Cairo University

Faculty of Engineering

Computer Engineering Department

Third year

COMMUNICATION ENGINEERING

Frequency Division Multiplexing Using DSB-SC AM

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January 2022

Signal Transmission

Requirement 1:

Modulated Signal:

$s(t) = m_1(t) * \cos(w_{c1}t) + m_2(t) * \cos(w_{c2}t) + m_3(t) * \cos(w_{c3}t)$

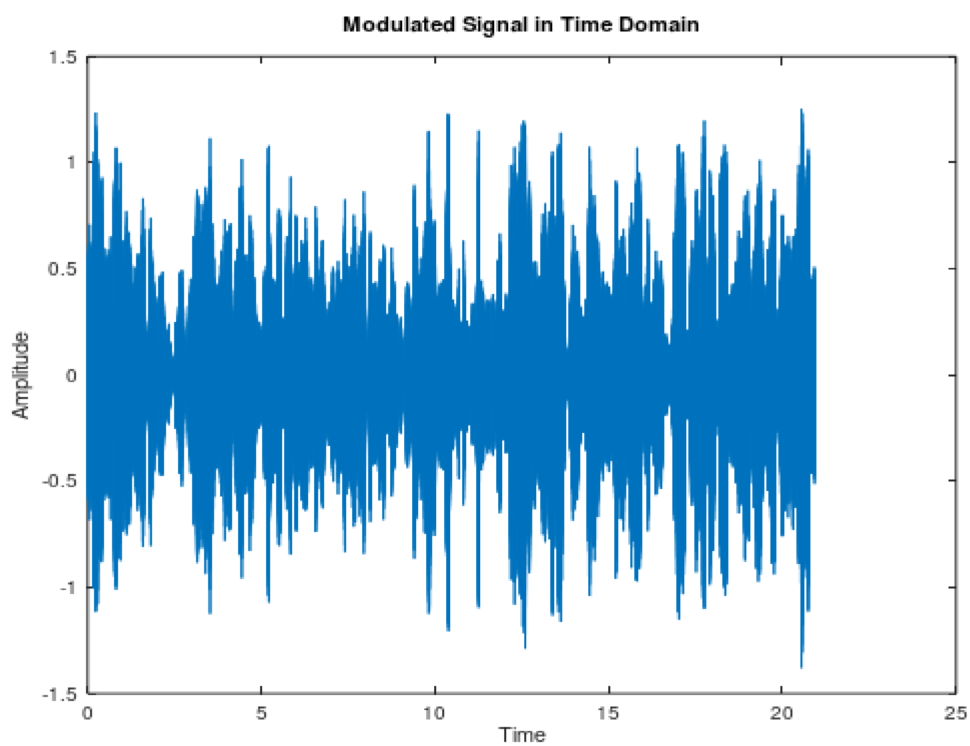
The selected bandwidth is chosen to be the maximum bandwidth of the three spectra.

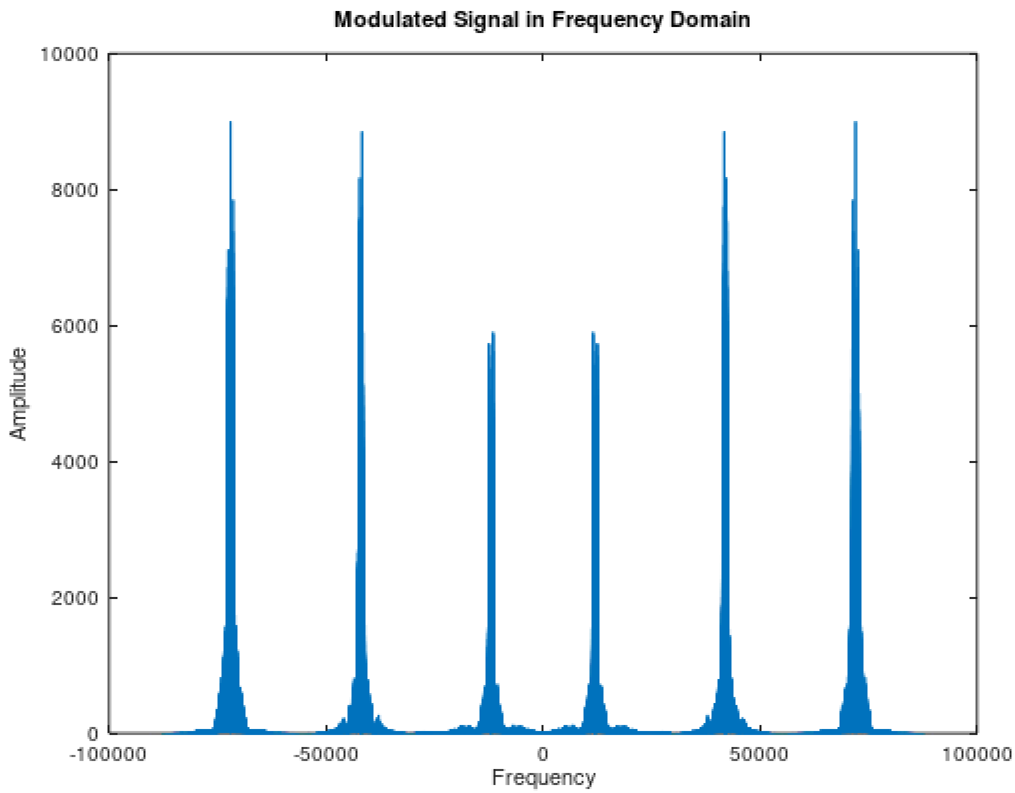
BW = 10 kHz

F_{c1} = 12 kHz, F_{c2} = 42 kHz, F_{c3} = 72 kHz

To avoid attenuation and interference errors in the modulation process the following conditions must hold:

$F_{c1} > BW$, $F_{c2} > F_{c1} + BW$, $F_{c3} > F_{c2} + BW$





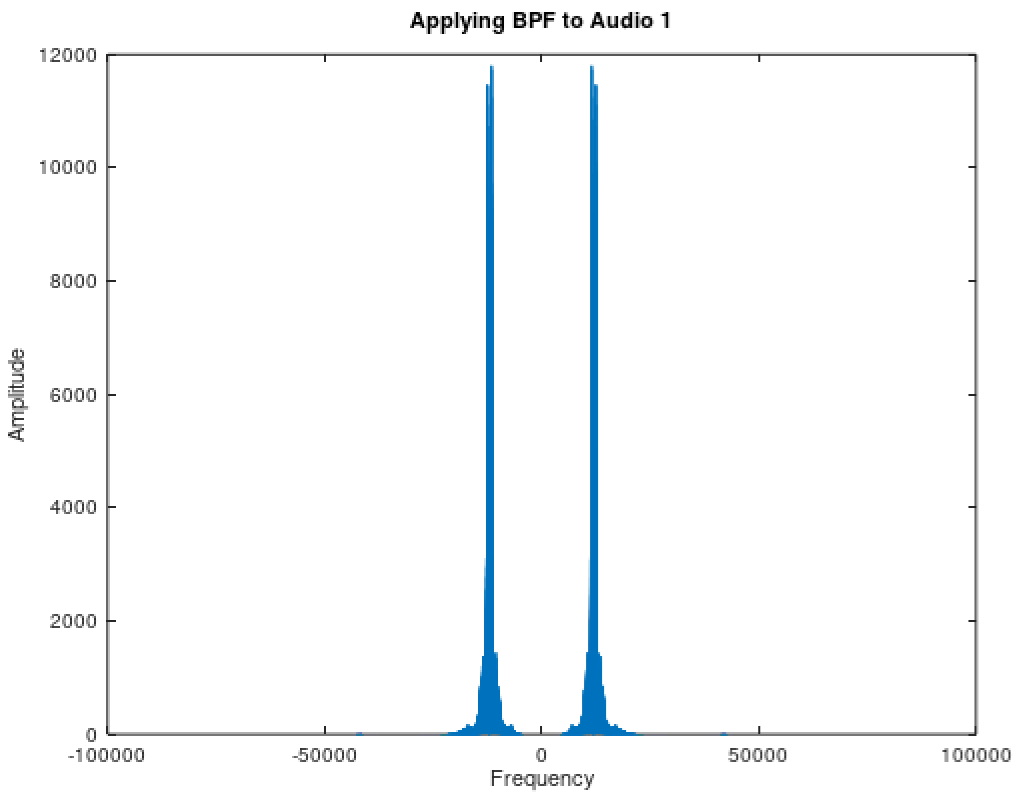
Modulated Signal in Frequency Domain
Amplitude
Frequency
10000
8000
6000
4000
2000
0
-100000
-50000
0
50000
100000

-

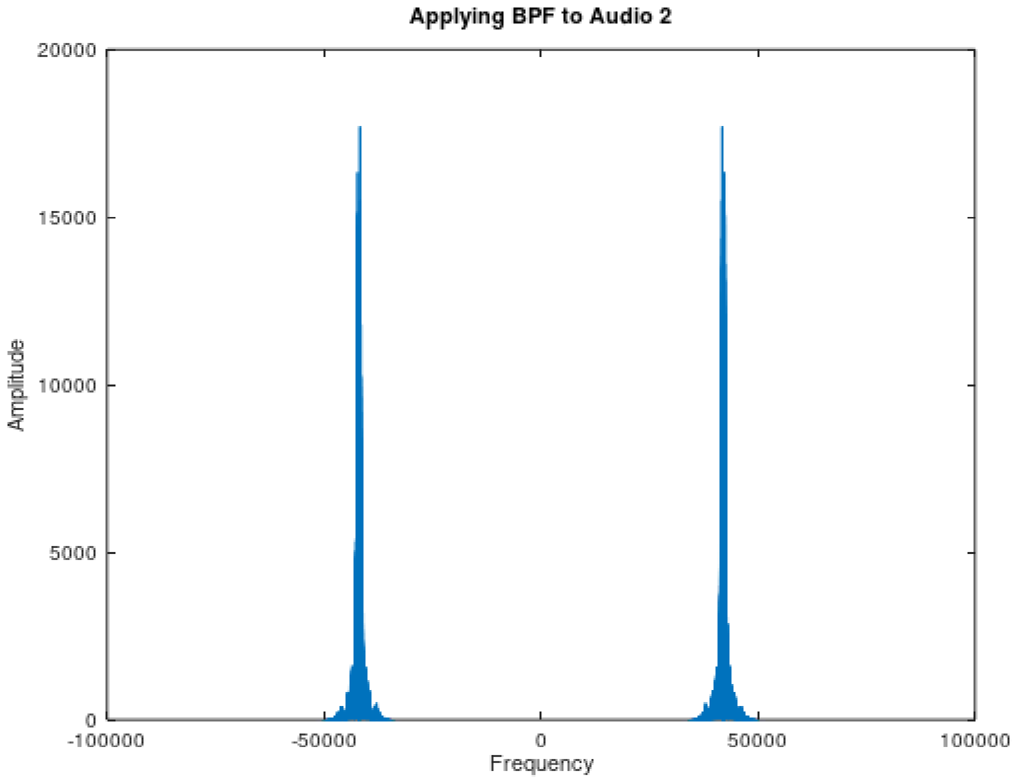
Signal Receiving

Requirement 2:

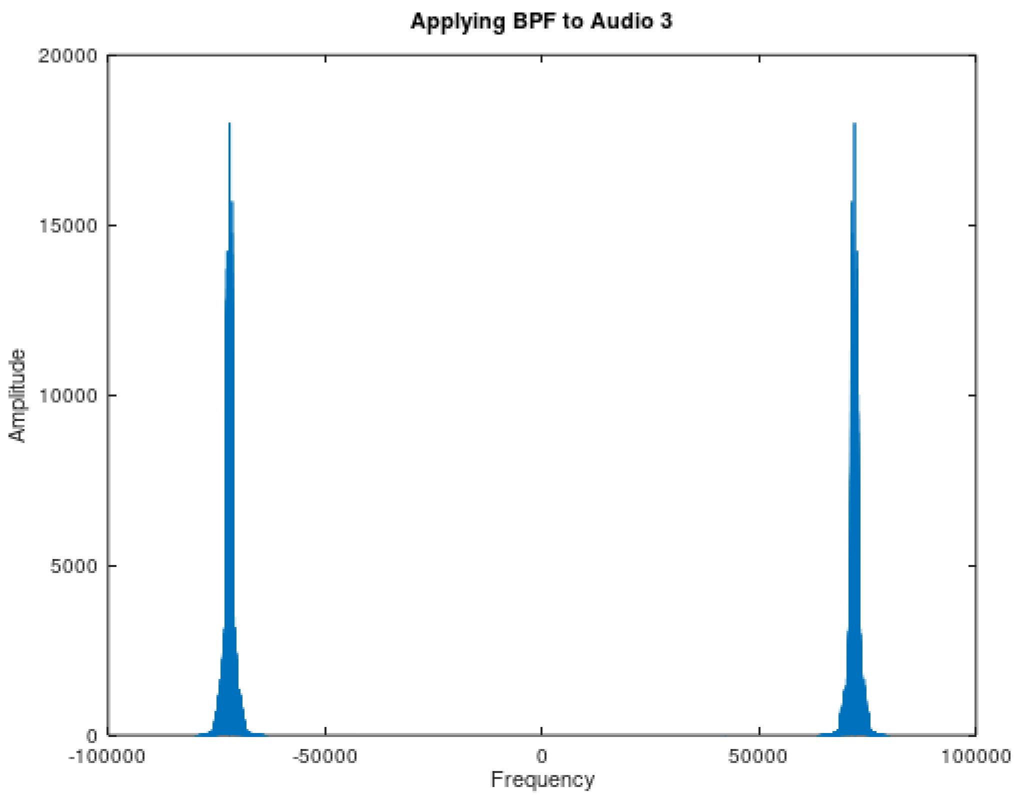
- By applying BPF (which also amplifies the signal) to s(t) whose center frequency equals F_{c1}, the following signal [$s_1(t) = 2 * m_1(t) * \cos(w_{c1}t)$] is obtained.



- By applying BPF (which also amplifies the signal) to s(t) whose center frequency equals F_{c2}, the following signal [$s_2(t) = 2 * m_2(t) * \cos(w_{c2}t)$] is obtained.

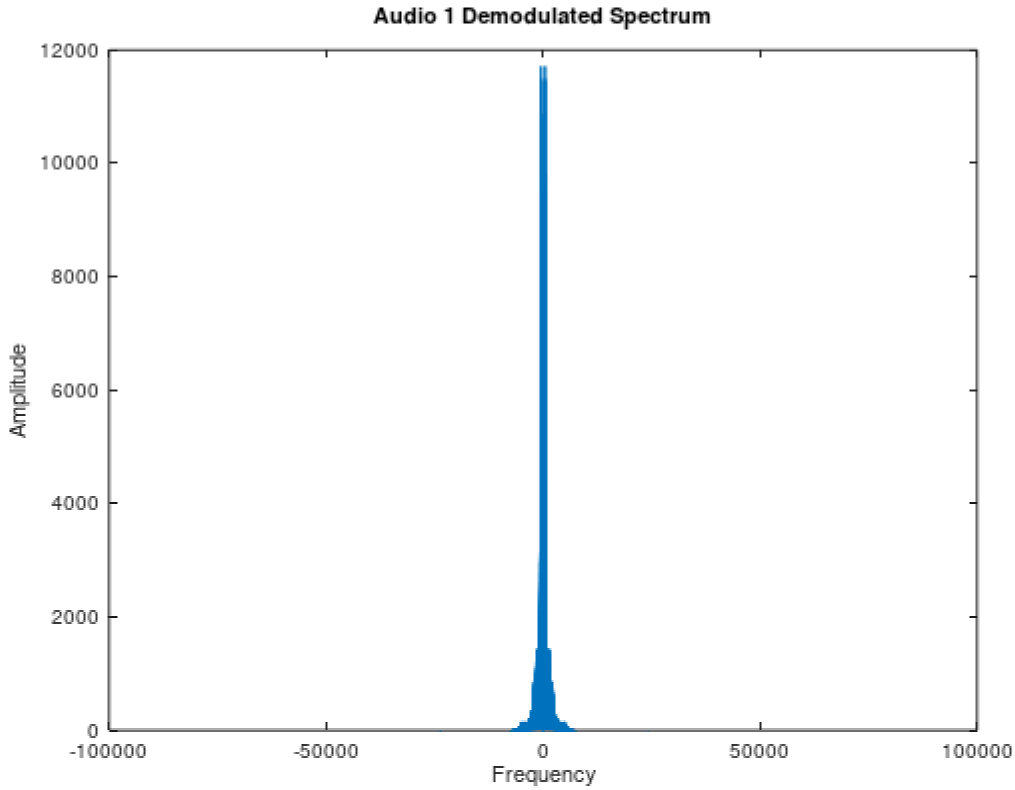


- By applying BPF (which also amplifies the signal) to s(t) whose center frequency equals F_{c3}, the following signal [$s_3(t) = 2 * m_3(t) * \cos(w_{c3}t)$] is obtained.

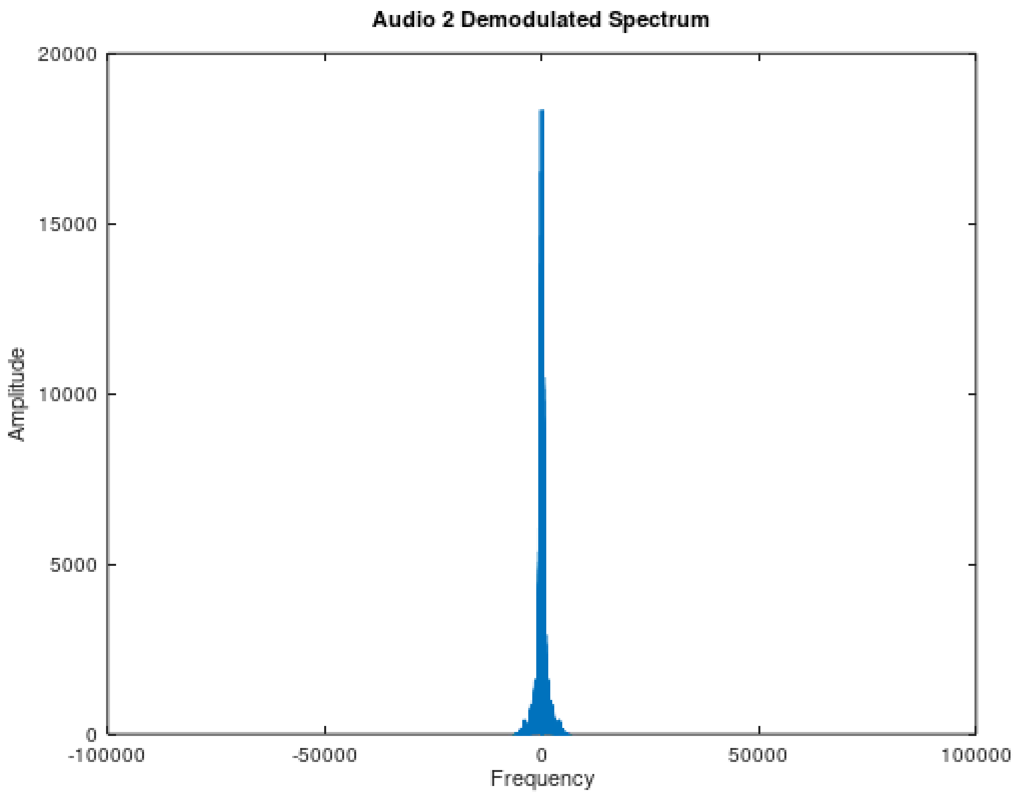


Synchronous Demodulation:

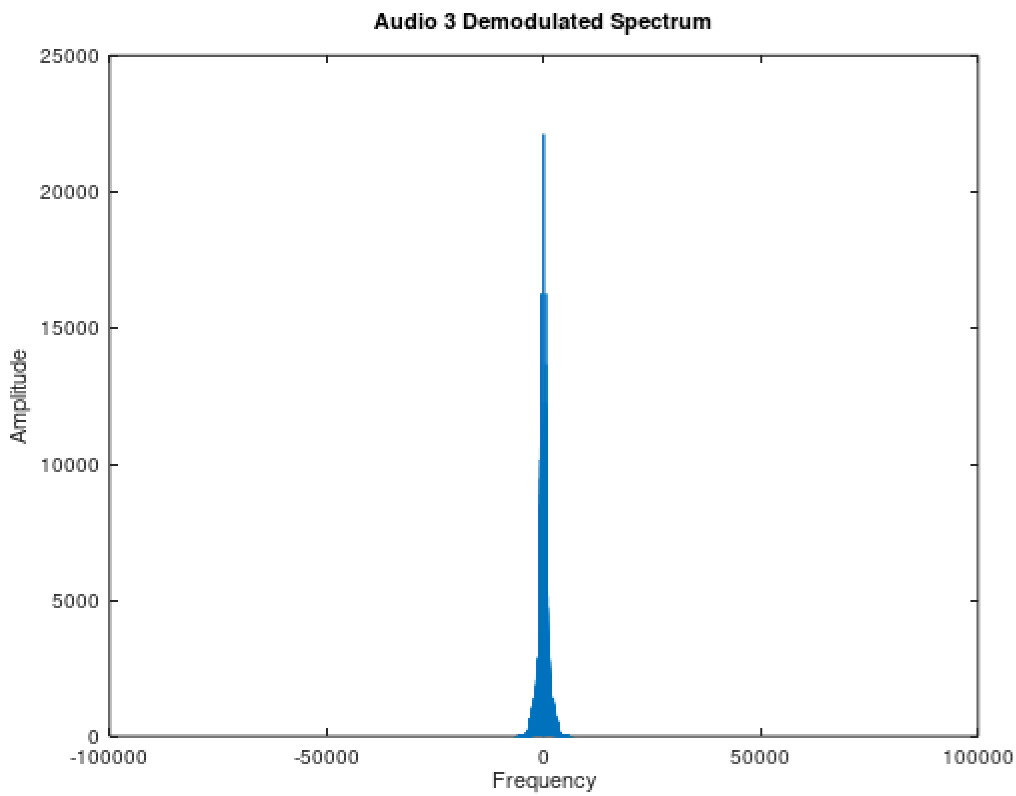
- By multiplying the signal [$s_1(t) = 2 * m_1(t) * \cos(w_{c1}t)$] by [$\cos(w_{c1}t)$], $x_1(t) = 2 * m_1(t) * \cos^2(w_{c1}t) = m_1(t)\,(1 + \cos(2w_{c1}t))$, then by applying a lowpass filter to this signal, the following signal is obtained (which the original signal).



- By multiplying the signal [$s_2(t) = 2 * m_2(t) * \cos(w_{c2}t)$] by [$\cos(w_{c2}t)$], $x_2(t) = 2 * m_2(t) * \cos^2(w_{c2}t) = m_2(t)(1 + \cos(2w_{c2}t))$, then by applying a lowpass filter to this signal, the following signal is obtained (which the original signal).

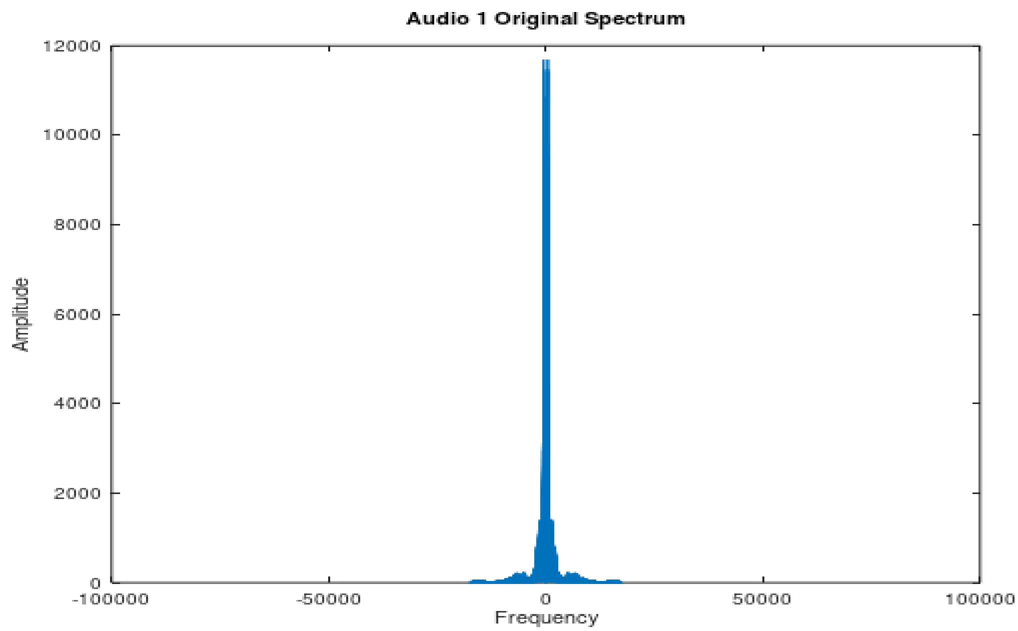


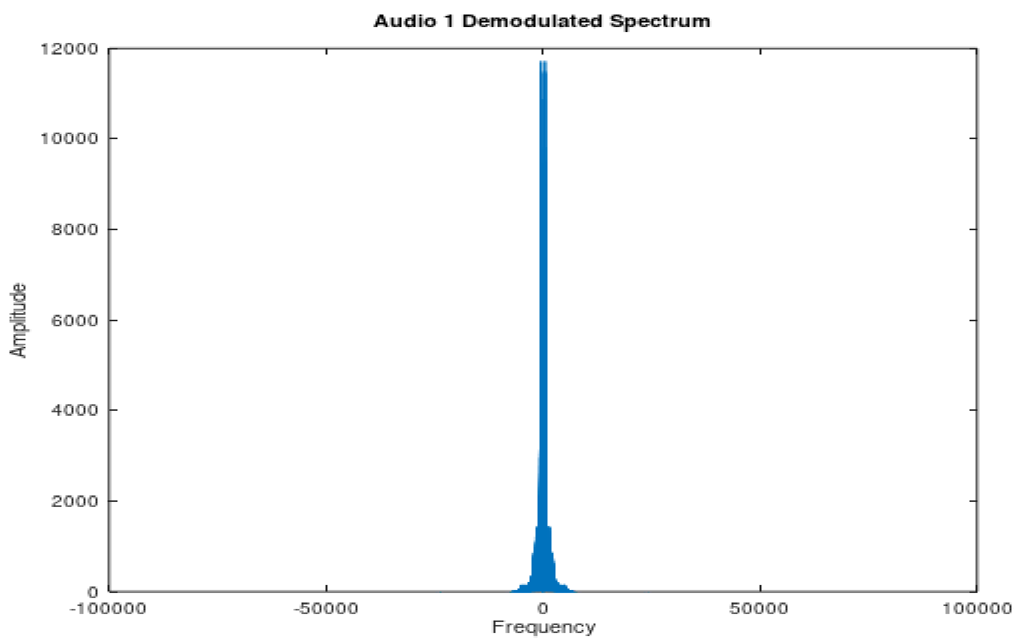
- By multiplying the signal [$s_1(t) = 2 * m_3(t) * \cos(w_{c3}t)$] by [$\cos(w_{c3}t)$], $x_3(t) = 2 * m_3(t) * \cos^2(w_{c3}t) = m_3(t)(1 + \cos(2w_{c3}t))$, then by applying a lowpass filter to this signal, the following signal is obtained (which the original signal).

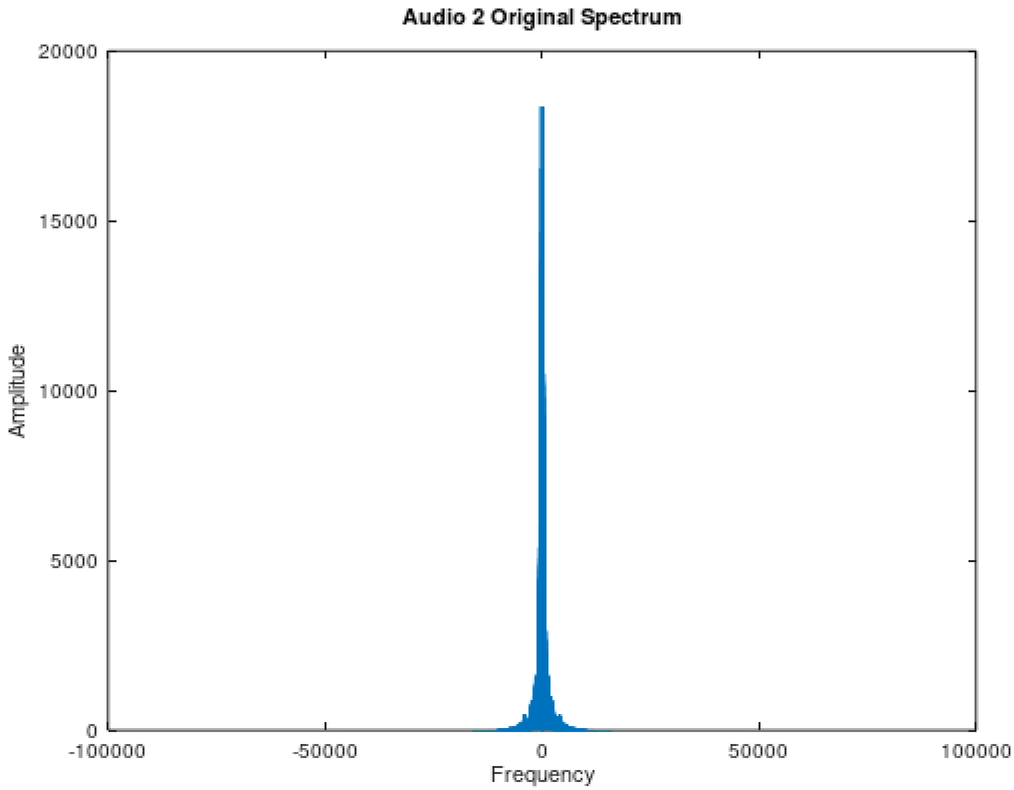


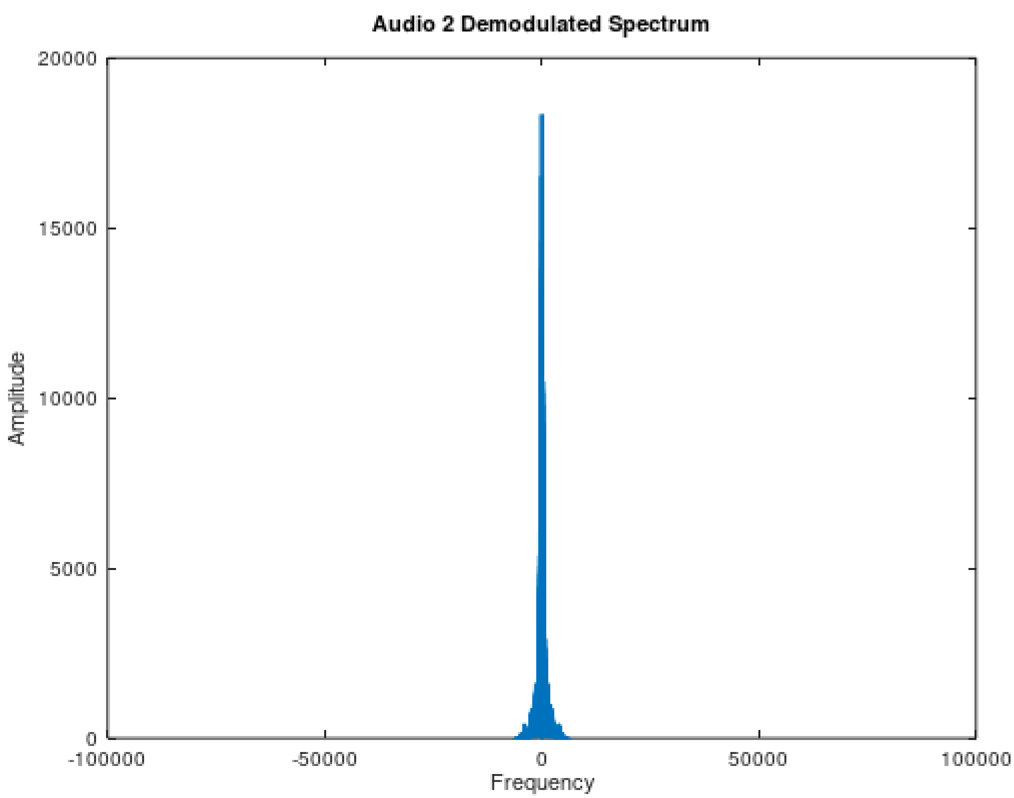
Requirement 3:

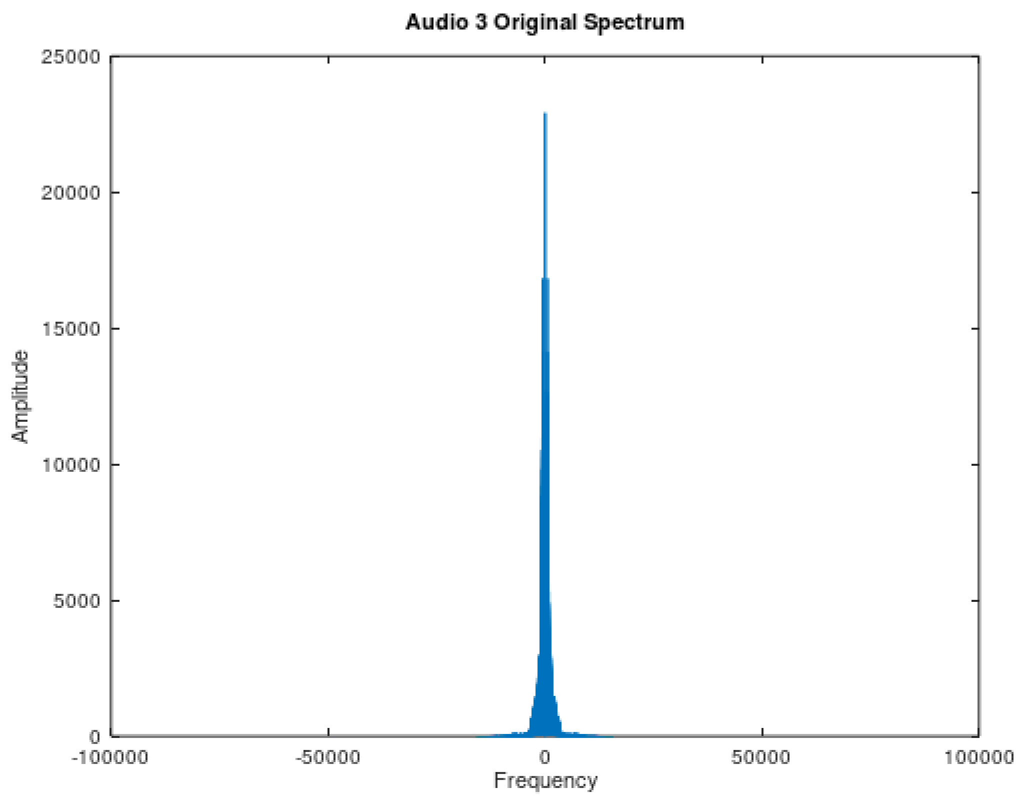
The following figures show the spectra of all audio signals before modulation and after demodulation. There is a small difference in the spectra and the voice of the original and demodulated signals because the used filters (bandpass and lowpass filters) are not ideal.

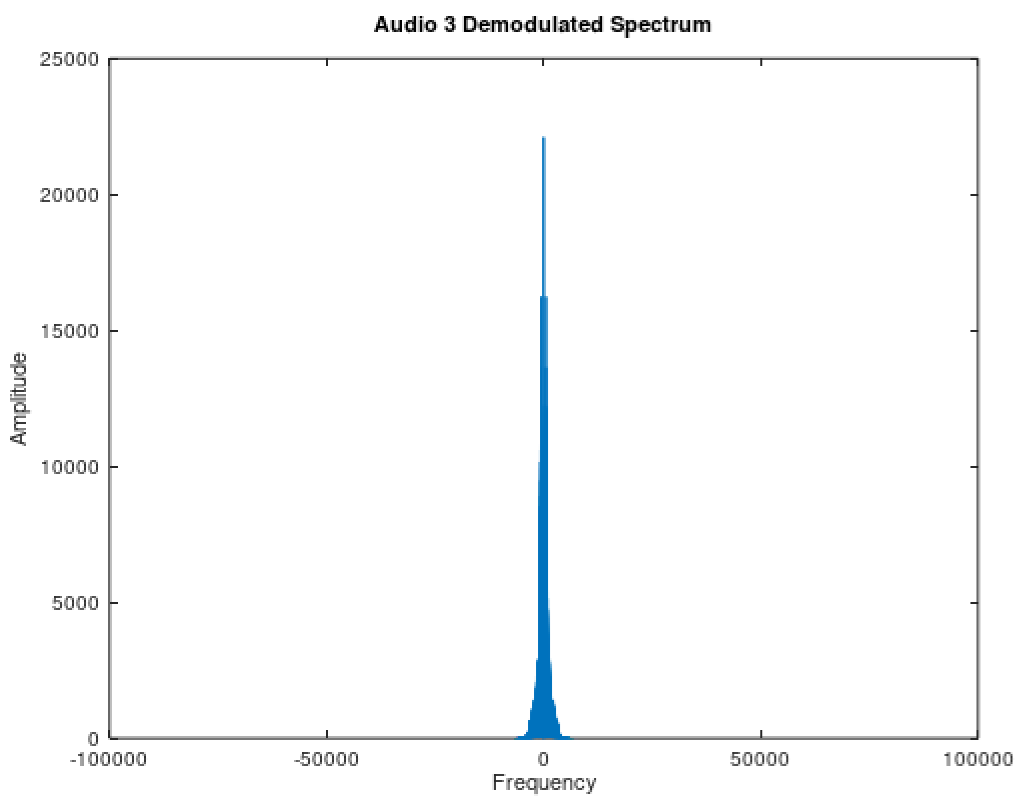





Audio 2 Original Spectrum
20000
15000
10000
5000
0
Amplitude
-100000
-50000
0
50000
100000
Frequency


Audio 2 Demodulated Spectrum
20000
15000
10000
5000
0
Amplitude
-100000
-50000
0
50000
100000
Frequency


Audio 3 Original Spectrum
Amplitude
Frequency
25000
20000
15000
10000
5000
0
-100000
-50000
0
50000
100000


Audio 3 Demodulated Spectrum
Amplitude
Frequency
25000
20000
15000
10000
5000
0
-100000
-50000
0
50000
100000

Requirement 4:

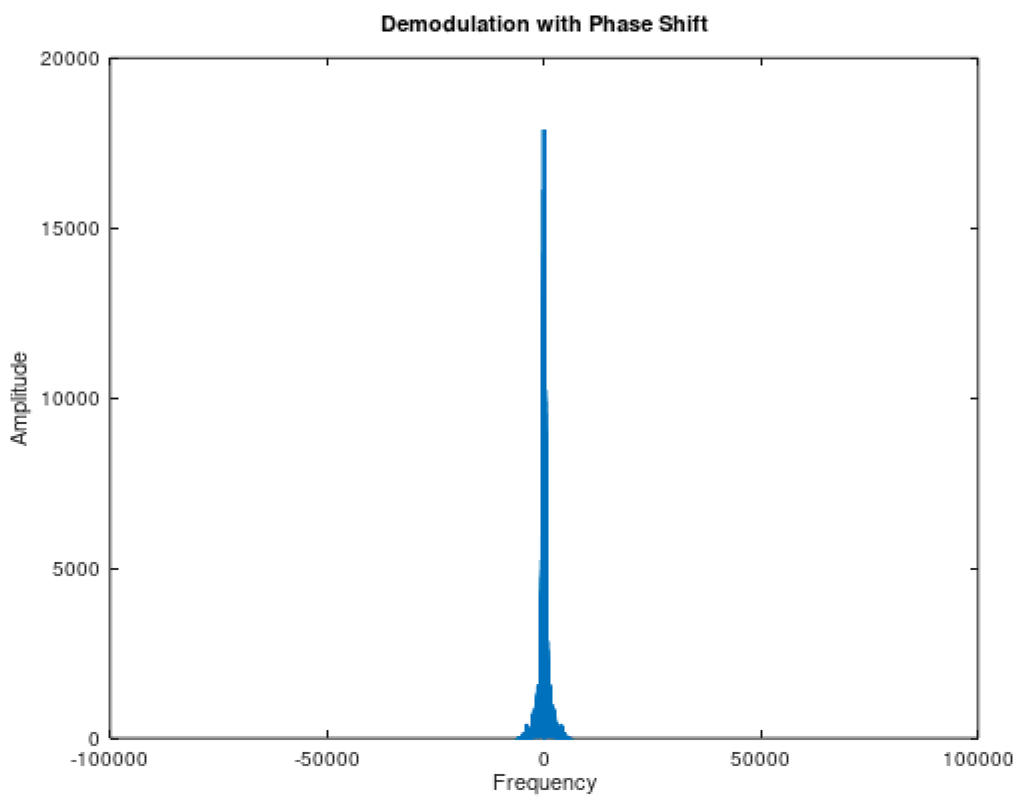
Demodulation of an audio signal with different phase shifts:

After applying the BPF to the signal s(t), demodulation process with different phase shifts relative to the oscillator of the modulator is done.

$x(t) = 2 * m_2(t) * \cos(w_{c2}t) * \cos(w_{c2}t + phi)$

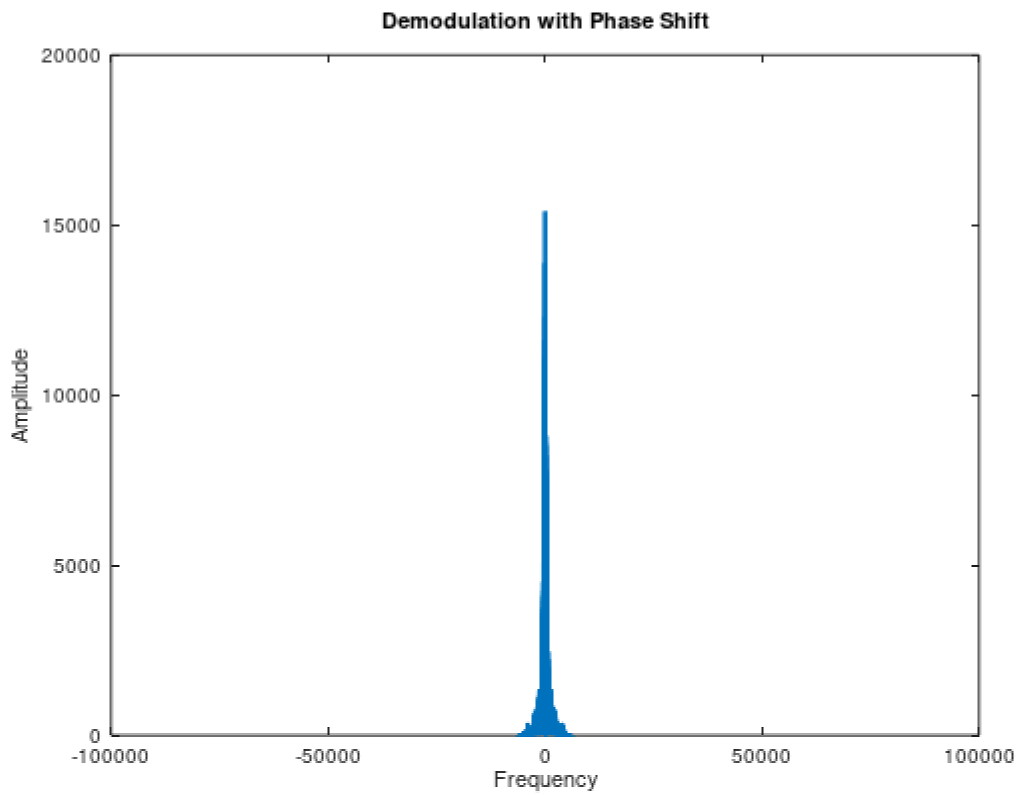
By applying LPF: $x(t) = m_2(t) * \cos(phi)$, which causes attenuation to the signal according to the value of phi (when the value of phi increases, the amplitude of the spectrum decreases). The phase shift may cause attenuation of the output signal without causing distortion as long as it is constant.

- Phase shift (phi) = 10 degrees



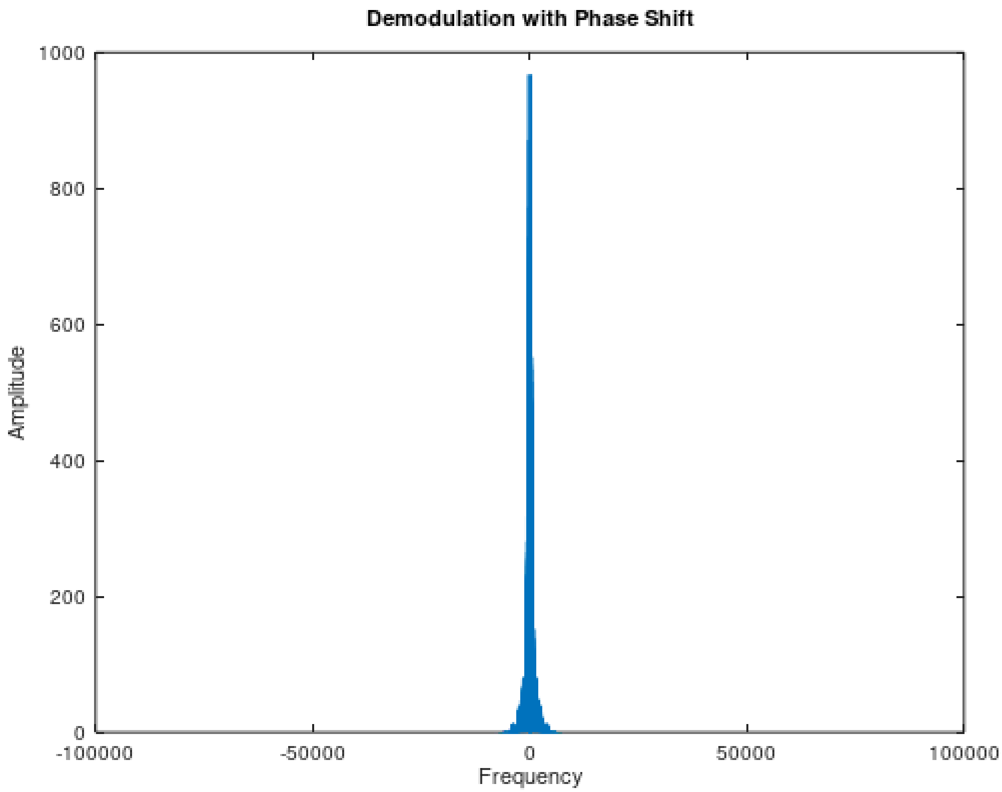
- Phase shift = 30 degrees

Note that the amplitude of the spectrum is decreased by a factor greater than the factor of the previous case (because cos(30) < cos(10)).



- Phase shift = 90 degrees

Note that in the ideal case, the value of the signal spectrum should be zero (cos(90) = 0), but due to the non-ideality of the used filters, the spectrum is not completely diminished. However, its amplitude is decreased by a very large scale (from 17,000 to less than 1,000).



Requirement 5:

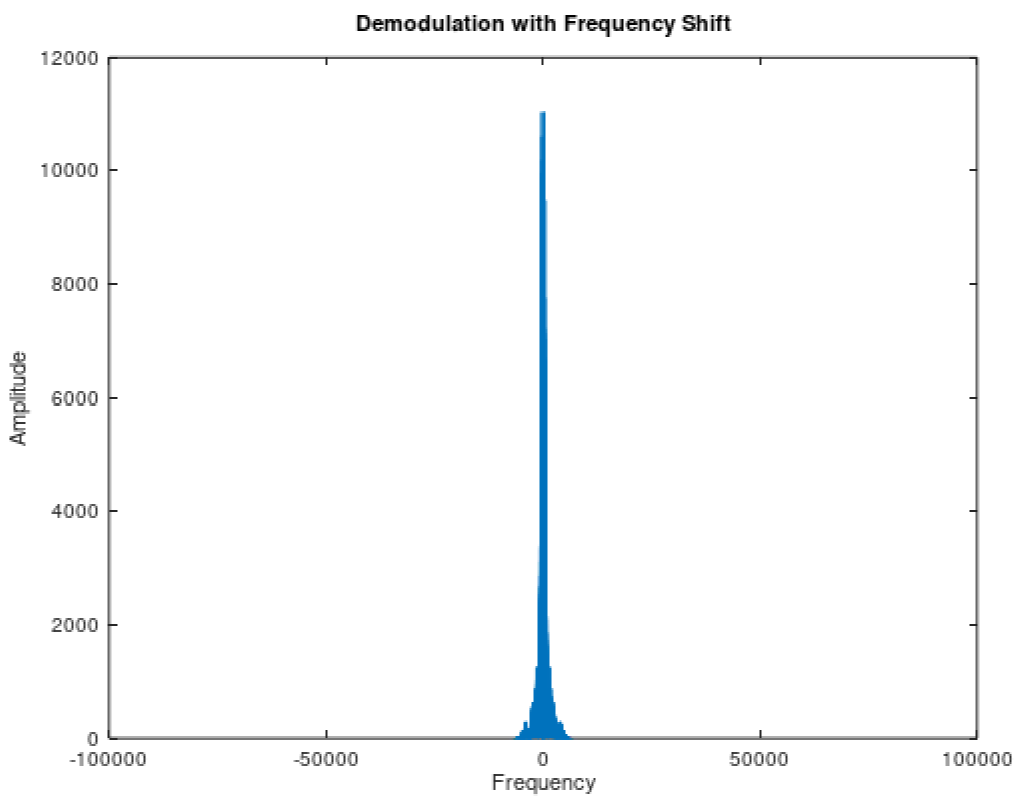
Demodulation of an audio signal with different frequency shifts:

After applying the BPF to the signal s(t), demodulation process with different frequency shifts relative to the oscillator of the modulator is done.

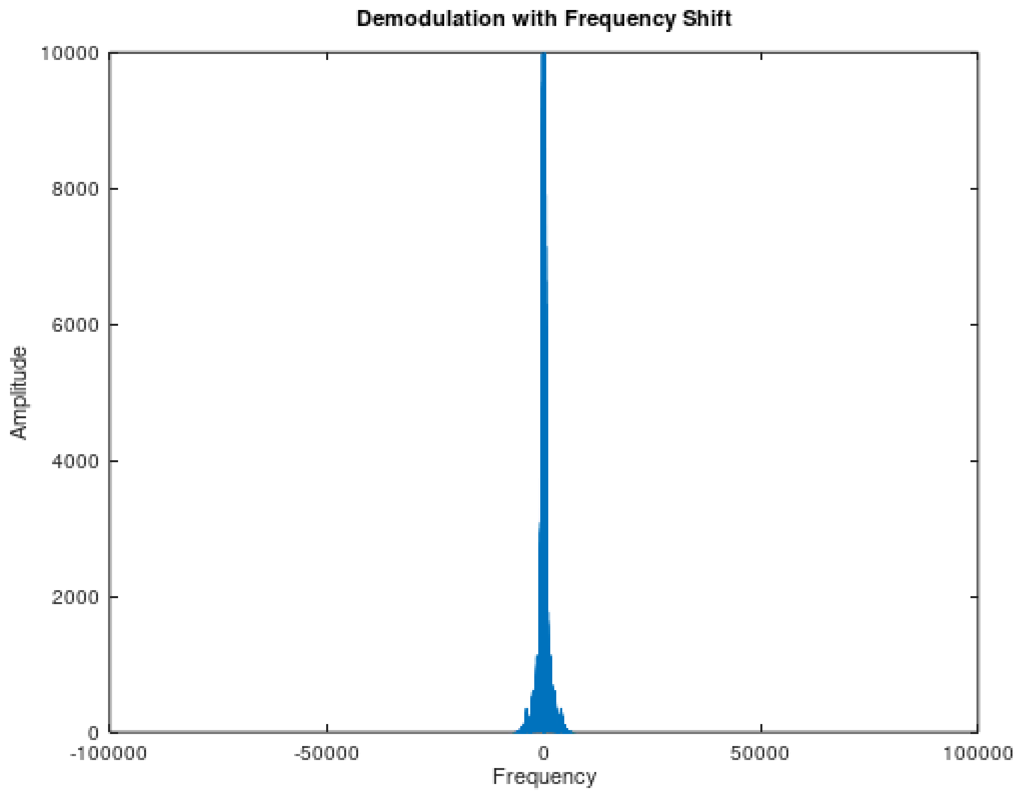
$x(t) = 2 * m_2(t) * \cos(w_{c2}t) * \cos((w_{c2} + w_0)t)$

By applying LPF: $x(t) = m_2(t) * \cos(w_0 t)$. The output is multiplied by a low frequency sinusoid, this causes attenuation and distortion of the output signal.

- Frequency shift = 2 Hz



- Frequency shift = 10 Hz



Requirement 6:

The audio files obtained after demodulation in case of phase shifts in demodulator oscillator:

Phi = 10 degrees => Almost the same audio signal

Phi = 30 degrees => The audio signal was clearly attenuated

Phi = 90 degrees => No voice was heard

The audio signals obtained after demodulation are very distorted in both cases in case of frequency shifts in the demodulator oscillator.

The DSB-SC AM requires exact synchronization between the local carrier at the receiver and the incoming carrier in the received modulated signal. Serious problems happen due to frequency shift and phase shift. Frequency shifts causes high distortions to the audio signals, while phase shifts cause attenuation to the audio signals. The needed synchronization technique makes the receiver more complex and expensive. So, other AM methodologies are suitable to many systems such: DSB-LC.