h,'r');
title("Impulse Response (h)");
ylabel("x[n]");
```

```

xlabel("n");
disp(h'); %display 10 response values of h
Y0=filter(h',1,x); %output of low-pass filter
subplot(3,1,3);
stem(Y0,'r');
title("Low-pass filter output (Y0)");
ylabel("x[n]");
xlabel("n");

```

Part c/d



The first plot shows the original signal, the second plot shows the impulse response, and the third plot shows the high-pass filter output. The high-pass filter shows the rapid increases of the original signal hence the sharp plot of points on the third graph.

Below is the code I created on MATLAB to formulate these plots.

```

x=[-0.001 -0.0045 -0.01 -0.0042 0.002 0.005 0.006 0.007 0.004 -0.001 -0.005 -0.009
-0.01 -0.001 0.005 0.004 -0.001 0.005 0.004 -0.002 0.007 0.008 -0.004 -0.005 -
0.012 -0.007 -0.005 -0.001];
n=8; %number of responses to plot
%y[n]=2.7y[n-1]-3.3y[n-2]+2y[n-3]-0.57y[n-4]+0.35x[n]-1.3x[n-1]+2x[n-2]-1.3x[n-
3]+0.35x[n-4]
%y[n]-2.7y[n-1]+3.3y[n-2]-2y[n-3]+0.57y[n-4]=0.35x[n]-1.3x[n-1]+2x[n-2]-1.3x[n-
3]+0.35x[n-4]
num=[0.35 -1.3 2 -1.3 0.35]; %x-coefficients
den=[1 -2.7 3.3 -2 0.57]; %y-coefficients
h=impz(num, den, n);
subplot(3,1,1);
stem(x,'r');
title("Original Signal");
ylabel("x[n]");
xlabel("n");

```

```
subplot(3,1,2);
stem(h,'r');
title("Impulse Response (h)");
ylabel("x[n]");
xlabel("n");
disp(h'); %display 10 response values of h
Y0=filter(h',1,x); %output of low-pass filter
subplot(3,1,3);
stem(Y0,'r');
title("High-pass filter output (Y0)");
ylabel("x[n]");
xlabel("n");
```

Conclusion

To conclude, it's interesting to see how MATLAB is used within the engineering industry. The software can provide precise analysis and the ability to maintain and improve the integrity of signals. Signals can be modified to eliminate any unwanted interferences. Furthermore, through experience and extensive use of MATLAB or similar software, the capabilities they have can improve an extremely important tool within the industry.

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