



Cairo University



Faculty of Engineering

Electronics and Electrical Communications Engineering Department

Second Year

Course: ELC2030

## Signals Project Report

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# Communication system simulation

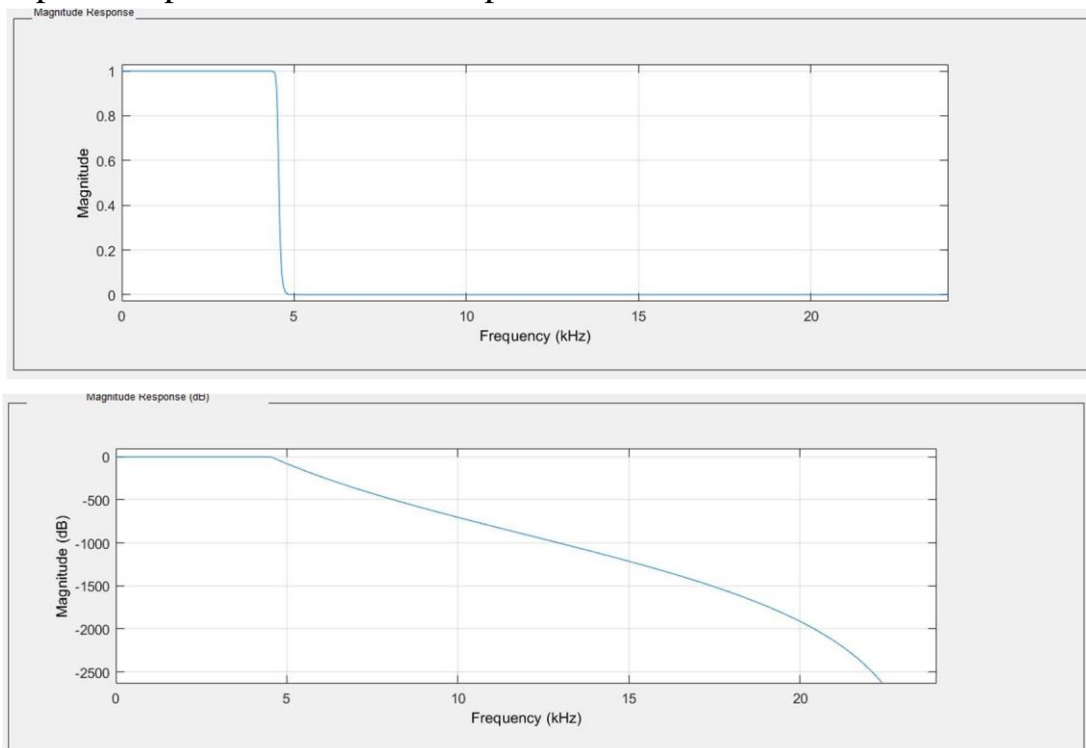
a. We used the function audiorecorder to record the two segments and we use the function audiowrite to save them. We used sampling rate 44100 HZ , and bit depth 16.

**Fs (44100):** because the human can receive and perceive up to 20khz, and to reproduce the signal the sampling freq must be at least twice the highest frequency sampling frequency of 44.1 kHz allows for accurate representation of audio signals up to approximately 22 kHz (sampling theorem).

**Bit depth (16):** determines the number of possible amplitude values we can record for each audio sample. The higher the bit depth, the more amplitude values per sample are captured to recreate the original audio signal, **Higher bit depths mean higher resolution audio; if the bit depth is too low, some information of the original audio signal will be lost.**

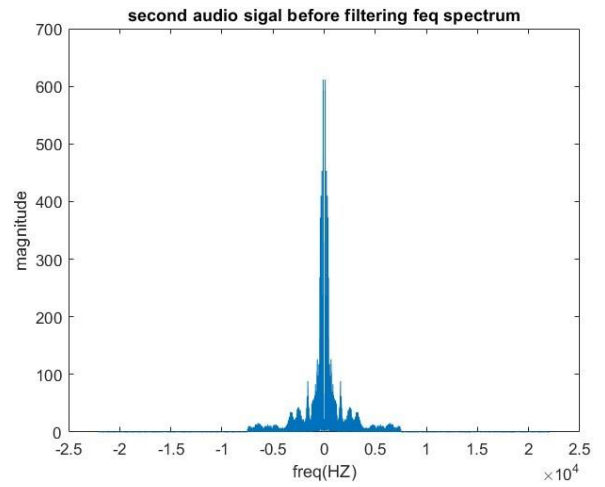
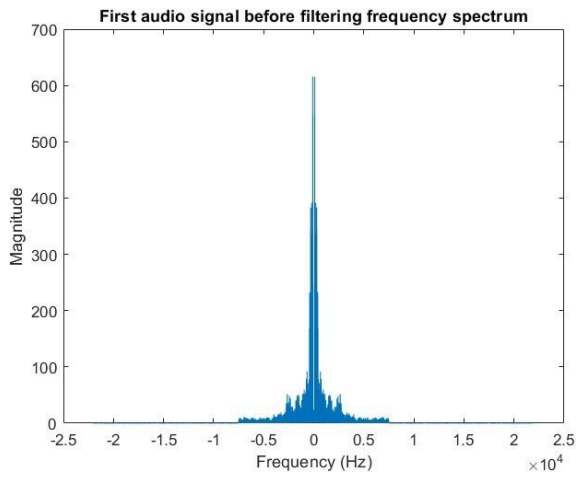
16-bit: 65,536 values

b. We use filter designer tool to make low pass filter ,We iterate a number of low pass filters to Limit the maximum frequency of both signals to a suitable value and we choose lpf with fpass 4500Hz and fstop 5000Hz.

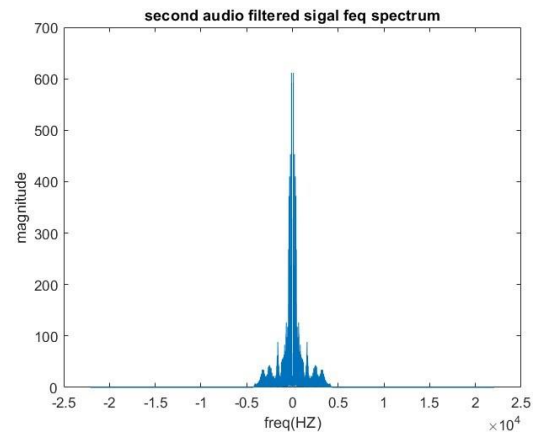
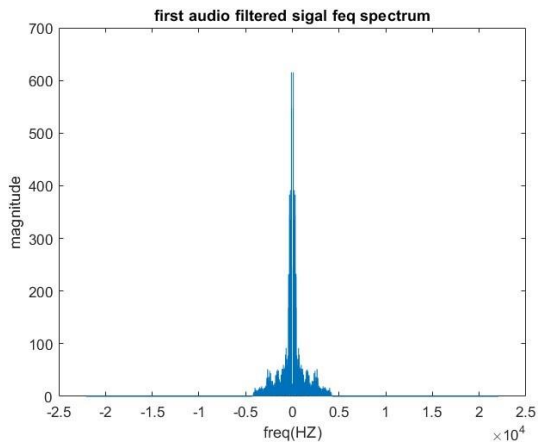


*Figure 11 the frequency response of the filter*

C.



*Figure 12 magnitude spectrum of both signals before filtering against the frequency*



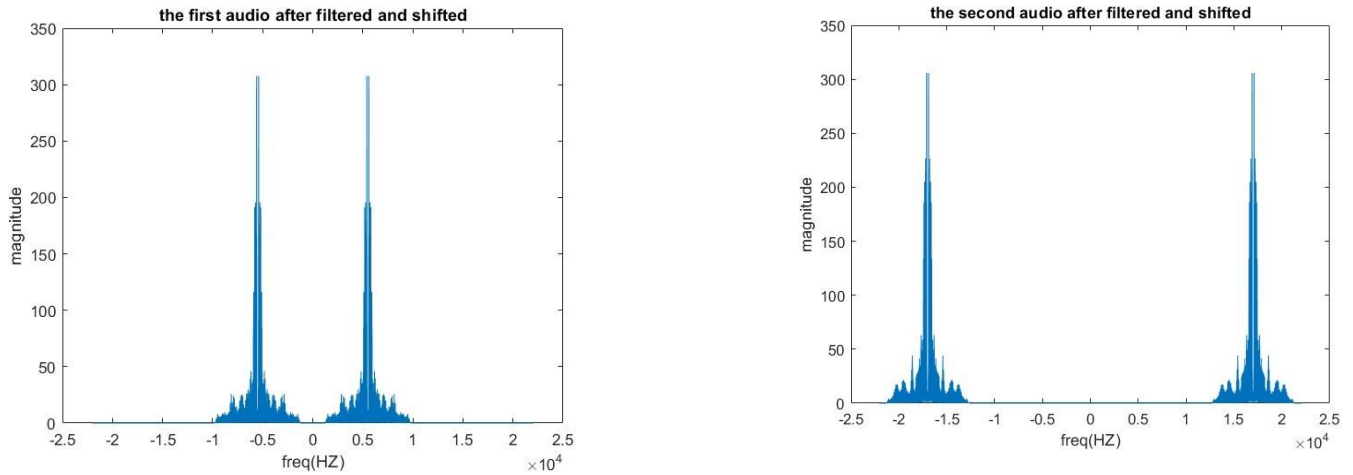
*Figure 13 magnitude spectrum of both signals after filtering against the frequency*

**d.** We multiply each filtered signal by a carrier ( $\cos(2\pi f_c t)$ )

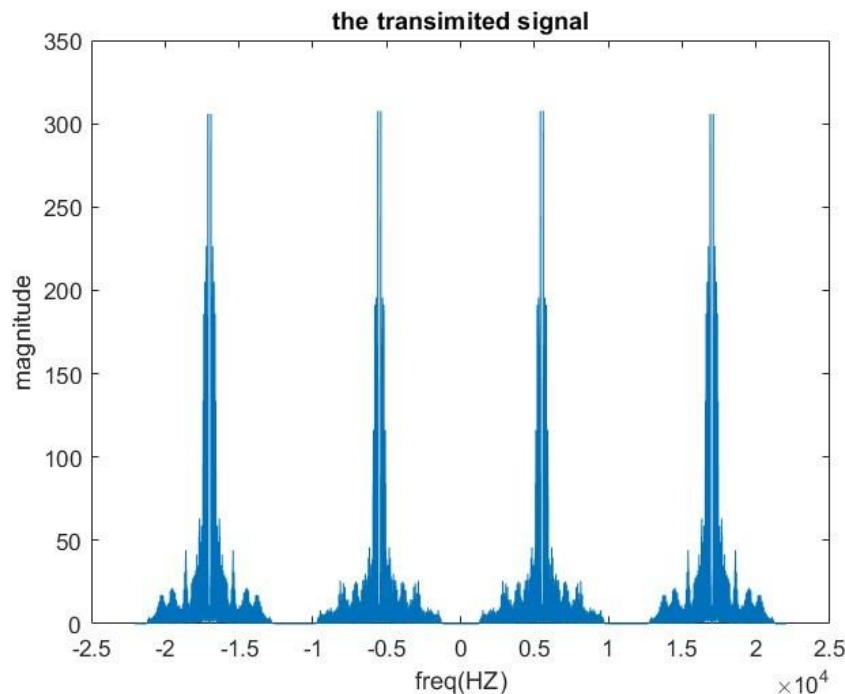
We shift the first signal by 5500Hz And the second by 17000Hz

And we add them in one signal (the transmitted signal)

We choose the first carrier frequency to be greater then ( $f_{pass}=4500$ ) and the second carrier to be greater than (first carrier frequency +  $2*f_{pass}=13500$ ) to avoid any interference between two signals in the transmitted signal.

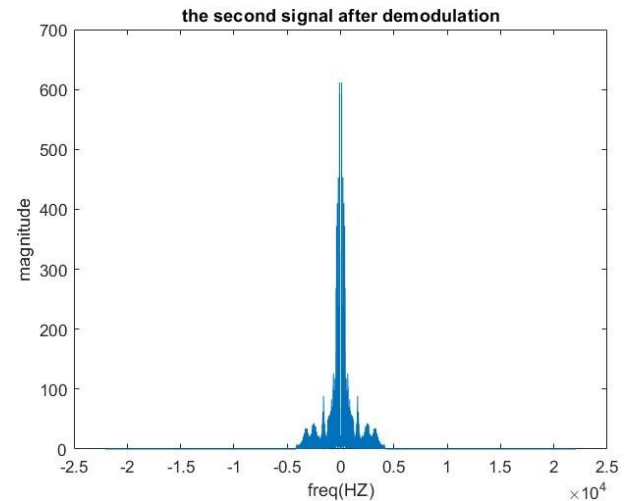
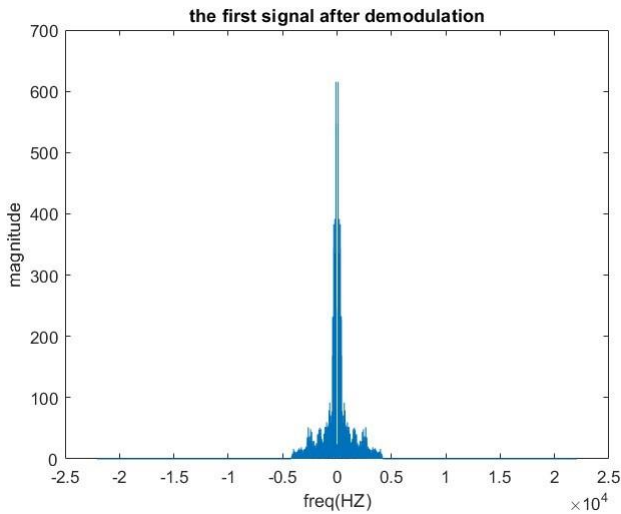


*Figure 14 magnitude spectrum of both signals after filtering and multiplying by carrier against the frequency*



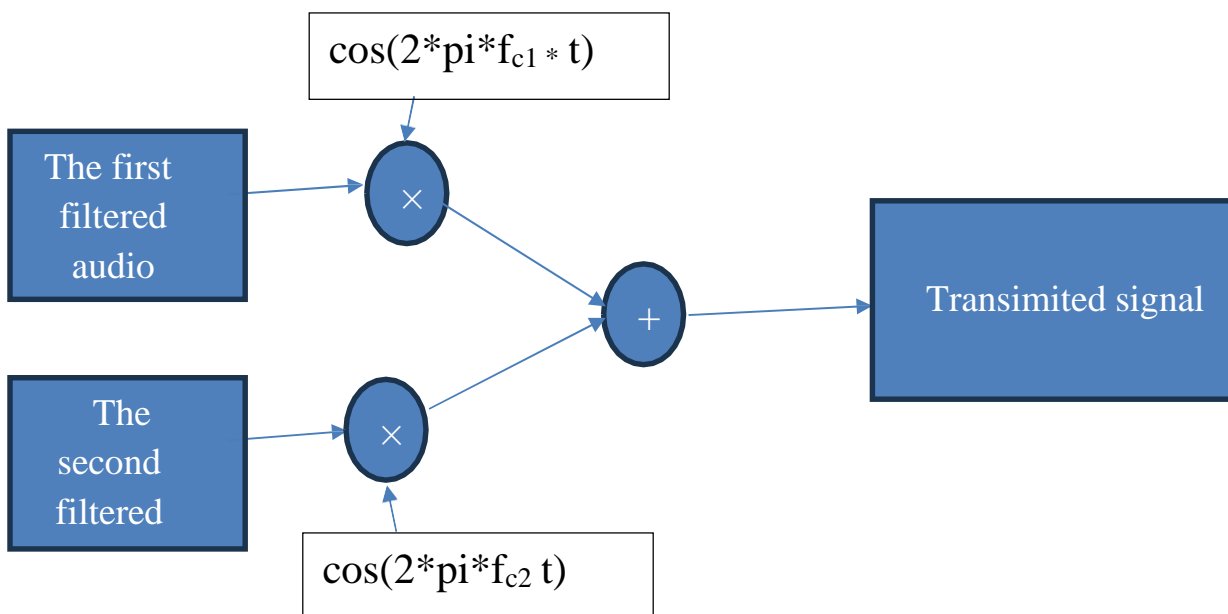
*Figure 15 magnitude spectrum of transimitted signal against the frequency*

e. If we need to obtain each of the two signals from the transmitted signal by the same carrier that we multiply by it in (d) then we use the same filter that we design on it. We don't make a band pass filters because we didn't need it as the signals didn't interfere.

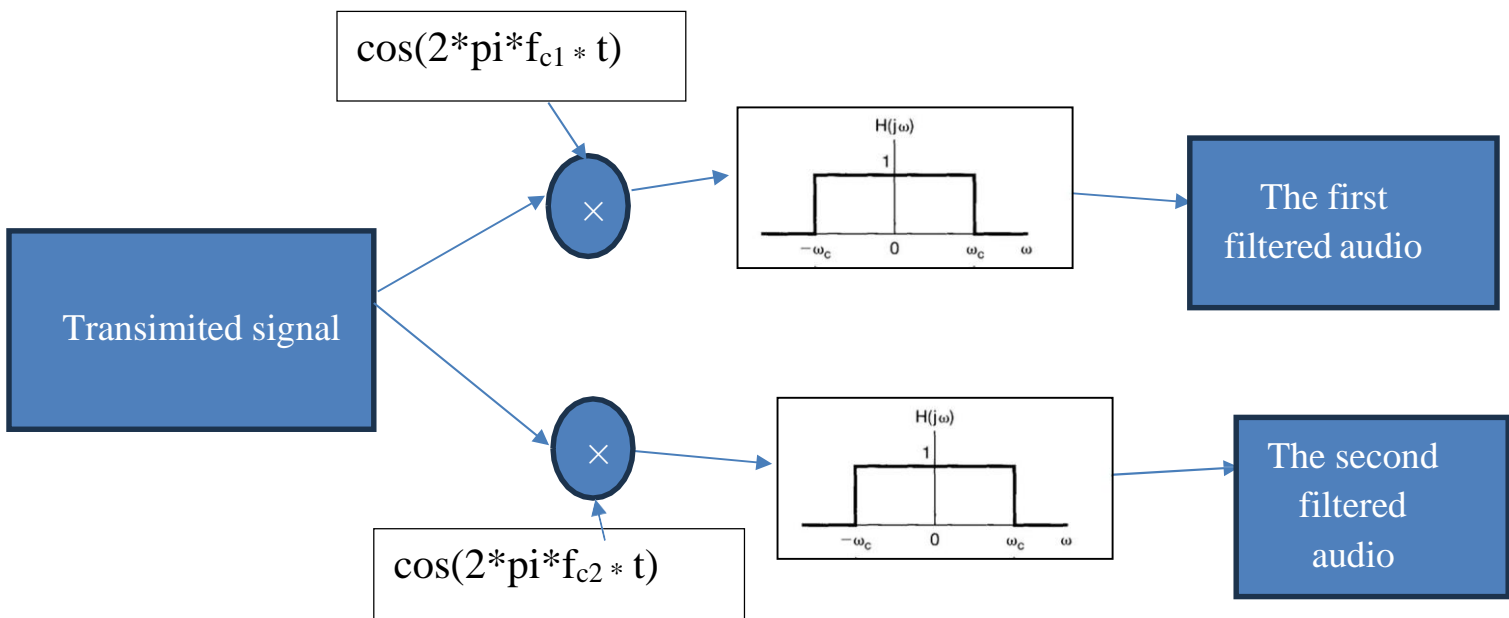


*Figure 16 magnitude spectrum of each signal after demodulation against the frequency*

## Transmitter



## receiver



### Equations of the operations in the receiver

Time domain	Frequency domain
1) $\text{Transmitted} \times \cos(2\pi f_c t) = y(t)$	1) $\frac{1}{2}(\text{transmitted}(f-f_c) + \text{transmitted}(f+f_c))$ $= \frac{1}{2} [\text{transmitted}] * [\delta(f-f_c) + \delta(f+f_c)] = Y(f)$
2) $2 \times (y(t) * h(t)) = \text{send signal}$	2) $2 \times Y(f) \times H(f) = \text{the send signal}$

First : we shifted the transimitted signal by carrier frequency

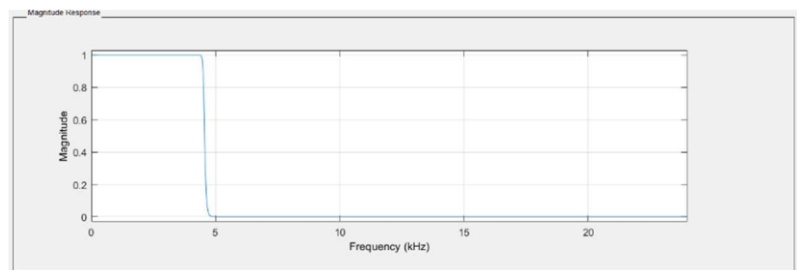
Second: we make to it a low pass filter

$\times$  : multiply

$*$  : convolution

$h(t)$  : function of low pass filter in time domain.

$H(f)$  : Fourier transform of  $h(t)$  ,  
function of low pass filter in frequency domain.



## References:

[1] <https://www.izotope.com/en/learn/digital-audio-basics-sample-rate-and-bit-depth.html>

[2] [https://www.mathworks.com/help/matlab/import\\_export/record-and-play-audio.html](https://www.mathworks.com/help/matlab/import_export/record-and-play-audio.html)