Ryerson University

Department of Electrical, Computer, & Biomedical Engineering
Faculty of Engineering & Architectural Science

Course Title:	SIGNALS AND SYSTEM
Course Number:	ELE532
Semester/Year (e.g.F2016)	F2020

Instructor:	BEHESTI

Assignment/Lab Number:	4
Assignment/Lab Title:	FOURIER SERIES ANALYSIS

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A1.

$z(t) = x(t) * x(t)$

$t \le 0 \quad z(t) = 0$

$t \le 1 \quad z(t) = \int_0^1 1\,dt = 1$

$t \le 2 \quad z(t) = \int_0^2 1\,dt = 2$

$\vdots$

$t \le 10 \quad z(t) = \int_0^{10} 1\,dt = 10$

$t \le 11 \quad z(t) = \int_1^{10} 1\,dt = 9$

$\vdots$

$t \le 19 \quad z(t) = \int_9^{10} 1\,dt = 1$

$t \ge 20 \quad z(t) = 0$

A.2- The answer is in the zip folder

A3.

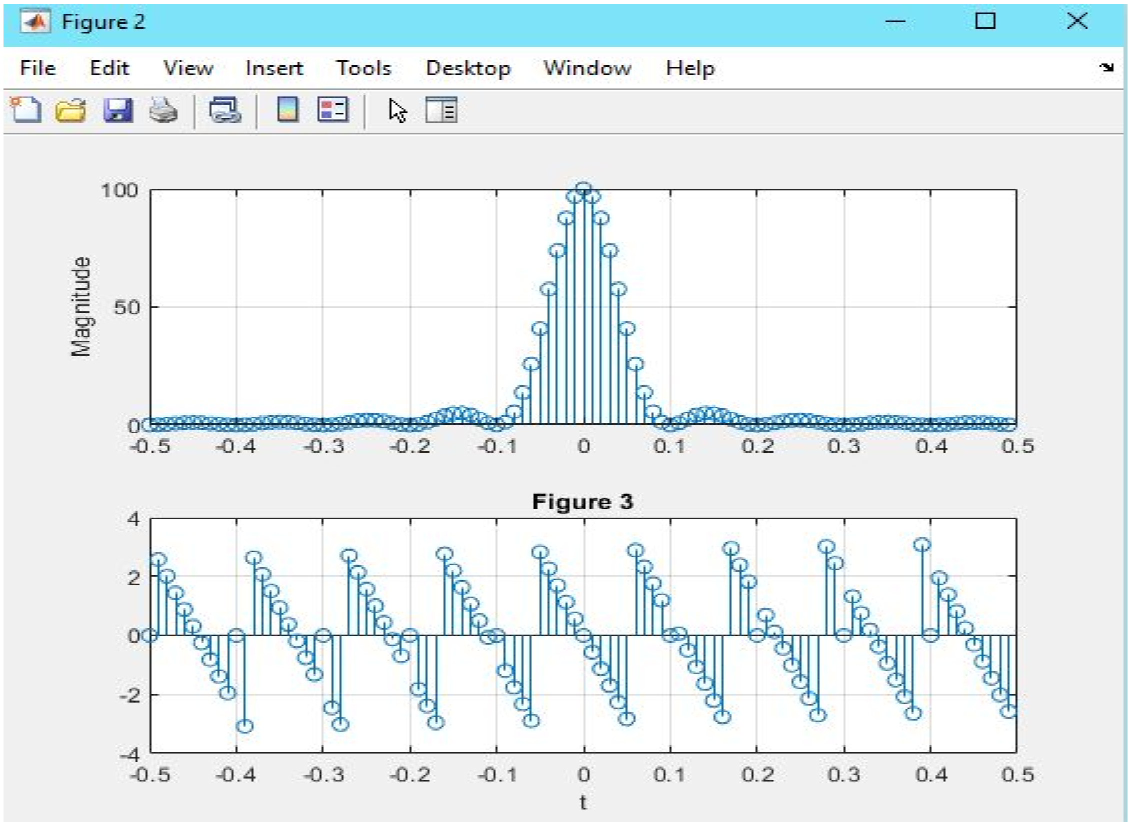


Figure 2
File Edit View Insert Tools Desktop Window Help
Magnitude
Figure 3
t

A.4

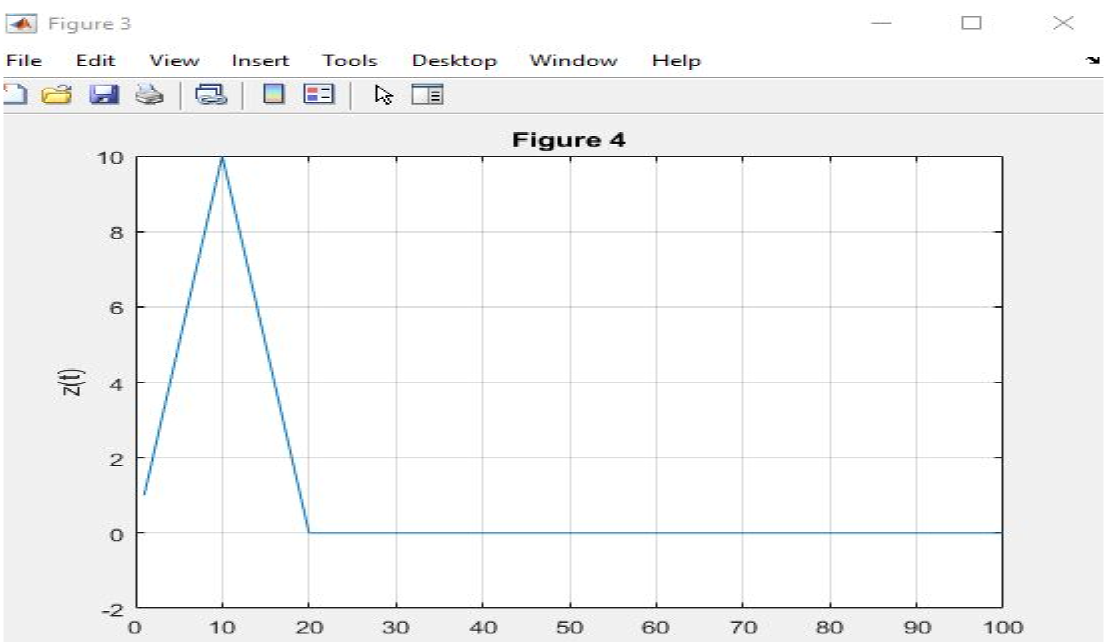
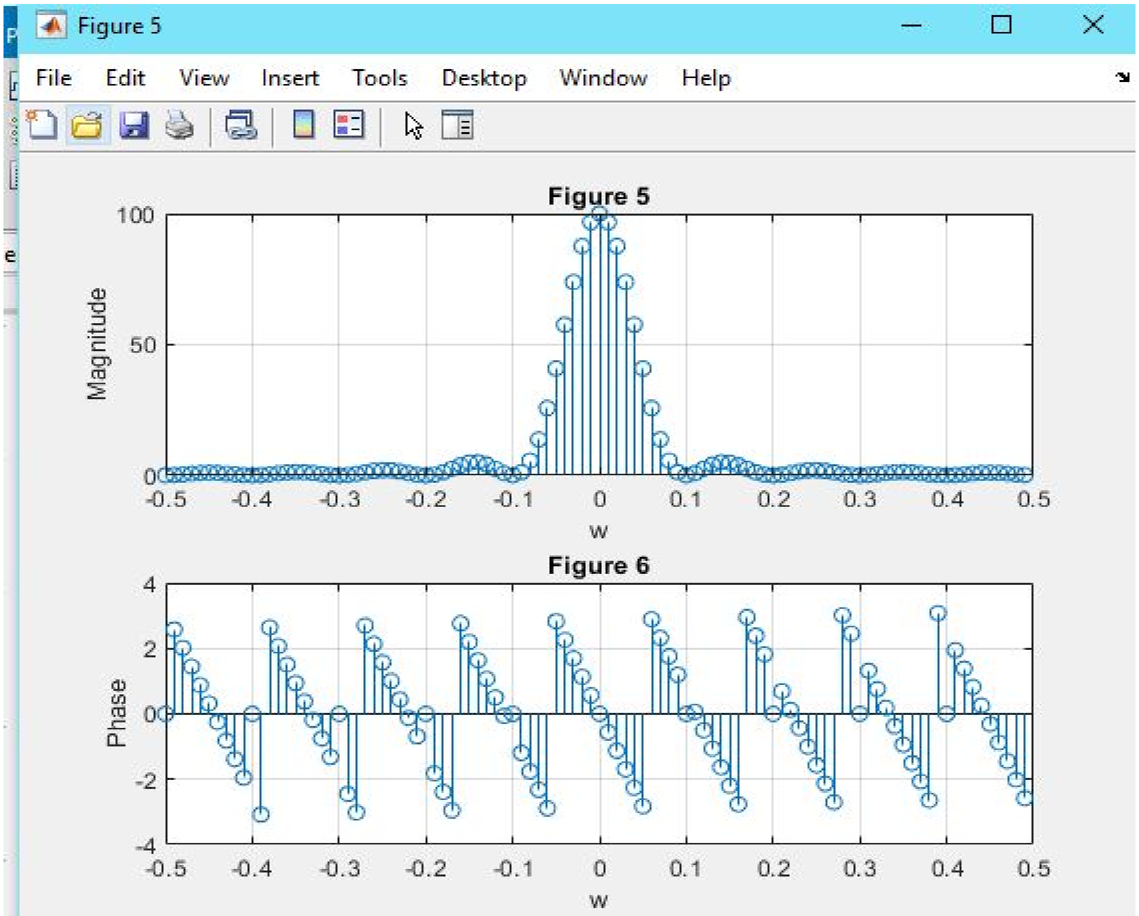


Figure 3
File Edit View Insert Tools Desktop Window Help
Figure 4
z(t)
t



Both results from A.4 and A.1 are the same. The convolution property of the Fourier transform is being demonstrated.

A.5

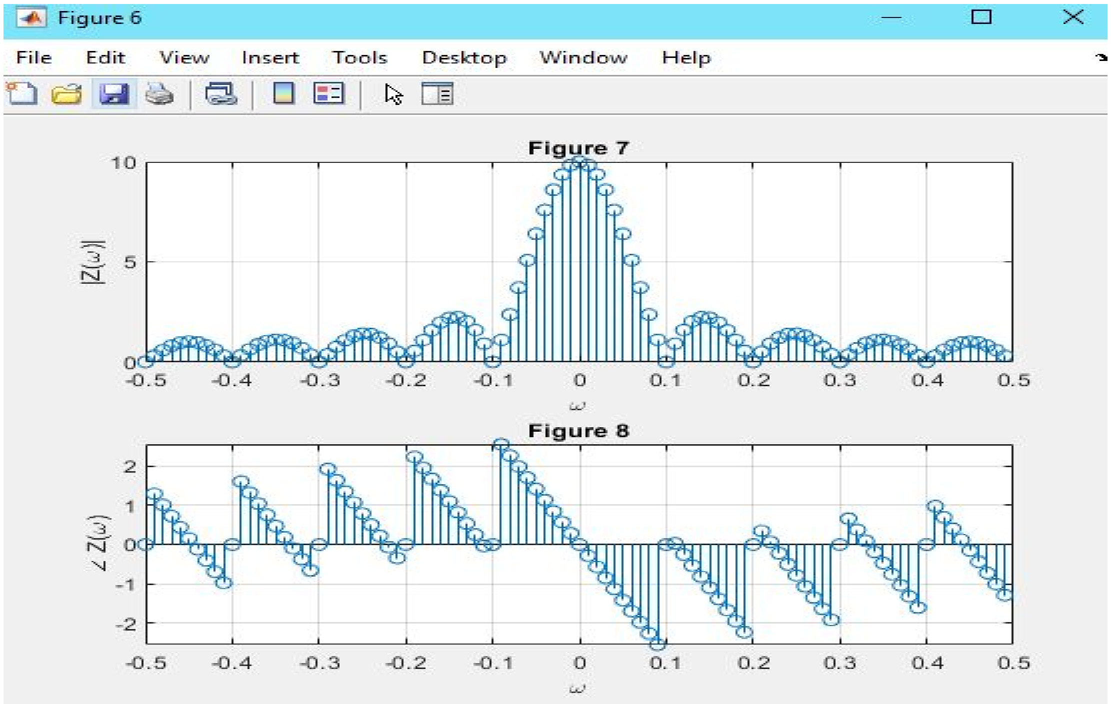


Figure 6
File Edit View Insert Tools Desktop Window Help
Figure 7
|Z(ω)|
ω
Figure 8
∠ Z(ω)
ω

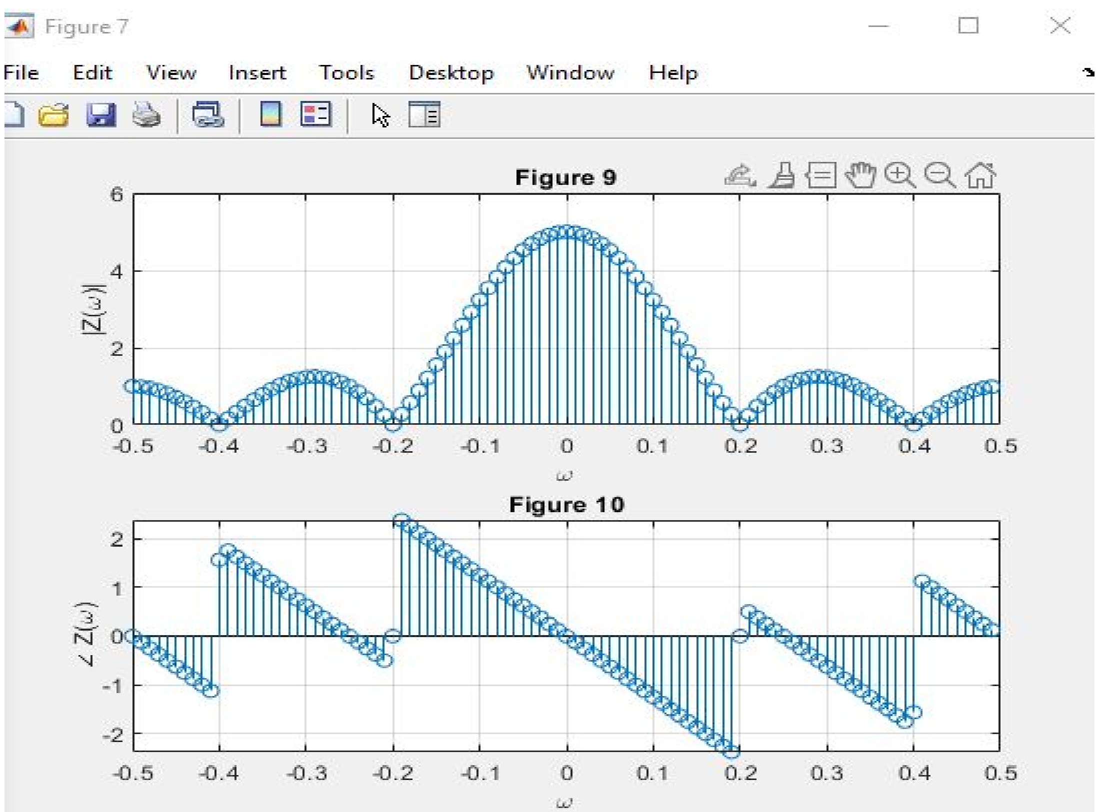
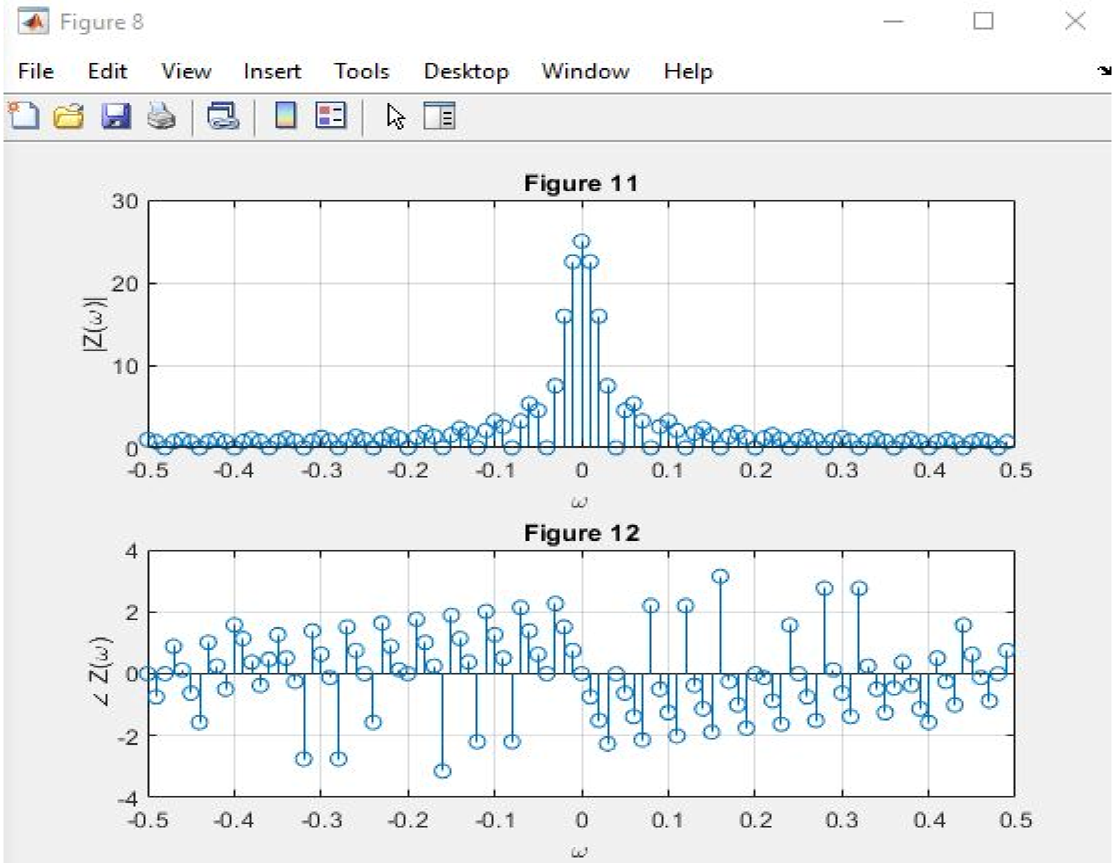


Figure 7
File Edit View Insert Tools Desktop Window Help
Figure 9
|Z(ω)|
ω
Figure 10
∠ Z(ω)
ω



The three pulses have the same shape and amplitude, however, compared to figure 7, the other two are being time stretched. The Fourier transform property used is the time scaling property.

A.6

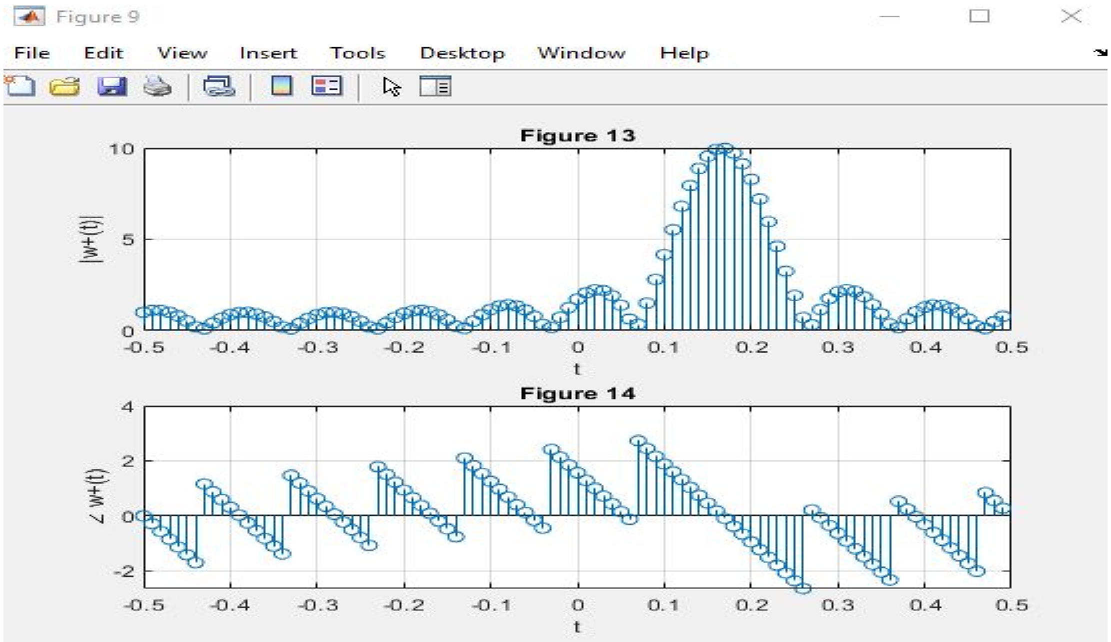


Figure 9
File Edit View Insert Tools Desktop Window Help
Figure 13
|w+(t)|
t
Figure 14
∠ w+(t)
t

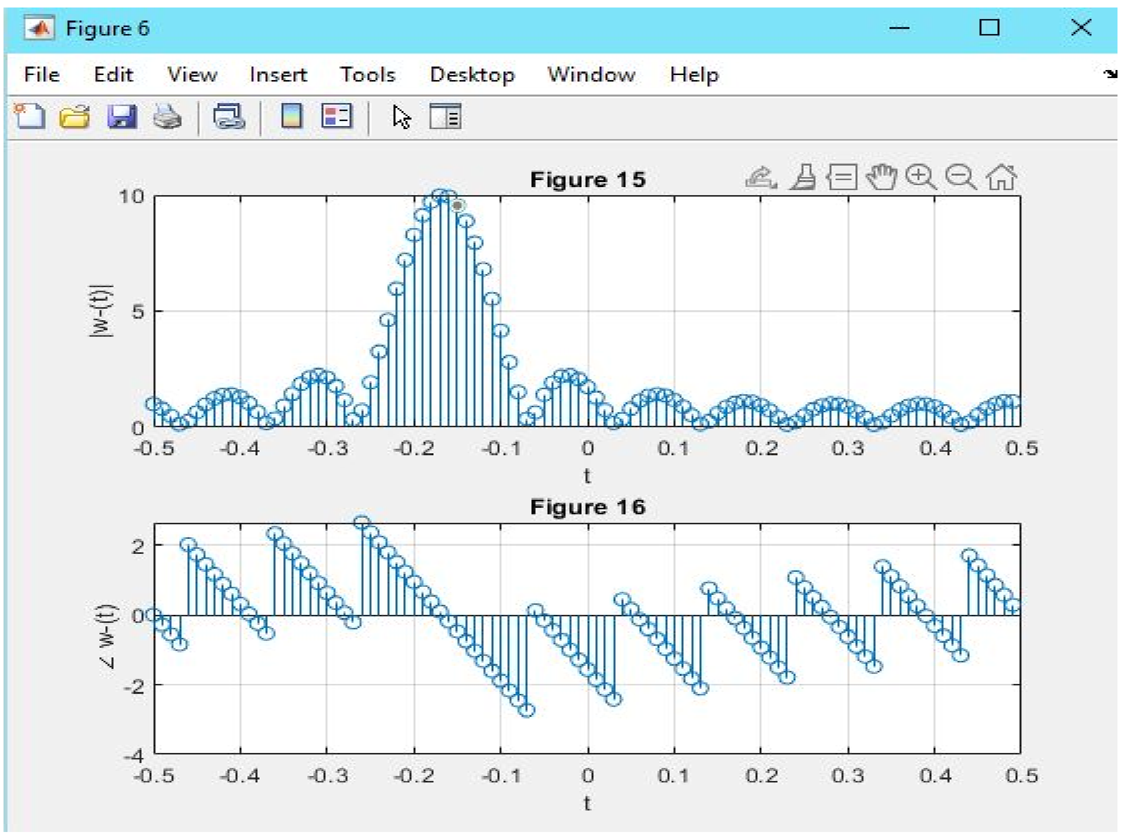
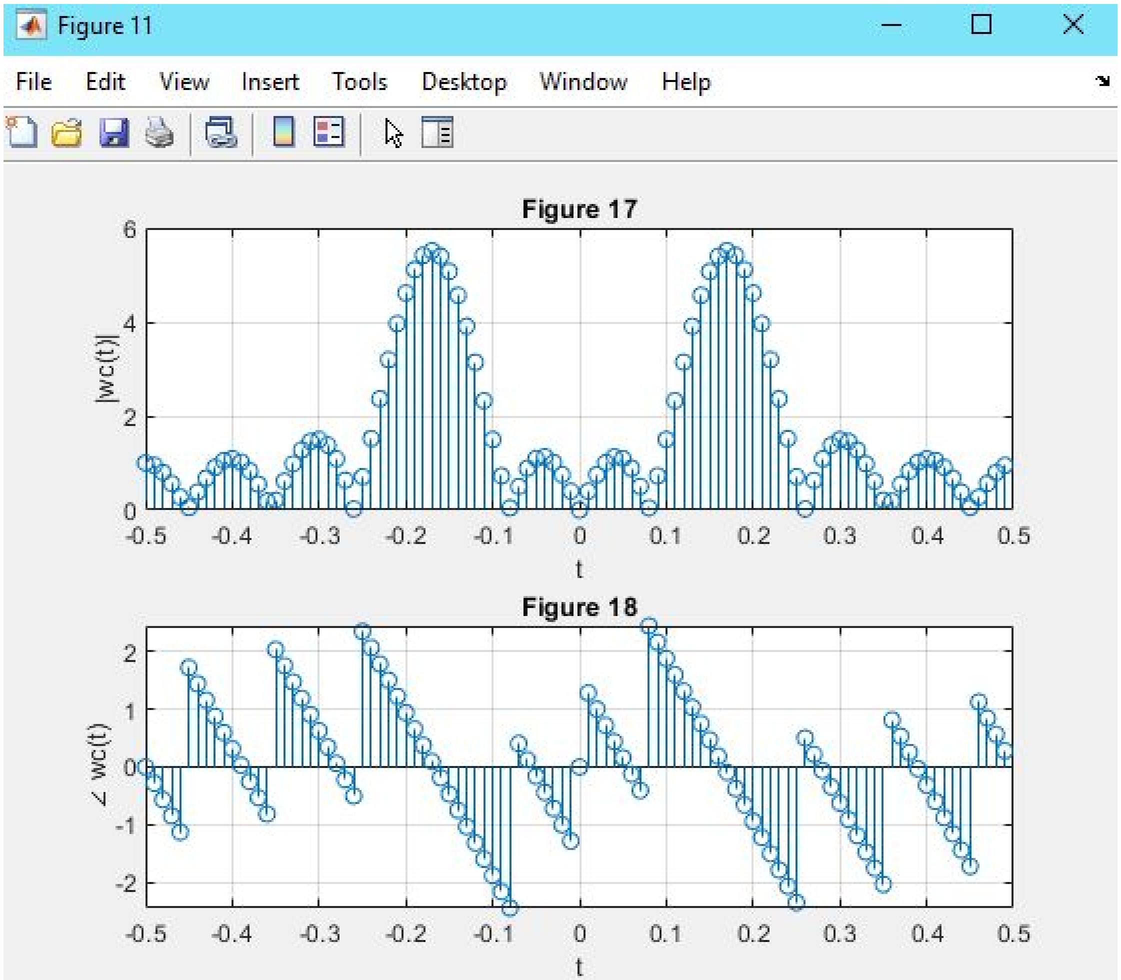


Figure 6
File Edit View Insert Tools Desktop Window Help
Figure 15
|w-(t)|
t
Figure 16
∠ w-(t)
t

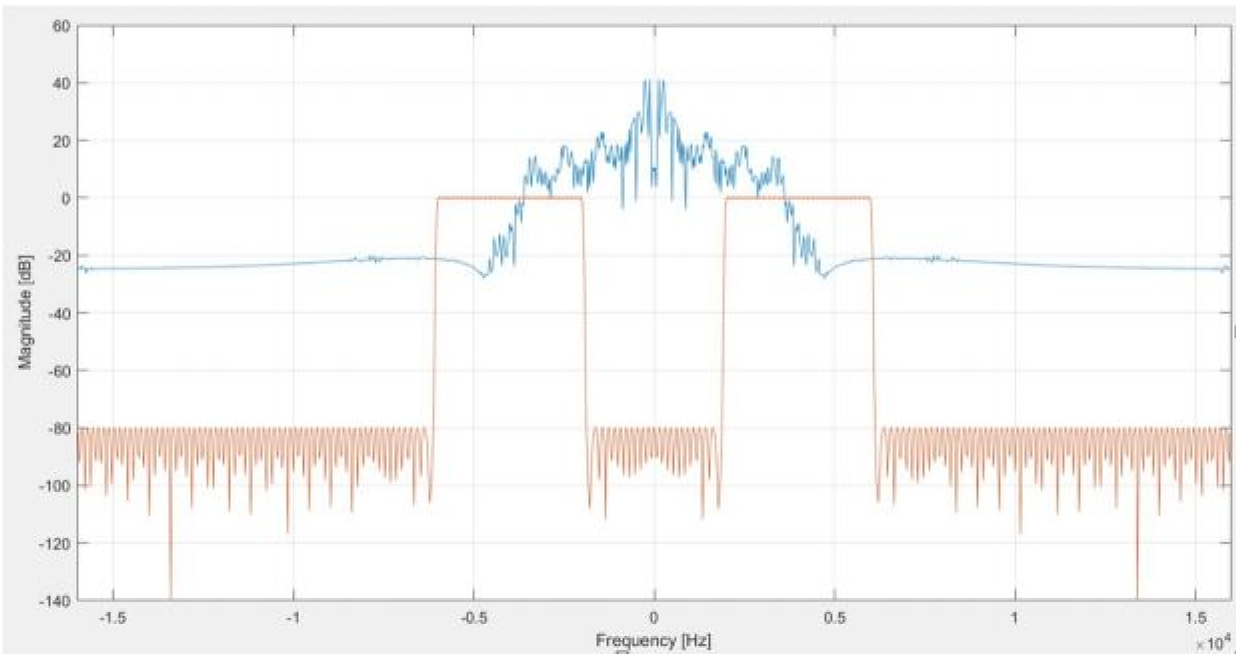


The Fourier transform property being demonstrated is the frequency shift property. The graph w+(t) shifts to the right by pi/3. w-(t) shifts to the left by pi/3. wc(t) contains both the shifts.

Part B :

The problem in this question involves solving the static sound that makes the xspeech audio incomprehensible. This is because hchannel cannot pick up the xspeech signal due to the passband of hchannel ranging from about 2-3khz to 6khz.You can see in the graph below that xspeech's signal does not get detected under hchannel completely. The goal is to make xspeech more comprehensible to fit in the passband of hchannel by modulating it.

[xspeech(blue) and hchannel(red) plotted together]



We can do this by

1.Putting it xspeech signal through the first low pass filter that uses the portion of xspeech that we want to work with which would be in the +/- 2000Hz cuttoff range.

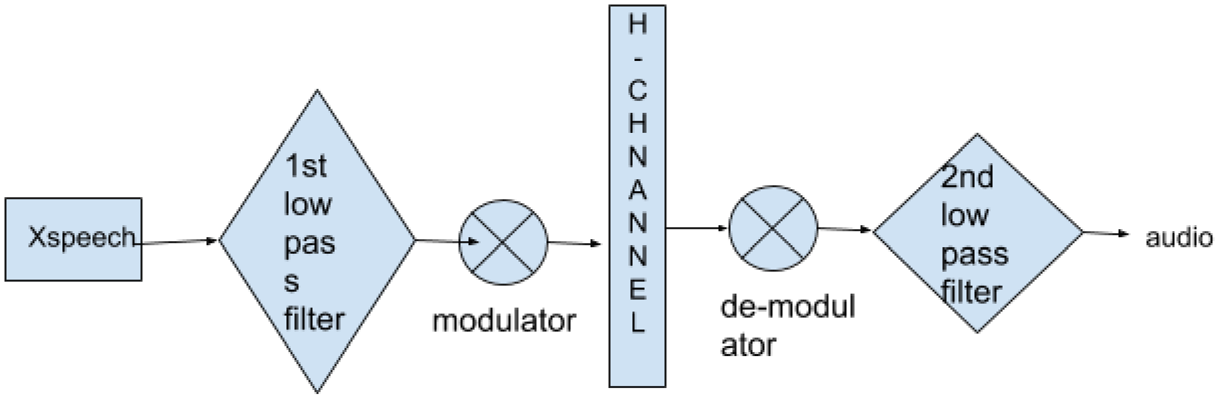
2 . Modulating the signal from step 1 by shifting it by 3000Hz to fit in the passband of hchannel. We do this by using the osc function and setting the parameter to 3000).Note that 3000 is an approximation.

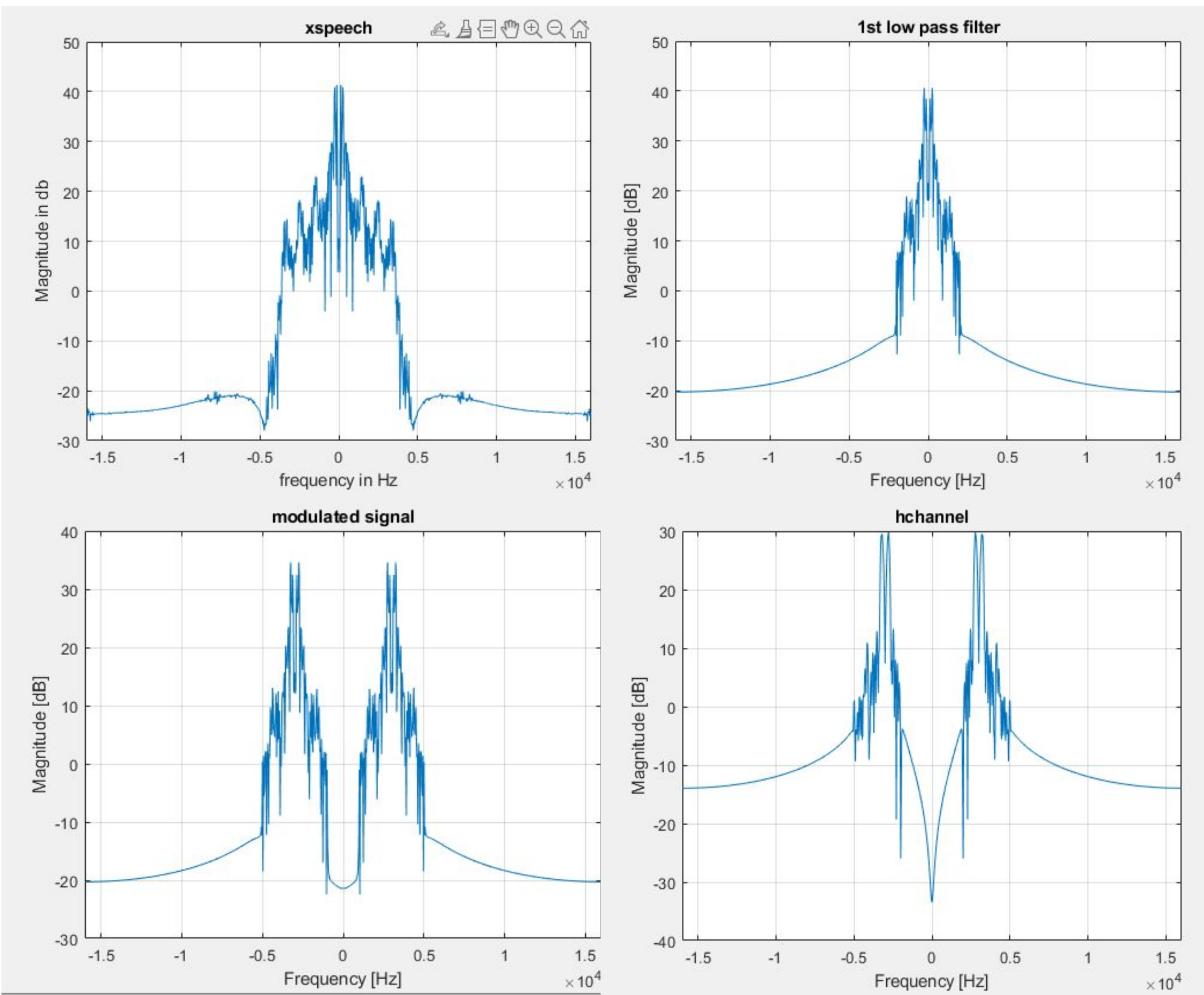
3. We will use this signal from step 2 and send it through hChannel .

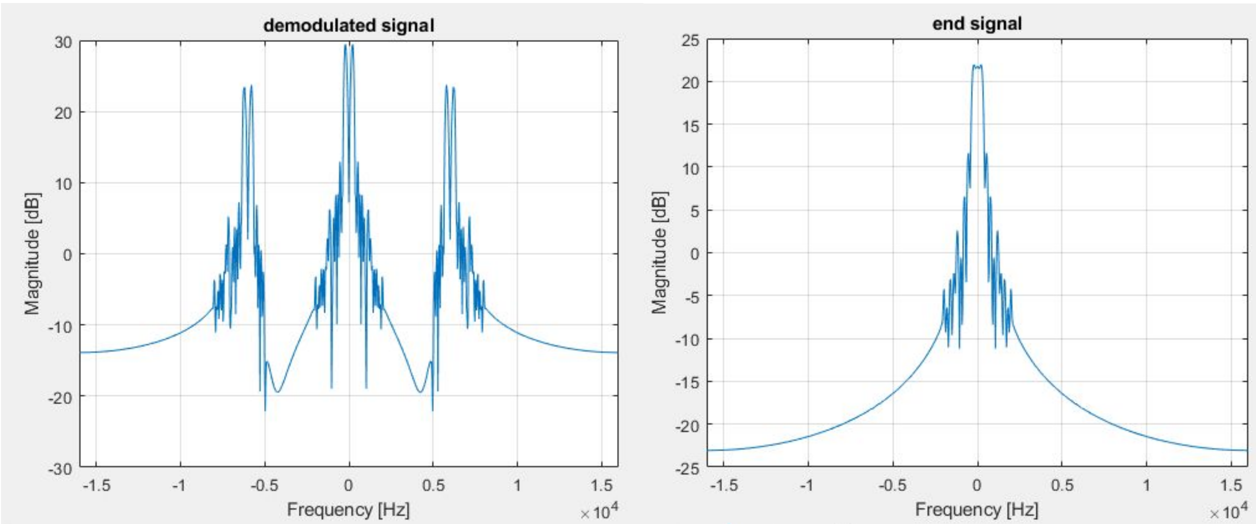
4.We then demodulate the signal from step 3 to get it's original state.

5.We then put the signal in step 4 through another low pass filter to only use the portion of the signal that we need/want to hear.

You can now hear the differences between the original signal's sound and the outcome of the signal's sound put through our design. Our design produces a much more comprehensible audio than the original.


Xspeech
1st
low
pas
s
filter
modulator
H
-
C
H
N
A
N
N
E
L
de-modul
ator
2nd
low
pass
filter
audio


xspeech
Magnitude in db
frequency in Hz
1st low pass filter
Magnitude [dB]
Frequency [Hz]
modulated signal
Magnitude [dB]
Frequency [Hz]
hchannel
Magnitude [dB]
Frequency [Hz]


demodulated signal
Magnitude [dB]
Frequency [Hz]
$\times 10^4$
end signal
Magnitude [dB]
Frequency [Hz]
$\times 10^4$