Task 2

- **Explanation: Black color font**
- **Headlines: Blue color Font**
- **Notes: Red color font**
- **Equation: brown color**

a. We will select sampling frequency= 44100 Hz

Because the frequency that a human can hear ranges from 20 Hz to 20,000 Hz, and in order for the sound to be clear to those who hear it, the Fs must be greater than or equal to twice the highest frequency that a person can hear.

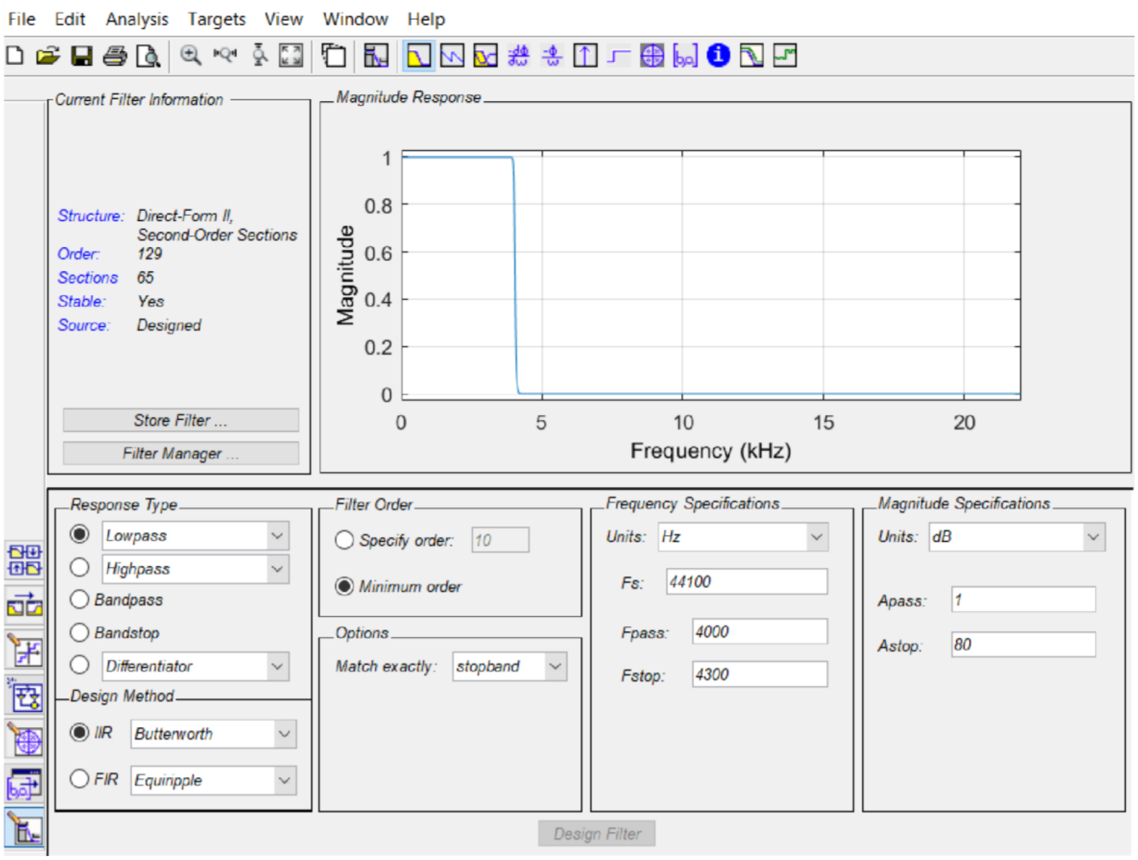
And we will select bit depth (Quantization levels) = 16 bits

In order to be able to determine the amplitude of each frequency so that it is convenient for the ear to differentiate between each amplitude and another.

The more (bit depth or sampling frequency) the number is, the better, but on the other hand, it is more complex and requires more processing operations.

The bit rate= 44100*16=705.6 kbit/s

b-



We select Fpass=4000 & Fstop=4300

Because if we reduce that, there will be unclear letter and high frequency letter may disappear the talk my be not understandable

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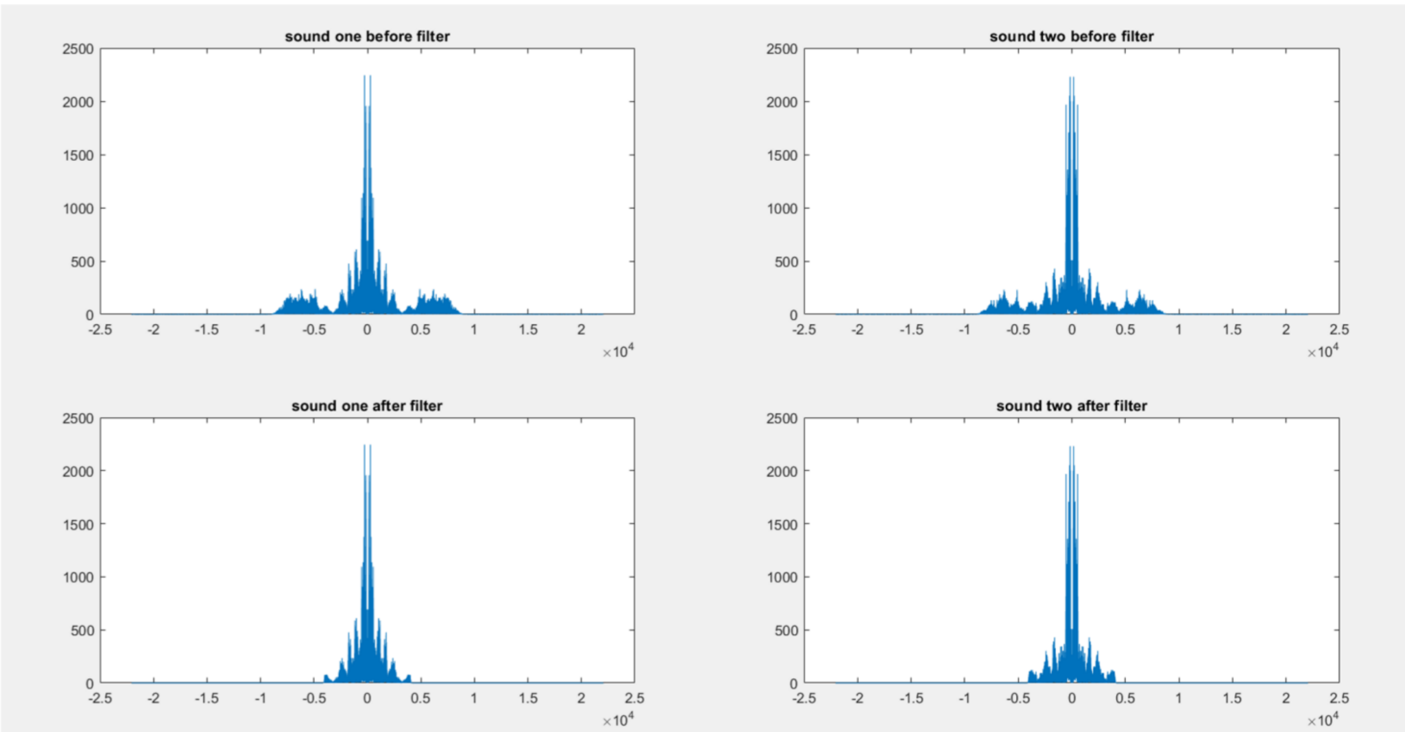
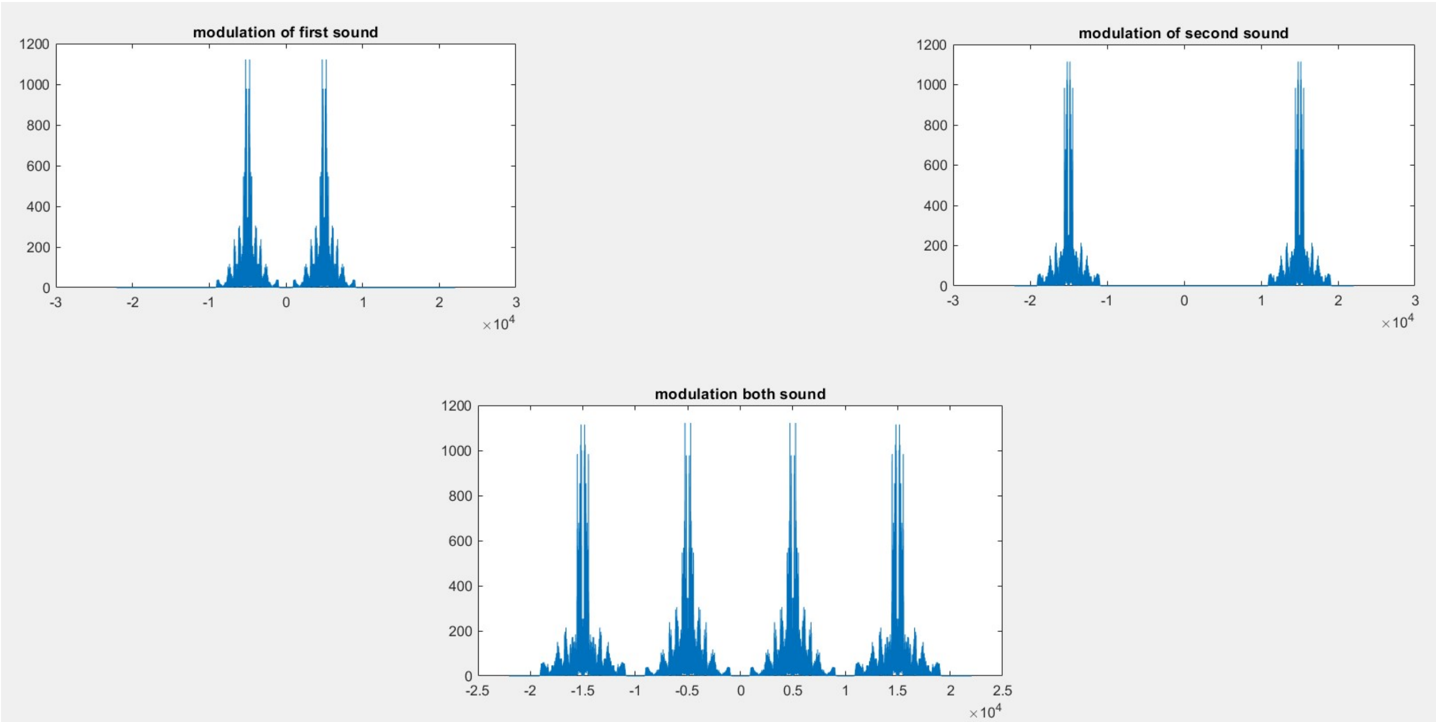
c-



d. You will make a modulation of the first sound at a frequency of (F_a= 5,000 Hz) and a modulation of the second sound at a frequency of (F_b= 15,000 Hz).

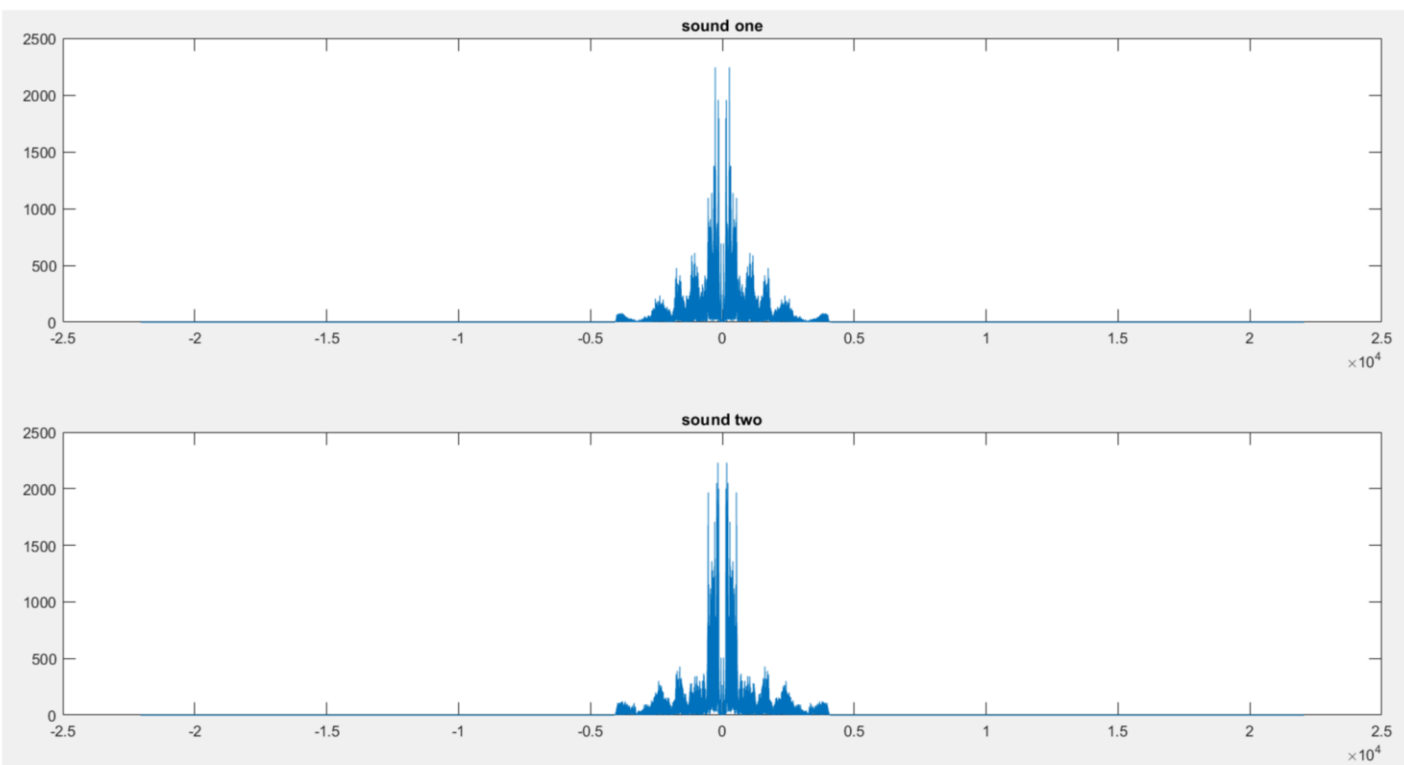
In order for there to be no interference between the two sounds, $F_b - F_a > 2F_{pass}$

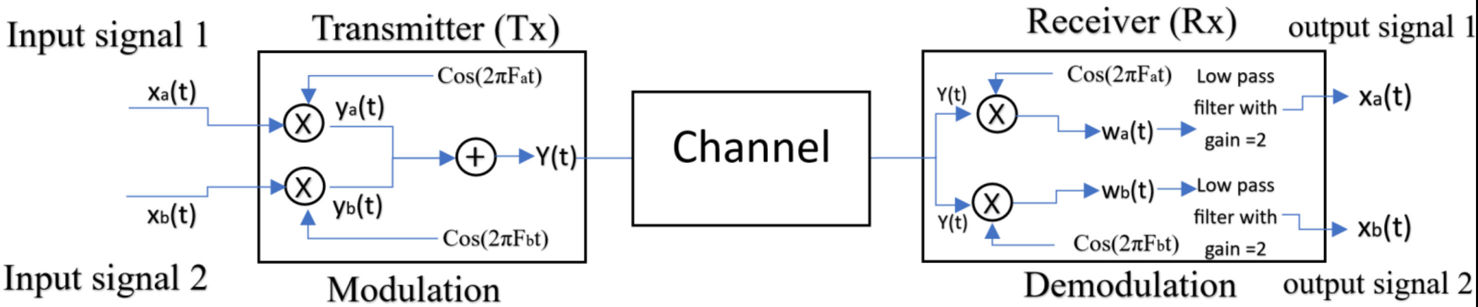
$F_b - F_a$= 10,000 Hz $\quad > \quad$ $2F_{pass}$= 2*4,000=8,000 Hz



The operation in time domain

e-





We will multiply the 2 input signal in sinusoidal 2 carrier

$Ya(t)=Xa(t)*Pa(t)$ $Yb(t)=Xb(t)*Pb(t)$ where :

$p_a(t)=Cos(2\pi F_a t)$ $p_b(t)= Cos(2\pi F_b t)$

- Then we sum the two signals in one signal
- $y(t)= y_a(t) + y_b(t)$
- Then we send Y(t) to channel (assume we have ideal channel)
- When the Y(t) reaches the receiver
- We will multiply Y(t) by the same two (sinusoidal carrier), in order to return each signal to its original bandwidth
- $w_a(t)= y(t)\ Cos(2\pi F_a t)$ $w_b(t)= y(t)\ Cos(2\pi F_b t)$
- Then I will create a L.P filter, because there is the same signal twice on the left and the right,

in addition to the other signal sent with it and the noises exists

- Note: It's more better to do band bass filter first in the receiver to cut off the noise before demodulation
- Note: the filter with gain =2
- let's assume the filter is Hd
- Note: I will use {*} to refer to a convolution operation
- In last step I will convolve the 2 signals with filter in (Time domain)
- $x_a(t)=w_a(t)*Hd(t)$ $x_b(t)=w_b(t)*Hd(t)$

The operation in frequency domain

- In Transmitter side We will convolve the 2 input signal with 2 sinusoidal carrier
- $Y_a(\omega)=\frac{1}{2\pi}Xa(\omega) * Pa(\omega)$ $Y_b(\omega)=\frac{1}{2\pi}X_b(\omega) * P_b(\omega)$
- Where: $P_a(\omega)= \pi\delta(\omega-\omega_a)+\pi\delta(\omega+\omega_a)$
- $P_b(\omega)=\pi\delta(\omega-\omega_b)+\pi\delta(\omega+\omega_b)$
- $Y_a(\omega)=\frac{1}{2}[Xa(\omega-\omega_a)+Xa(\omega+\omega_a)]$
- $Y_b(\omega)=\frac{1}{2}[X_b(\omega-\omega_b)+X_b(\omega+\omega_b)]$

- Then we sum the two signals in one signal

- $Y(\omega)= Y_a(\omega)+Y_b(\omega)$
- $Y(\omega)= \frac{1}{2}[\, X_a(\omega- \omega_a) + X_a(\omega+ \omega_a) + X_b(\omega- \omega_b) + X_b(\omega+ \omega_b)\,]$
- Then we send $Y(\omega)$ to channel
- When the $Y(\omega)$ reaches the receiver
- We will convolve $Y(\omega)$ by the same two (carrier in the transmitter), in order to return each signal to its original bandwidth
- But in this time we will transform the carrier to frequency domain first
- $W_a(\omega)= y(\omega) * P_a(\omega)$ $\quad W_b(\omega)= y(\omega) * P_b(\omega)$
- Where: $P_a(\omega)= \pi\delta(\omega - \omega_a) + \pi\delta(\omega + \omega_a)$
- $Pb(\omega)= \pi\delta(\omega - \omega b)+ \pi\delta(\omega + \omega b)$
- $Y(\omega)=\frac{1}{2}\, [X_a(\omega- \omega_a) + X_a(\omega+ \omega_a) + X_b(\omega- \omega_b) + X_b(\omega+ \omega_b)\,]$
- $Wa(\omega)= \frac{1}{4}\, [\, Xa(\omega) + Xa(\omega+2\omega a) + Xb(\omega- \omega b+ \omega a) + Xb(\omega+ \omega b+ \omega a)] +\frac{1}{4}\, [Xa(\omega- 2\omega a) + Xa(\omega) + Xb(\omega- \omega b-+ \omega a) +Xb(\omega+ \omega b- \omega a)]$
- By doing the same operation for $Wb\ (\omega)$ we will reach to similar expression
- $W_b(\omega)= \frac{1}{4}\, [X_b(\omega) + X_b(\omega+2\omega b) + Xa(\omega - \omega a + \omega b) +$

$X_b(\omega+\omega_b+\omega_a)] + \frac{1}{4}[X_b(\omega-2\omega_b) + X_b(\omega) + X_a(\omega-\omega_a-$

$+\omega_b) + X_a(\omega+\omega_a-\omega_b)]$

- Note: as we see Wa(ω) has the wanted signal which is the first input Xa(ω) but multiplied by ½ and other unwanted signal
- the same Wb(ω) has the wanted signal which is the second input Xb(ω) but multiplied by ½ and other unwanted signal
- In last step we will multiply the two W(ω) signal by the same L.p filter because the two-input signals X1(ω) and X2(ω) has the same bandwidth in this example function in the frequency domain and with gain =2

- $X_a(\omega)=W_a(\omega)\ Hd(\omega) \qquad X_b(\omega)=W_b(\omega)*Hd(\omega)$