Cairo University

Faculty of Engineering

Electronics and Electrical Communications Engineering Department

Second Year

Course: ELC2030

Signals Project

Report

Prepared By:

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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 sapling 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 belost.**

16-bit: 65,536 values

b. We use filer 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.

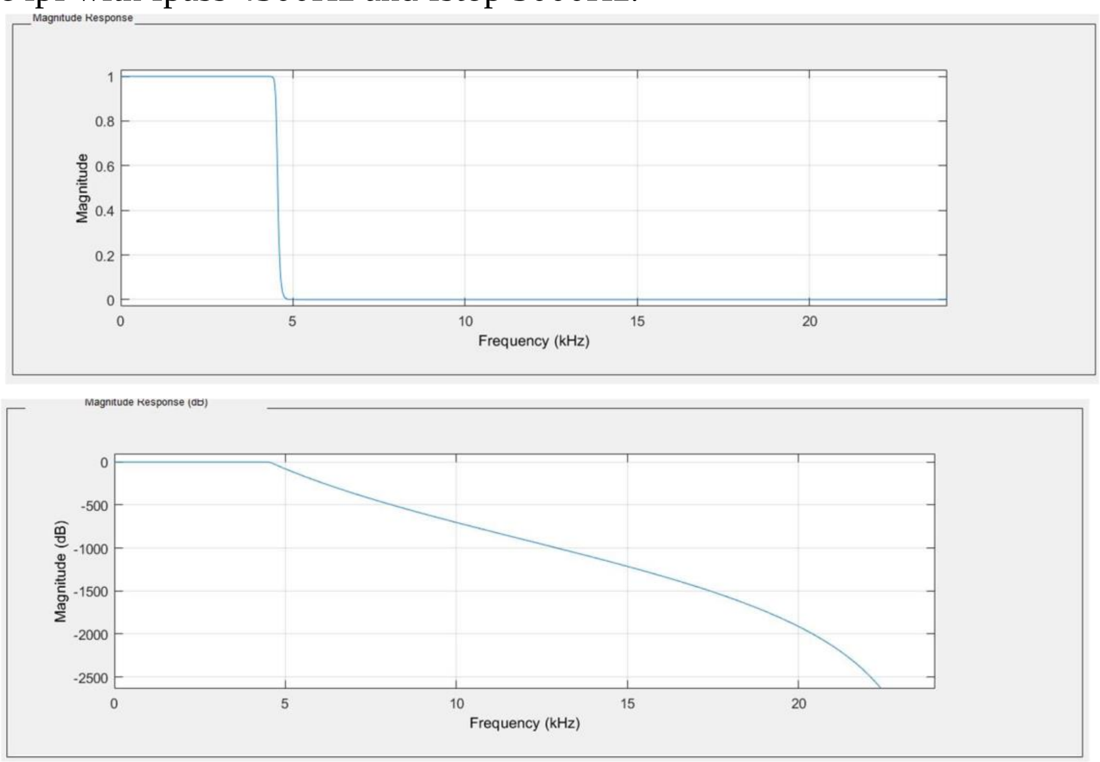


Figure 1. the frequency response of the filter

c.

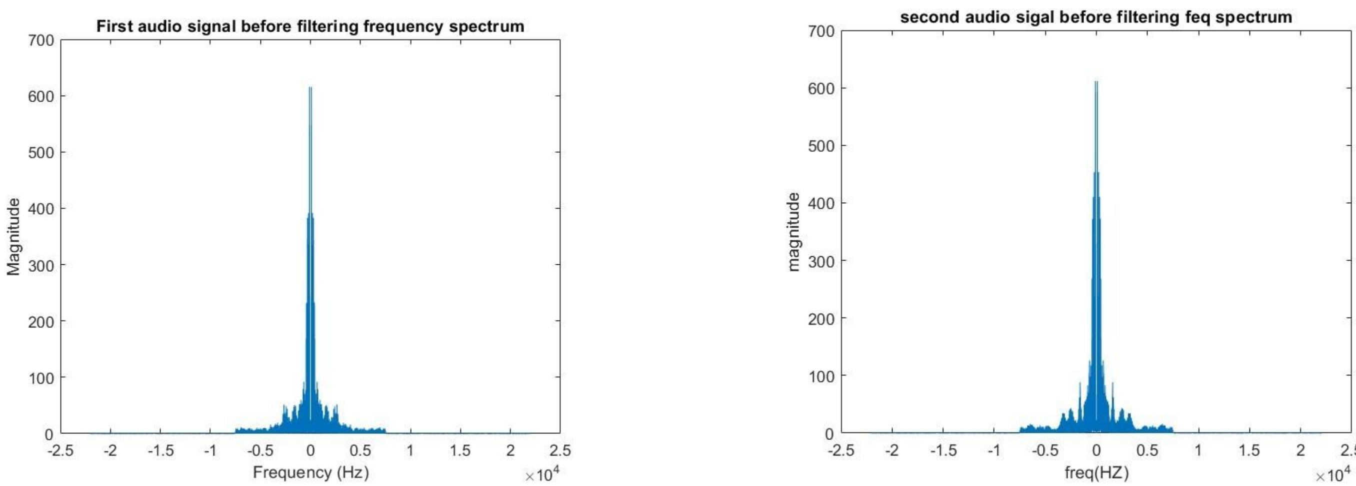


Figure 2. magnitude spectrum of both signals before filtering against the frequency

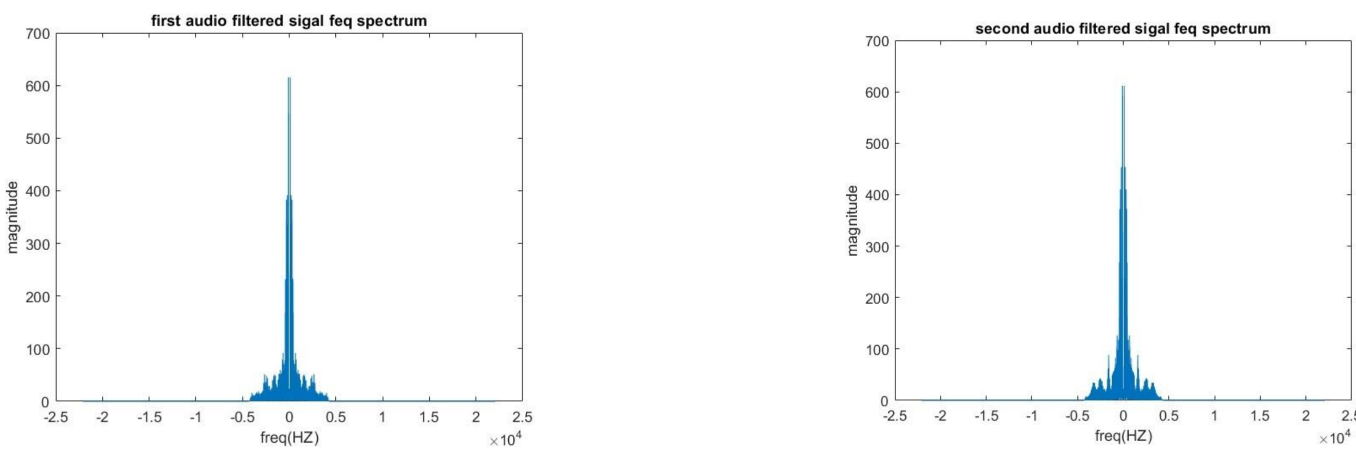


Figure 3. magnitude spectrum of both signals after filtering against the frequency

d. We multiply each filtered signal by a carrier (cos(2*pi*fc*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 (fpass=4500) and the second carrier to greater than (first carrier frequency + 2*fpass =13500) to avoid any interference between two signals in the transmitted signal.

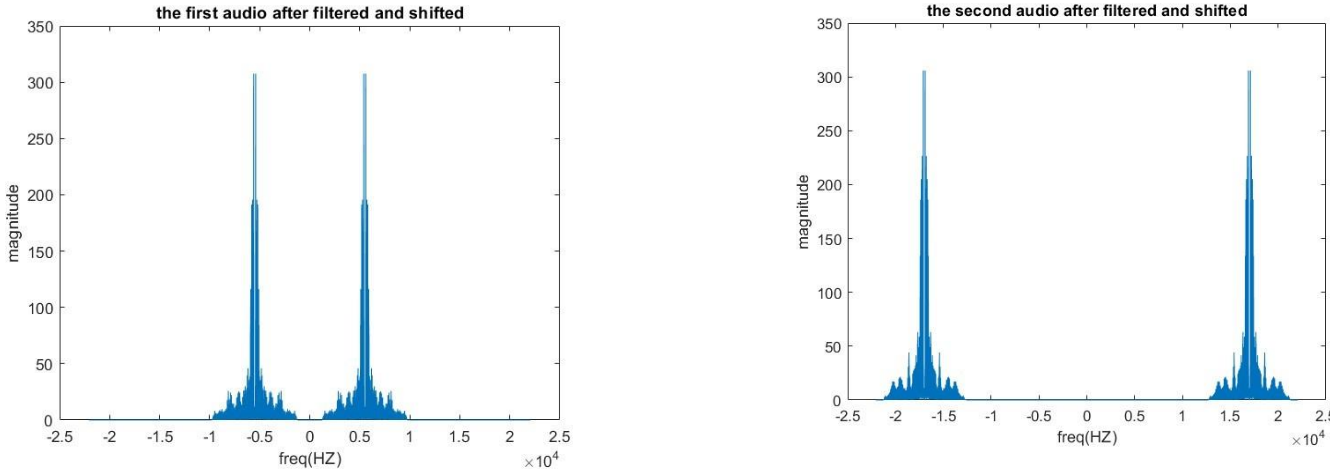


Figure 4. magnitude spectrum of both signals after filtering and multipling by crrier against the frequency

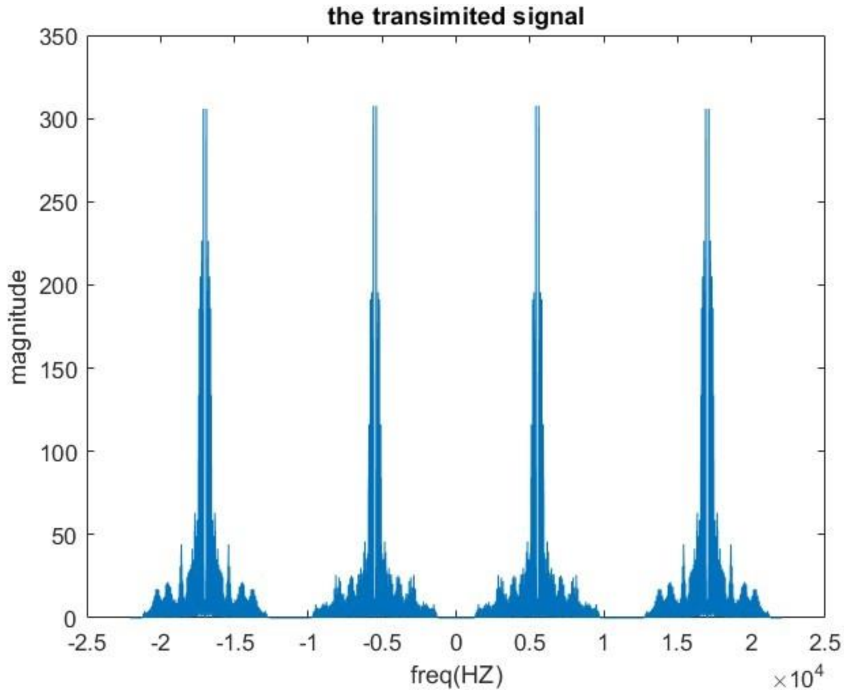
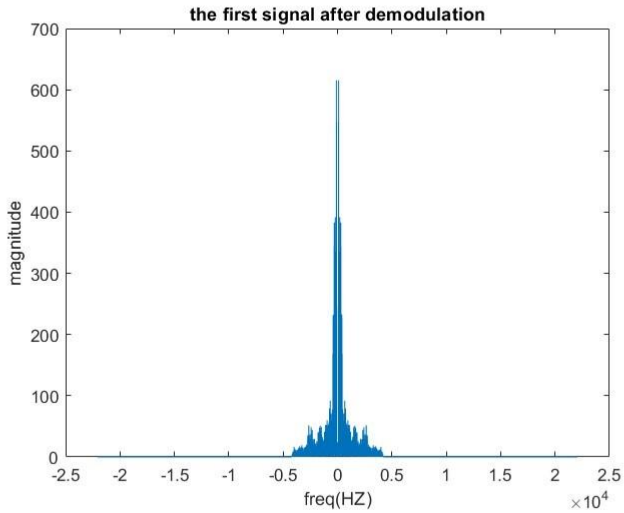


Figure 5. magnitude spectrum of transimitted signal against the frequency

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.



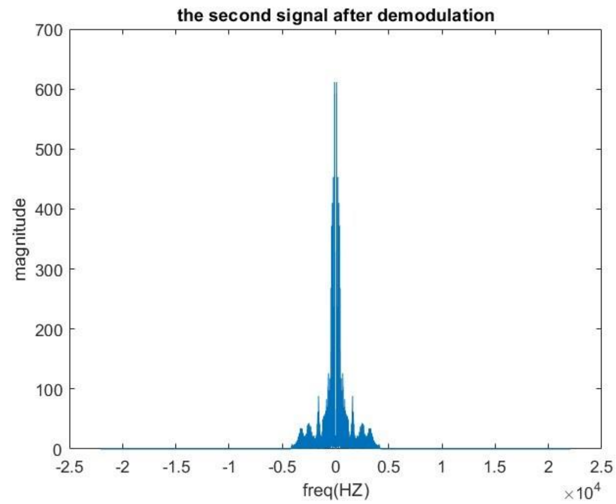
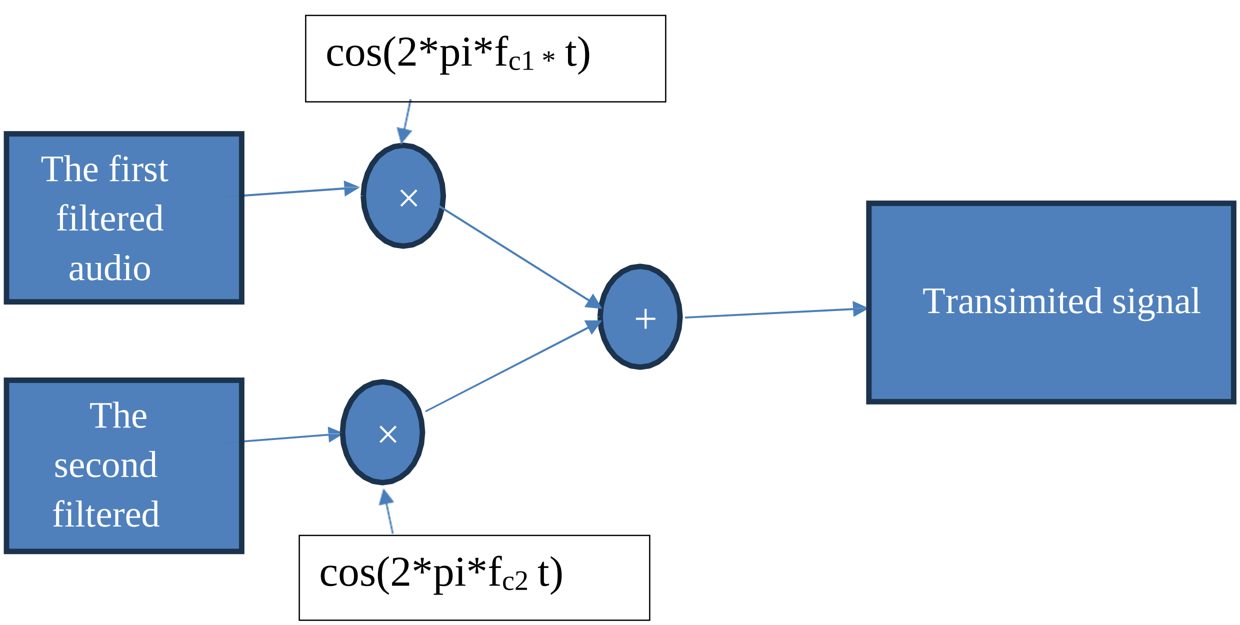
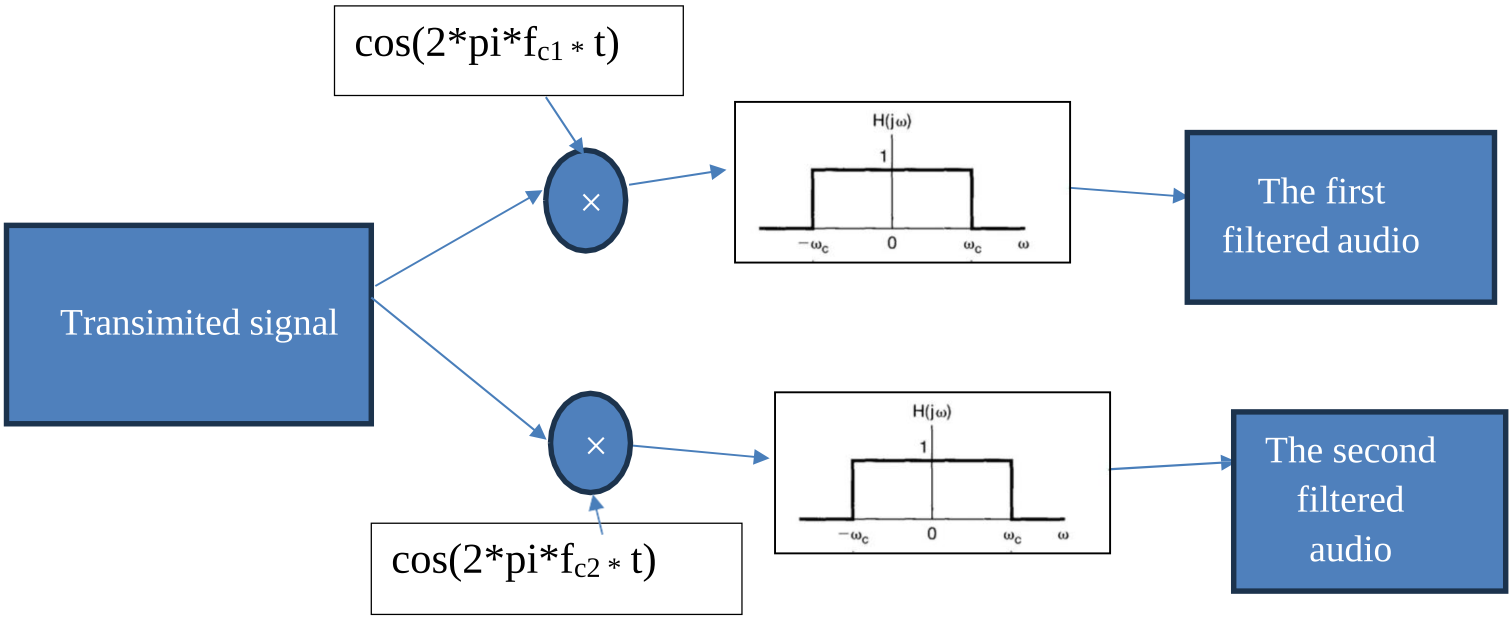


Figure 6. magnitude spectrum of each signal after demodulation against the frequency

Transmitter



receiver

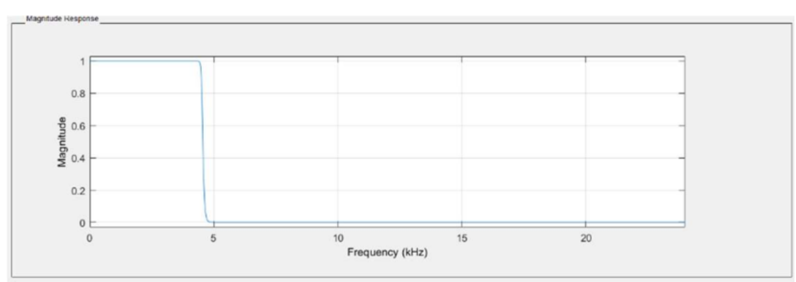


Equations of the operations in the reciever

Time domain	Frequency domain
1)Transimited ×cos(2 ×pi ×fc×t)=y(t)	1)½(transimited(f-fc)+transimited(f+fc)) = ½ [transimited] ∗[δ(f-fc)+ δ(f+fc)]= Y(f)
2) 2× (y(t) ∗ h(t))= send signal	2) 2×Y(f)×H(f)=the send signal

First : we shifted the transimited signal by carrier frequency
Second: we make to it a low pass filter

× : 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.



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[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