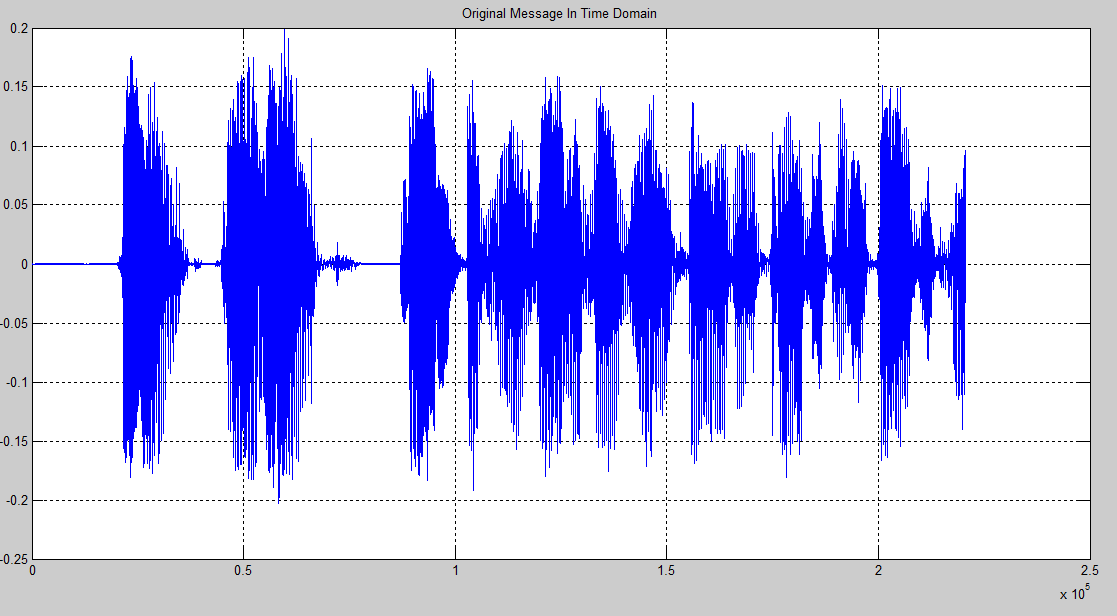
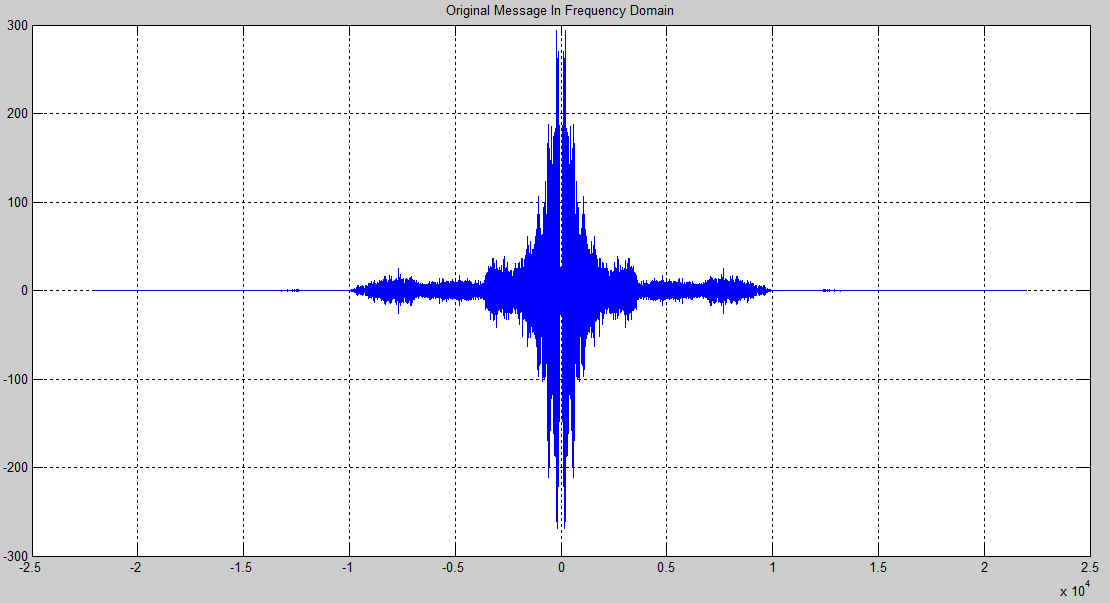
1 EXPERIMENT ONE: DOUBLE SIDEBAND MODULATION

Requirement 1:



Time domain representation of the original message.

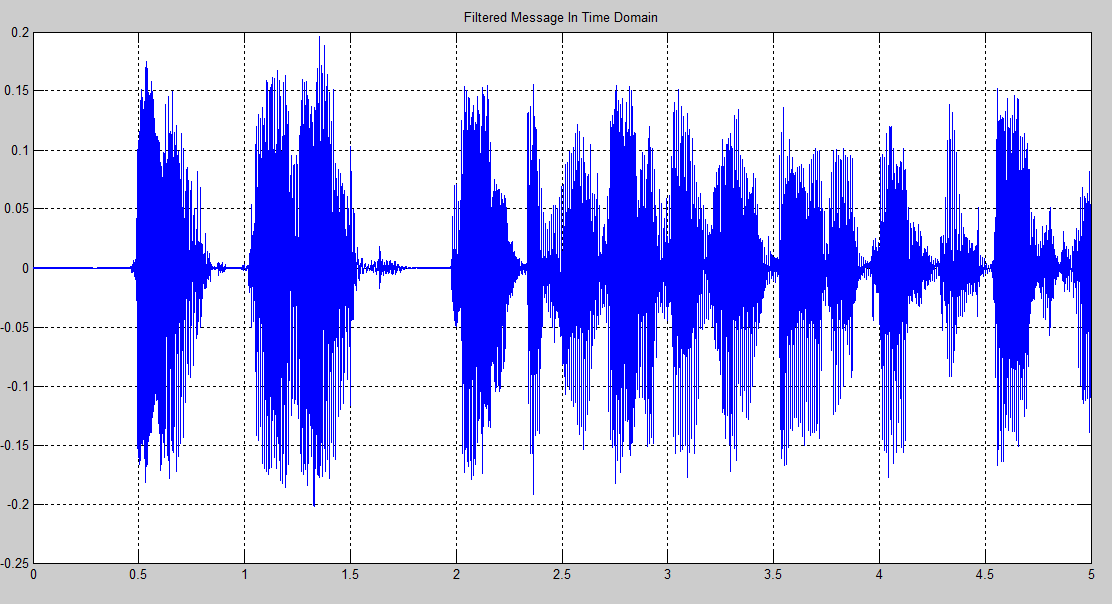


Frequency domain representation of the original message.

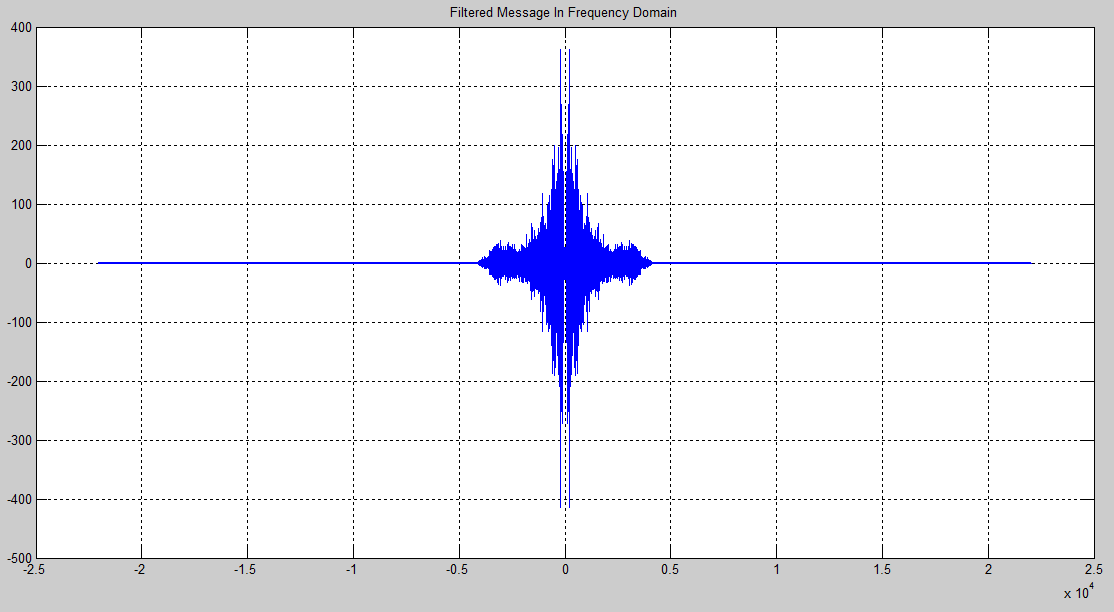
We just deal with the first 5 seconds of the audio file to save memory.

Audio file has its own sampling frequency so we -later- resample the message according to the required sampling frequency (500KHz).

Requirement 2,3&4:



Time domain representation of the filtered message.



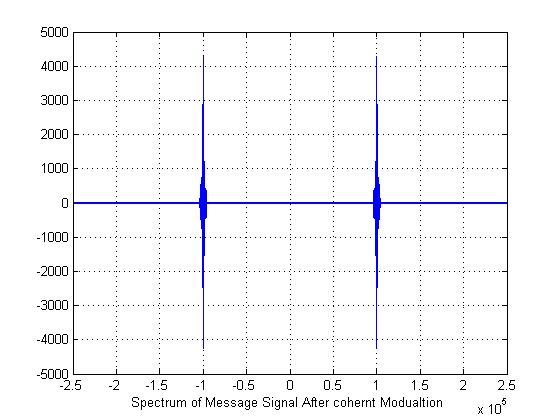
Frequency domain representation of the filtered message.

MSE = measerr (m,m\_fil). MSE equals 0.0032 it's a very small signal, also we play the message and its filtered version again and there was no significant difference.

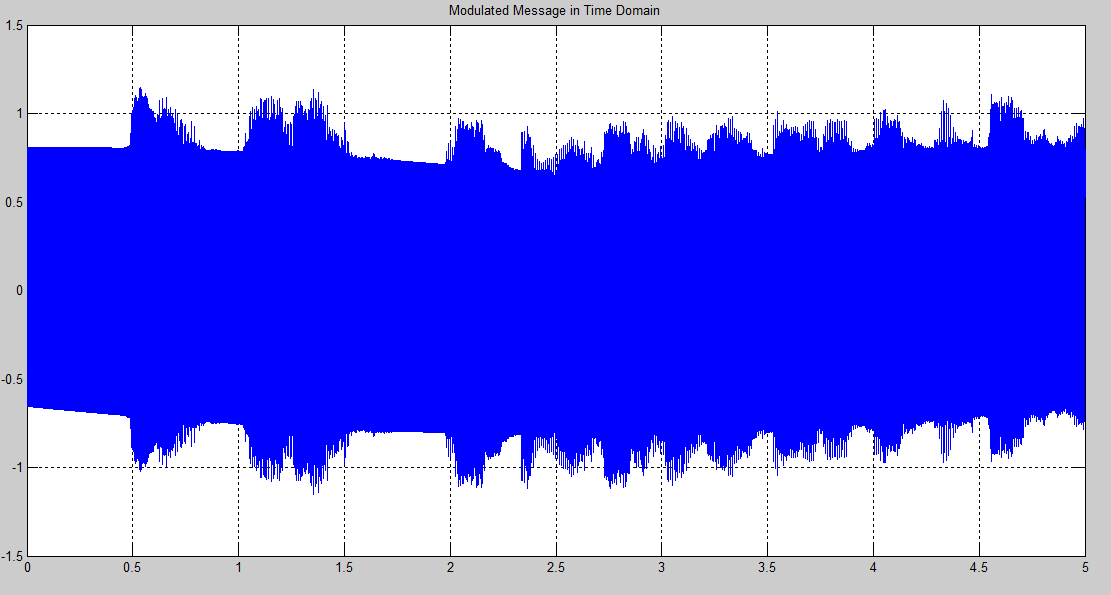
Requirement 5:

DSB-SC modulation:

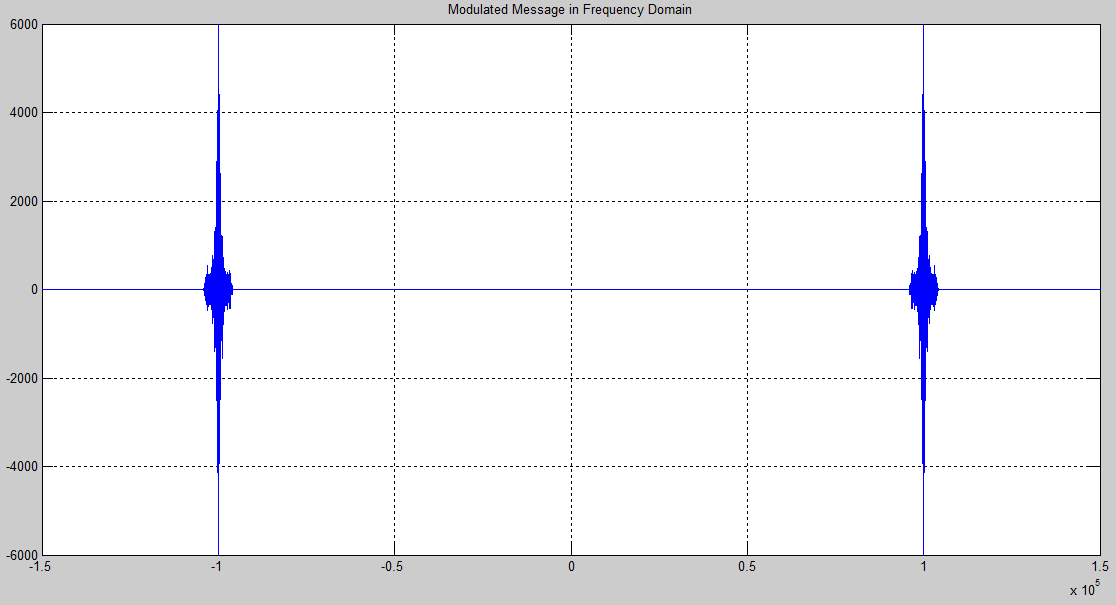
We transmitted the massage to the carrier frequency of 1MHZ by multiplying the massage with the carrier and it is spectrum as follows:



DSB-TC modulation:



Time domain representation of the modulated carrier.

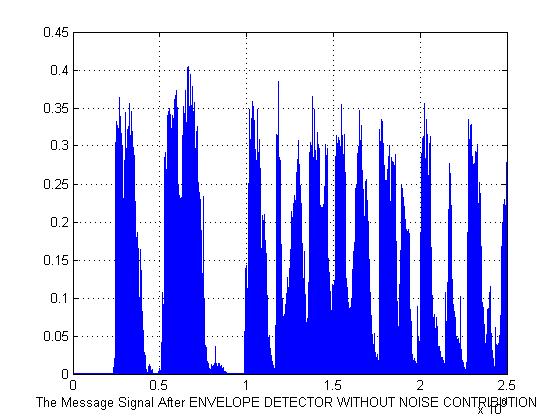


Message after modulation

Modulating message is the filtered message **plus** a dc bias which equals twice the absolute of lowest value contained in the message, to ensure that we can use an envelope detector for demodulation. But before that we resampled the message to the desired sampling frequency.

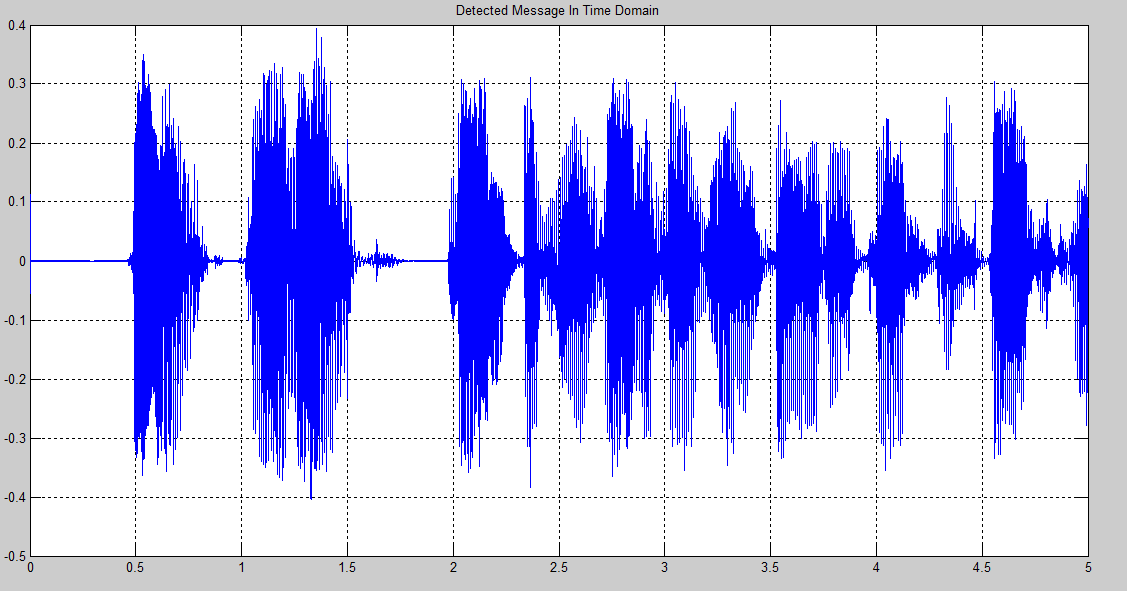
Requirement 6&7:

DSB-SC demodulation using envelope detector:

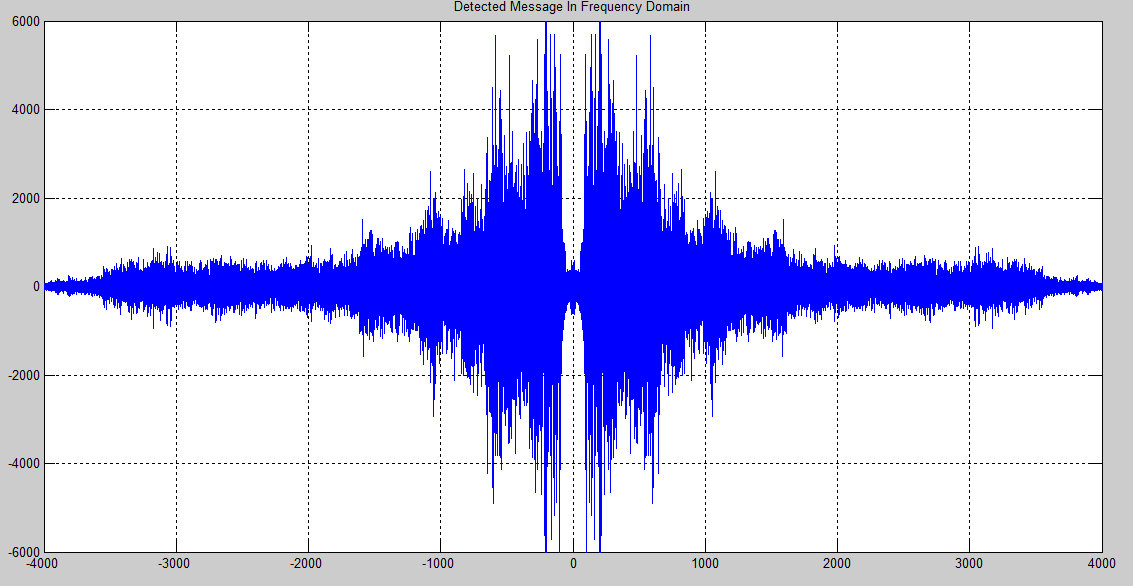


We use the envelope detector to demodulate the signal and take the positive value only of the modulated massage .

DSB-TC demodulation using envelope detector:



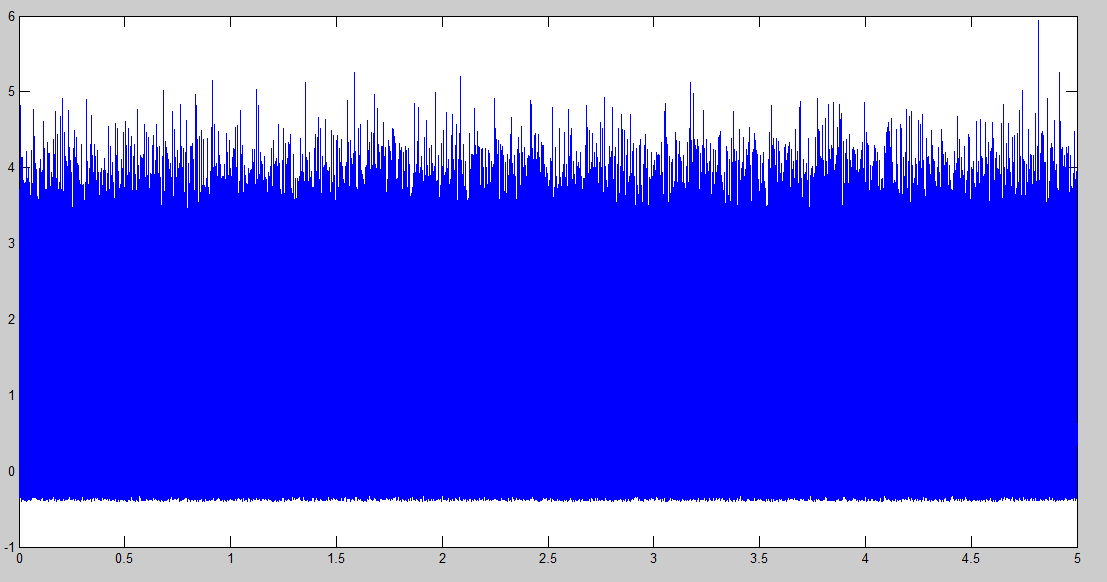
Detection of the transmitted message-in time domain- using envelope detector.



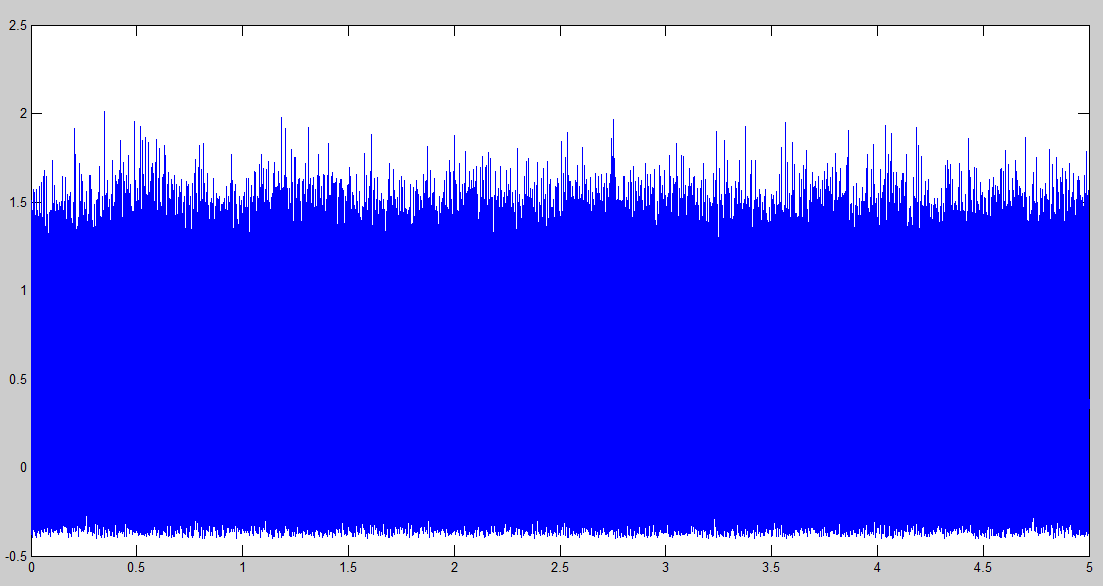
Detection of the transmitted message-in frequency domain- using envelope detector.

We use an envelope detector to demodulate the transmitted message and block the dc bias to extract the transmitted message. After demodulation we calculate error between the transmitted and the detected which equals 0.0015, and playing back both messages prove that error is very small.

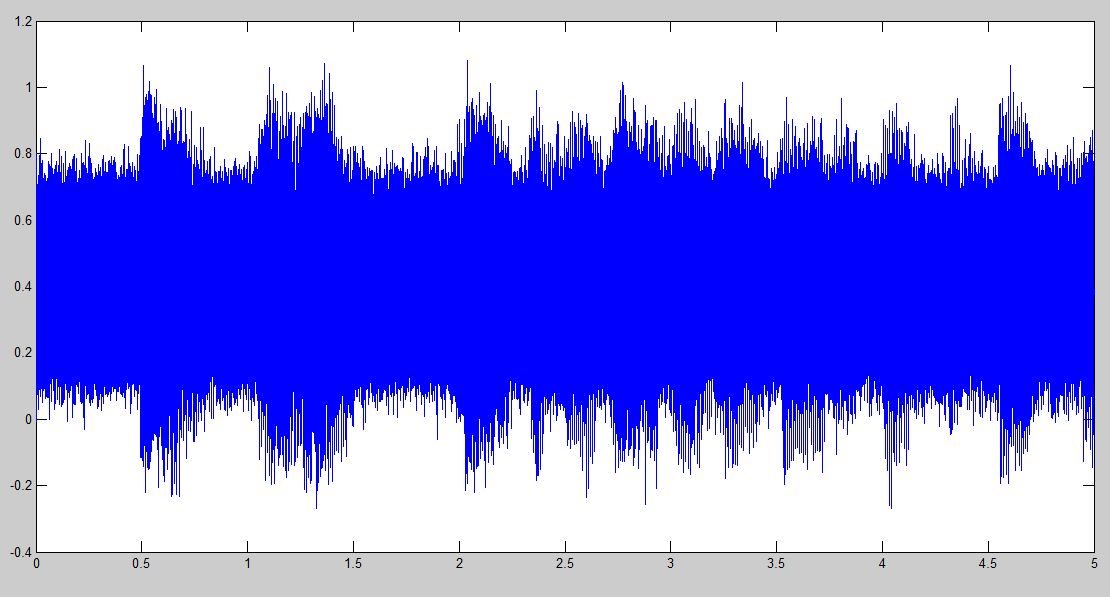
Requirement 8:



Signal plus noise at SNR=0dB in time domain.



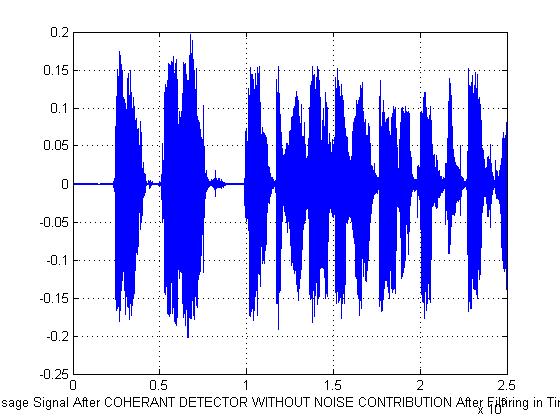
Signal plus noise at SNR=10dB in time domain.

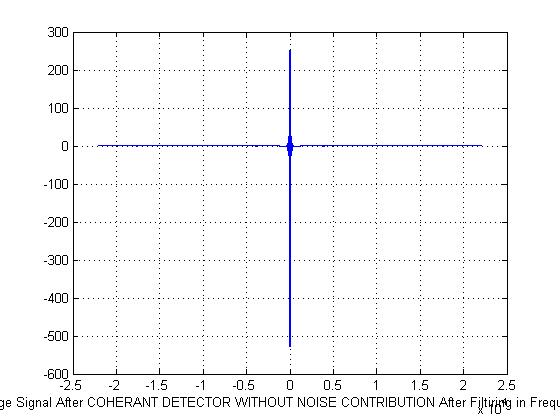


Signal plus noise at SNR=20dB in time domain.

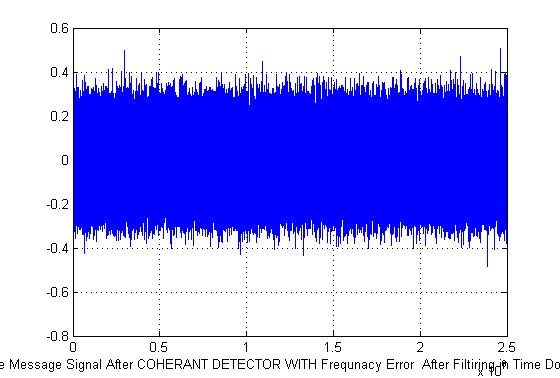
After taking the effect of noise in demodulation and play the sound back we find a lot of disturbance which decreases as long as SNR increases. So at SNR=0dB you cannot understand or hear anything just disturbance, but as SNR increases you can barely recognize the original voice.

Requirement 9:



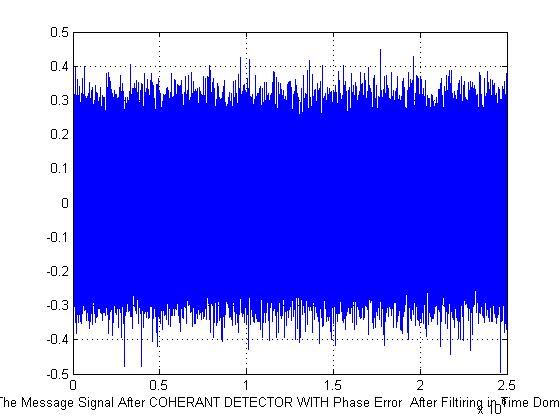


Requirement10:



Error =0.027086155317075

Requirement 11:



Error=0.027077763516222

Requirement 12:

Efficiency equals the ratio of the message power to the total transmitted power

Efficiency = half of message power / half of message power + carrier power

DSB-SC efficiency:

In DSB-SC we don't send the carrier with the message. So power efficiency=100%.

DSB-TC efficiency:

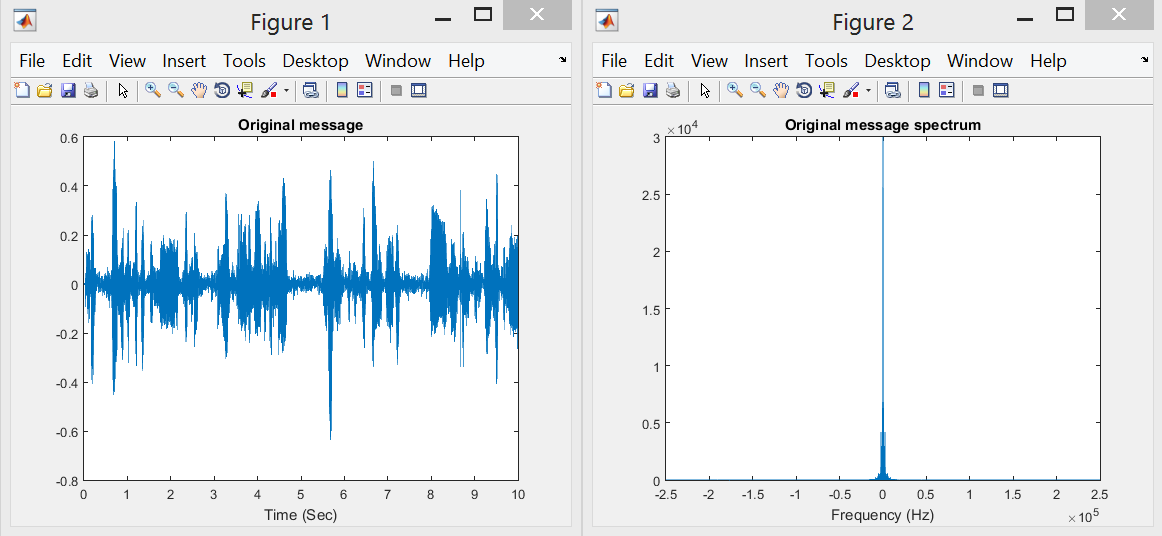
Message power = variance of the message so efficiency = var(m)^2 / var(m)^2 + A^2

Efficiency = 0.9099%

2 EXPERIMENT TWO: SINGLE SIDEBAND MODULATION

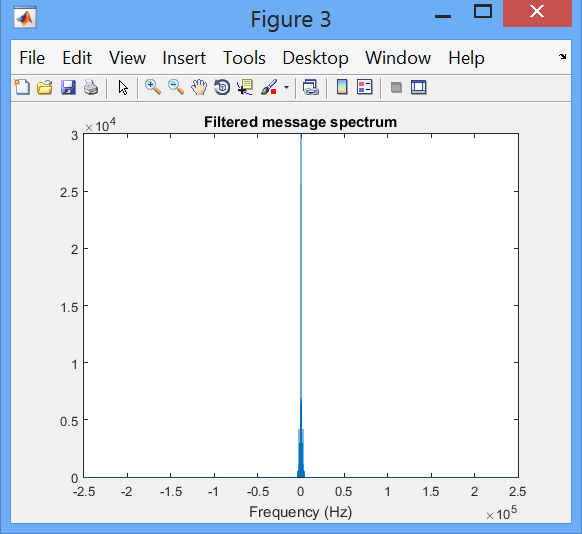
Requirement 1:

I took the absolute of the signal to get the effective spectrum (which is associated with power).



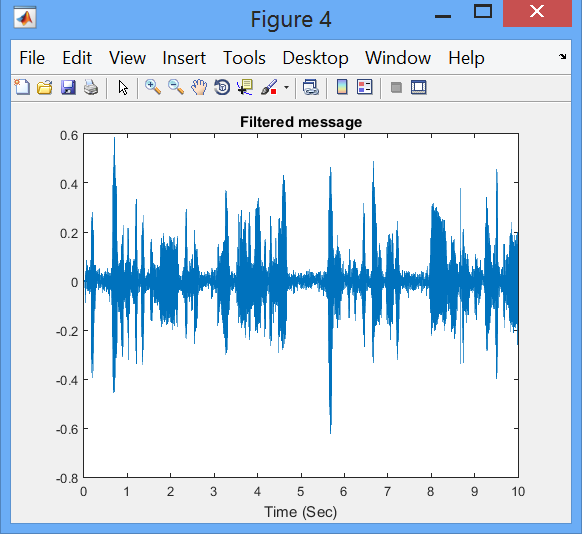
Requirement 2:

Ideal filtering message at 4000 KHz yields:



Requirement 3:

Here is the filtered message in time domain:



Requirement 4:

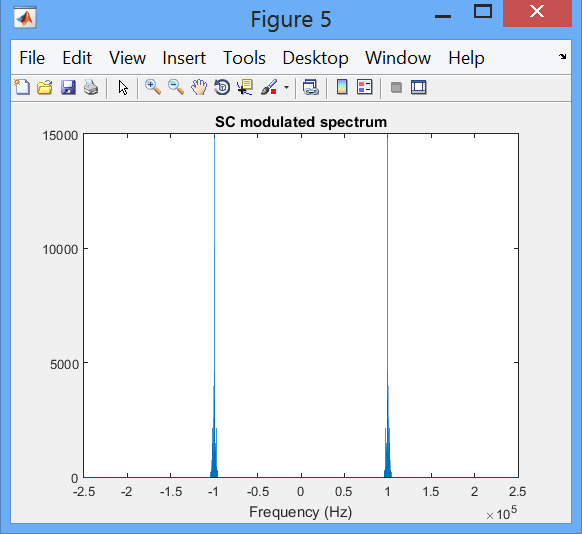
That's the MSE error which is small, as most human voice frequencies lie within 4 kHz.



The filtered message sounds very close to the original

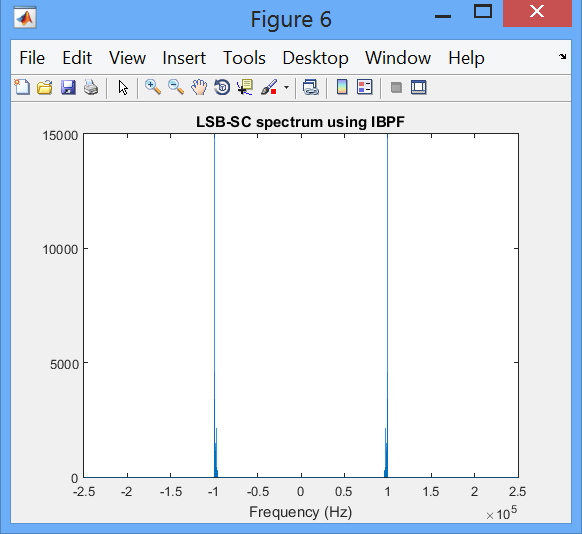
Requirement 5:

Here is the modulated spectrum, I had to multiply the 100 KHz in a scaling factor = 10 because every spacing in the f vector actually = 0.1 Hz in our sense.

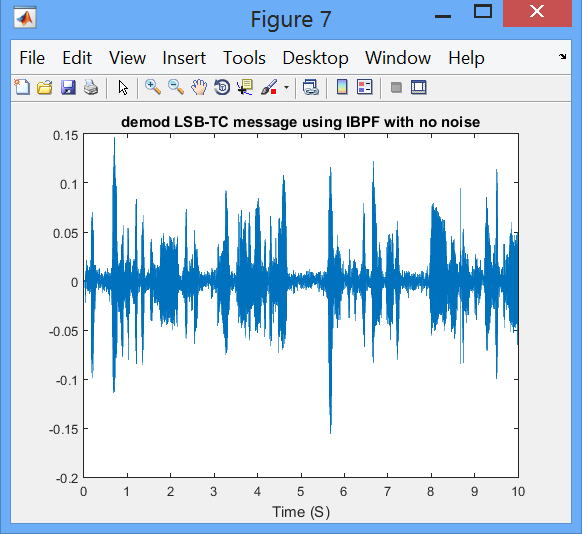


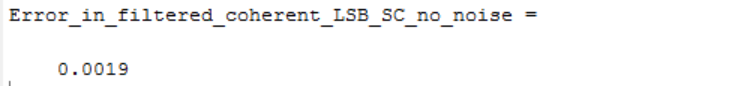
Requirement 6:

Filtering out USB yields:



Requirement 7:



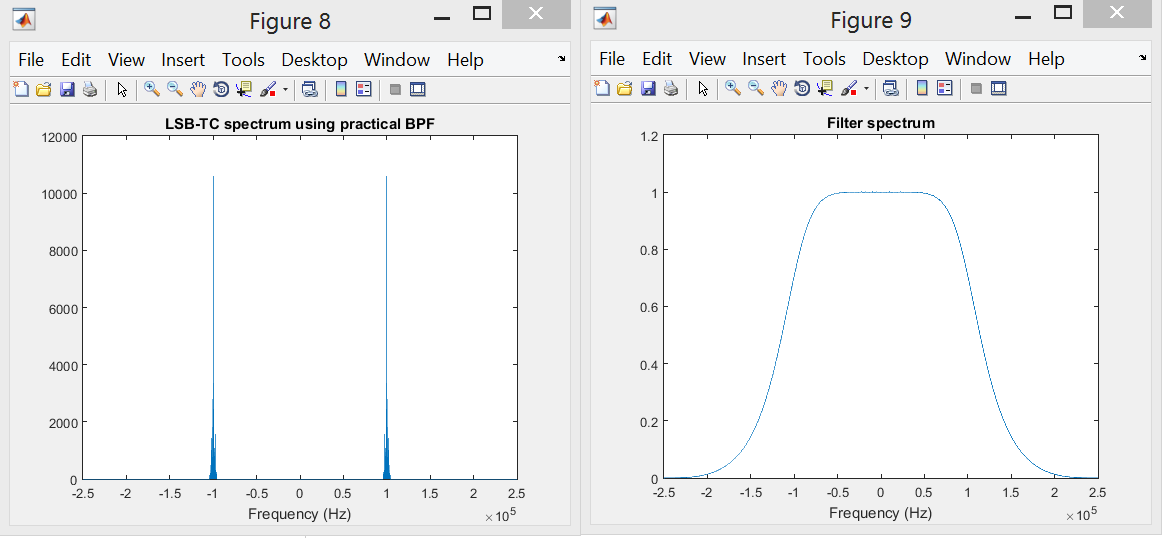
With error equals:   


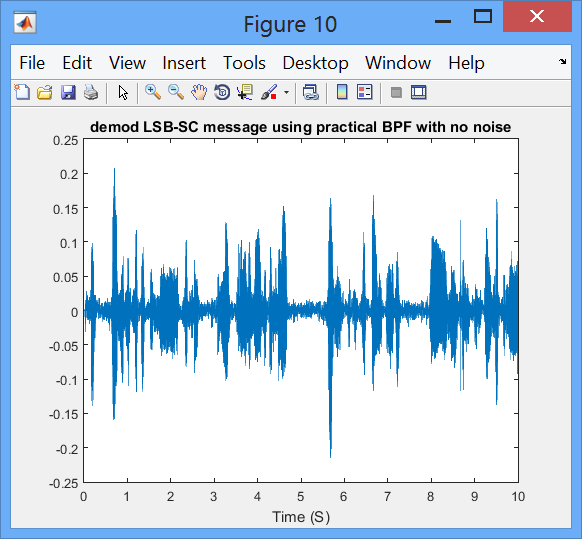
It's very small error, signal still sounds like the original.

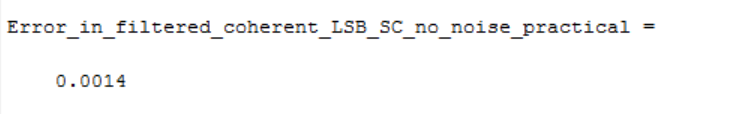
Requirement 8:

Unfortunately the practical filter's order is very low so resultant USB hasn't been trimmed much, it's more like VSB than USB actually.

Still sounds like original, it hasn't changed much in time domain.

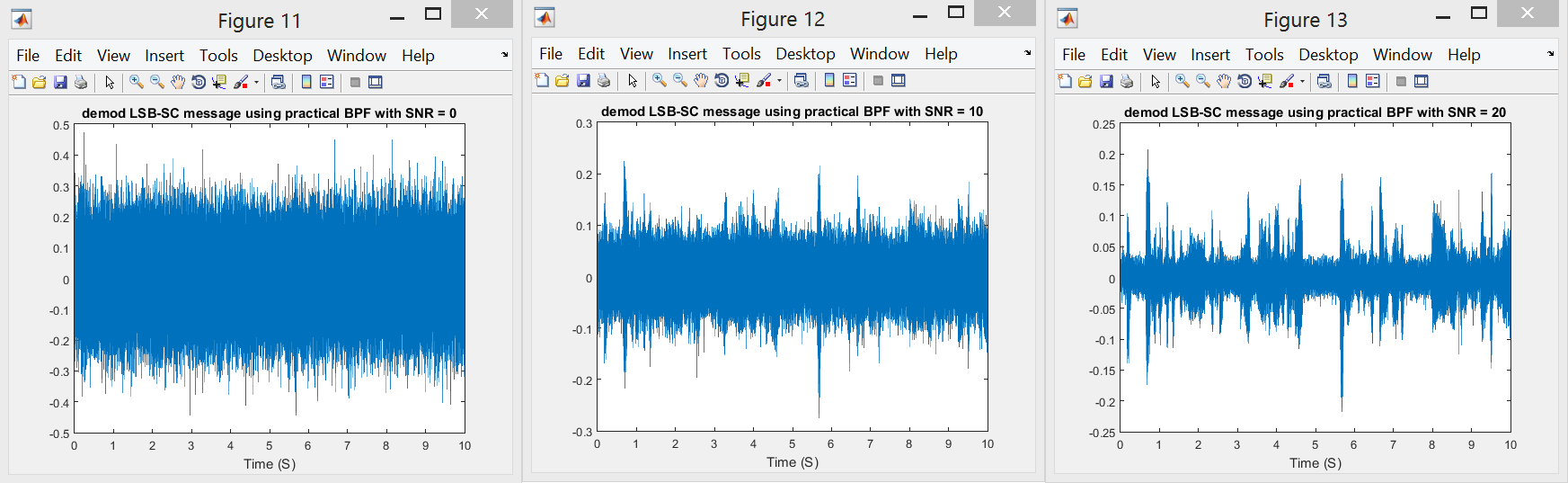


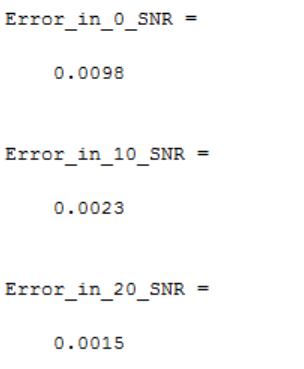




Requirement 9:

Signal is completely ruined at SNR=0, by plating it back it is barely audible at SNR=20 yet noisy.



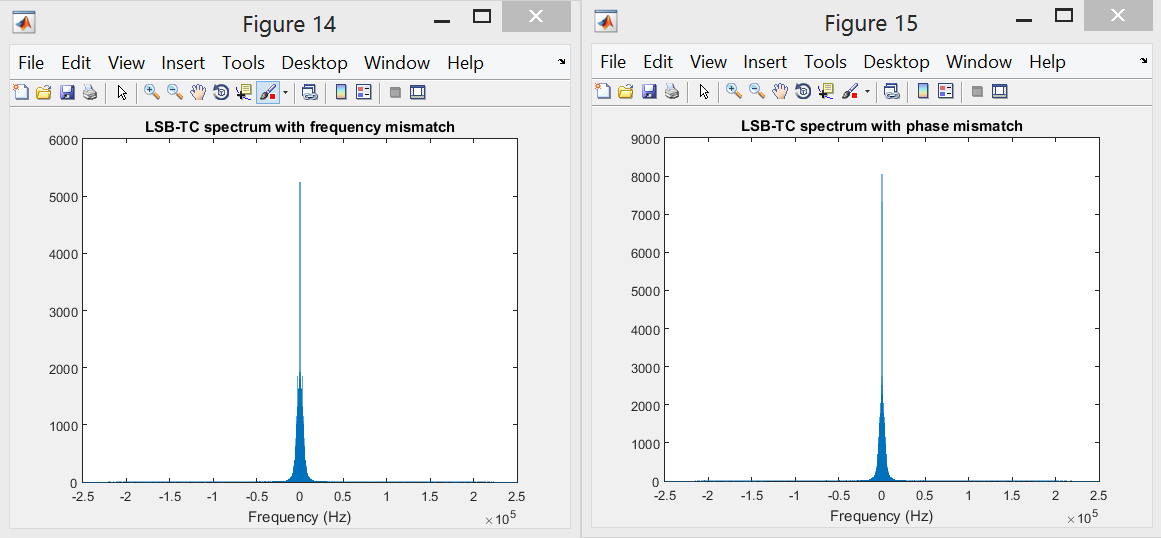


As we see, error increases at lower SNR.

Requirement 10:

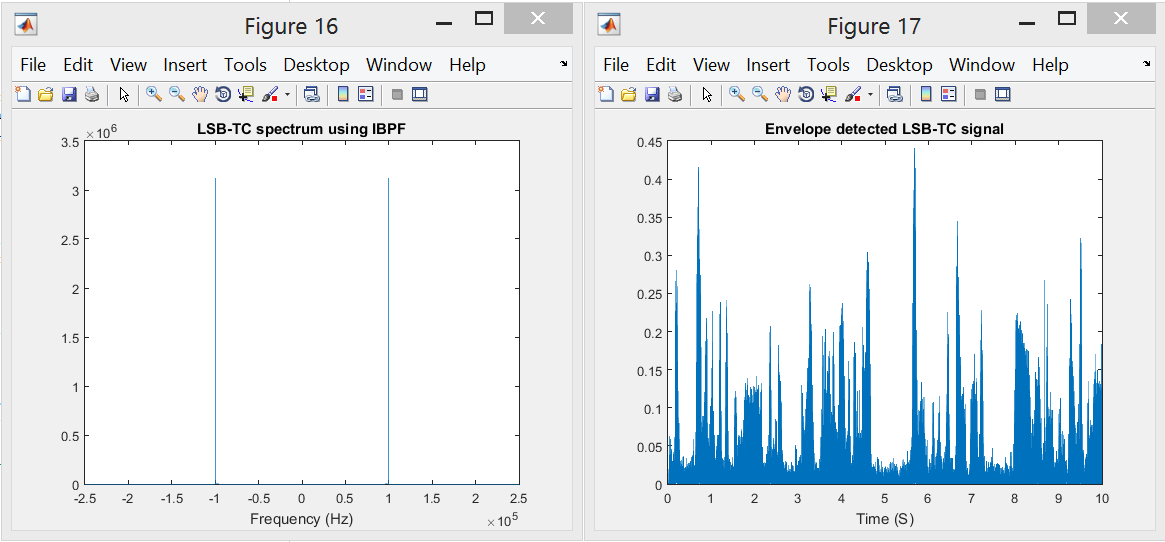
a) For frequency mismatch :  
In frequency domain it appears like signal has double spectrum around 0 frequency which is because of beats effect, when hearing the signal my voice appeared like an alien.

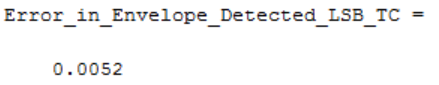
b) For phase mismatch:   
voice seemed to be more attenuated than before.



Requirement 11:

Using ideal filtering to get LSB case, then ED to retrieve signal, that results in a disturbed -yet audible- voice.





Requirement 12:

Disadvantages of SSB are:   
- Complex circuitry is required to implement high order filter.

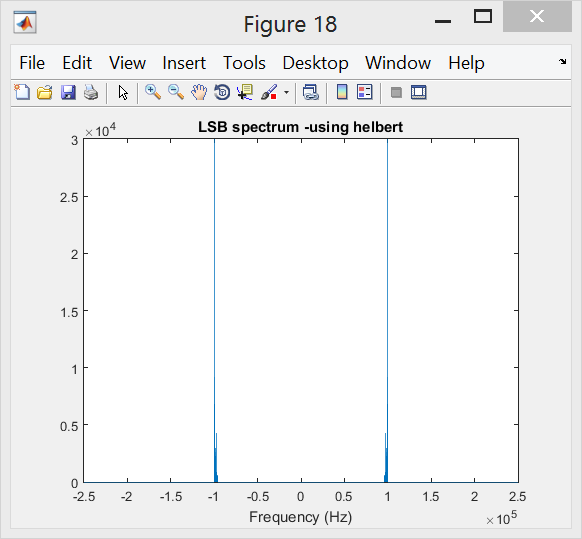
- Oscillator frequencies involved are very critical and must be stabilized otherwise distortion will occur.

- More expansive equipment in comparison to DSB.

Requirement 13:

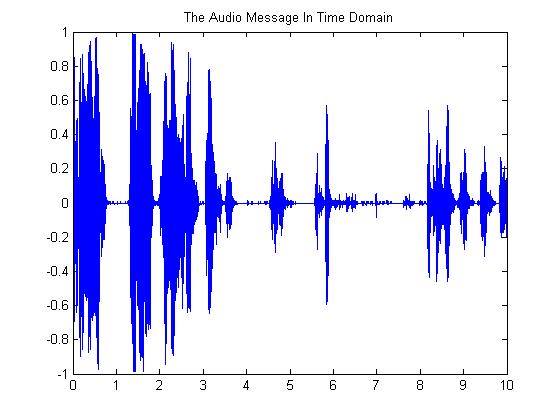
It can be generated as follows:



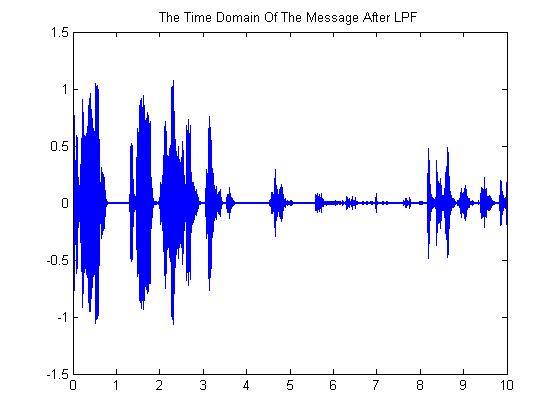


3 FREQUENCY MODULATION

1- The audio message in time domain

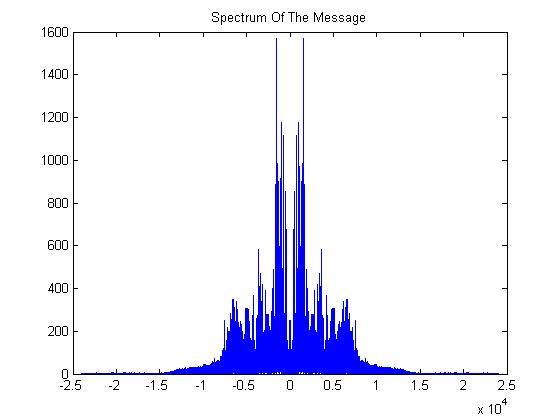


The audio message after LPF

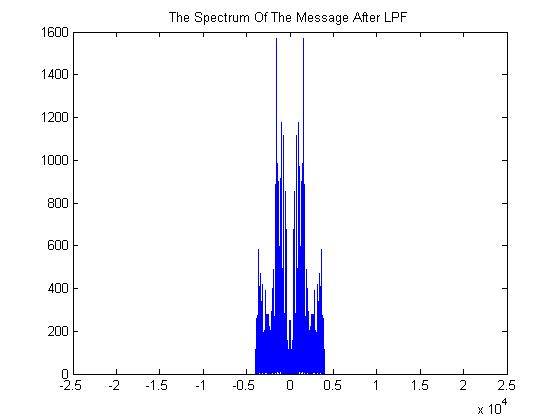


2)

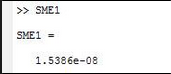
The audio message in frequency domain



The audio message in frequency domain after LPF



3) The Mean Square Error



4) At SNR=0 , The mean square error is much larger the first one before SNR .. and there’s no sound due to high noise.

At SNR = 10, The mean square error is also larger than the one before SNR .. there’s un clear sound ..

At SNR = 20, The mean square error is relatively small compared with that in SNR = 0,10 .. You can clear the sound clearly with little noise ..

Conclusion: As we increase the SNR the mean square error decreases and the noise decreases so that we can hear the sound clearly ..