BANDWIDTH EXTENSION AND QUALITY EVALUATION OF SPEECH SIGNAL BASED ON QMF AND SOURCE FILTER MODEL USING SIMULINK AND MATLAB

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ABSTRACT*: In modern scenario of wireless technology reconstructed voice signal(speech signal) by the side of receiver side is found stifled and slim; So limited bandwidth of 300-3.4KHz and power constraint so as to enlarge ambiguity, genuineness of voice signal(speech signal), Narrowband speech encoders must be upgrade to Wideband encoders supports Bandwidth 50-7KHz. one decade has been departed for upgrading existing NB systems to fully WB compatible systems. terminal and network must be changed to make NB system compatible to WB system. during that span novel technique has been urbanized to widen the Narrowband Bandwidth(BW) of speech (voice) signal at handset end (receiver) for humanizing final speech quality. This study has been undertaken to investigate the novel approach based on QMF filter to separate out the wideband speech into LF and HF part and processing it individually so that without wideband coder at transmitter side the near perfect original wideband speech has been recovered and the obtained speech signal can be further processed through source filter model intended for evaluating signal based on speech quality frequency domain spectrogram etc.. to judge the speech signal compression. From the obtained speech signal at the time varying synthesis filter it can be concluded that source filter model can requires less number of bits compared to whole original speech signal transmission at the cost of degradation in speech quality.one can notify the significant changes in original speech quality by observing the spectrogram of speech signal. less number of bit representation can help in reducing the storage requirement and bandwidth. In recent scenario of technology voice signal(speech signal) BW expansion be able to play a vital role to improve speech quality.*

Key Words: *Narrowband signal, Quadrature Mirror filter, Super wideband, Spectrogram*

I. INTRODUCTION

In digital telecommunication systems it is always necessity to transmit voice signal(speech signal) strongly[1,20,].Public switched telephone systems(PSTN) shrink the bandwidth of the transmitted voice signal(speech signal) from an effective frequency range of 50 Hz to 7 KHz to the range of 50 Hz to 3.4 KHz. The condensed bandwidth leads to a characteristic slim and muffled sound.. as per the recommendation of ITU-T conducted Listening tests have revealed that the speech bandwidth affects the perceived speech quality [2].Introduction of wideband communication system aggravated the transmission of wideband signal having cutoff frequency of at least 7 kHz for improved speech quality in form of the intelligibility and naturalness. the restricted access problem in the employment of Wideband coders and communication is up gradation of current NB coders and transmission to WB system where hardware and software up gradation and compatibility is forever a major issue. better voice signal(speech signal)quality performance offered by Wideband coders, at rest hasty alternate of complete NB coding and transmission systems is not sufficient due to remarkable infrastructure expenses incurred to network operators and also for the customers who wants to make use of the system. one decade has been departed been departed for changeover existing NB systems to fully WB compatible systems. terminal and network must be changed to make NB system compatible to WB system. during that span novel approach have been urbanized urbanized to widen the Narrowband Bandwidth(BW) of speech (voice) signal at handset end (receiver) for humanizing final speech quality. [3]. Bandwidth extension (BWE) system for speech signals is a compelling ,reasonable option to acquire wideband speech with excellent quality sound for the existing wireless communication infrastructures, like public switched telephone network (PSTN) and global system for mobile communication (GSM)[21].

Bandwidth extension techniques of speech are mostly categorized into two classes. In one class the missing frequency component is achieved from the available NB speech component with not necessary to transmit data about the stolen frequencies because of the expansion is wholly w.r.t. NB speech signal, these techniques can be done at the enviable end of the channel. The other depends on steganography(data hide) method. The greater part of the prior strategy make WB speech by source filter model (LP) demonstrate [4], excitation signal and LPC coefficient for spectral envelope. The BWE techniques of speech in view of data hiding method insert High frequency segments data into the NB speech bit stream, and then at the user end terminal the WB speech is recuperated w.r.t. High frequency data. The techniques are without charge and can be utilized with any NB speech codecs. A limitation of BWE technique with side data is that their abuse requires that a similar technique is bolstered at the two ends of the transmission line [5,6]. Then again, the execution of such techniques can clearly be better than that of prior extension techniques. In this paper QMF and source filter model based bandwidth extension and quality evaluation of speech signal is carried out. In QMF's, the whole signal could be partitioned into two parts meaning that full band voice signal(speech signal) is partitioned into two half band signal. The extraordinary properties of these channels permitted the half band signals to be sampled at half rate to the original sampled rate and then reproduced effectively. This disclosure offered ascend to an entire generation of speech sub band coders. the obtained speech signal can be further processed through source filter model for evaluating the performance of the speech signal in terms of speech quality, frequency domain spectrogram etc.. to judge the speech signal compression. this paper is organized as follows:- Sect.2. describes the problem and Sect.3. denotes the details of reported previous works. In Sect.4. theory related to novel approach for bandwidth extension based on QMF and source filter model has been discussed. In Sect. 5 the results obtained through series of simulation in MATLAB has been reported. Finally the concluding remarks are given in Sect.6.

II. Problem Definition

In the modern state of affairs of technological advancement in wired and wireless communication systems, surrounded by plentiful reasons for on the whole degradation of recovered voice signal(speech signal) quality at receiver end, major reasons to be considered is the utilization of Narrow Band (NB) end devices and NB transmission medium supporting bandwidth of 300 Hz-3.4KHz. The intrinsic negative aspect of such NB voice signal(speech signal) is that sounds quality appeared is stifled and thin because of absence of High Band (HB) spectral components[1].The limited frequency band trim down both quality and intelligibility of speech due to the missing high frequency components that are significant cues especially in consonant sounds[6]. factors which may affect the quality of recovered speech are limited acoustic bandwidth(BW),acoustic background noise, quantization noise due to source coding and residual error after channel decoding. Fricatives like /s/, /z/ and partly /f/, /S/, /Z/ are not easy to recover using only NB speech as considerable energy of these fricatives are located in Extension band (HF band) thus restrict performance, voice(speech) quality of recovered voice(speech) signal[9]. To get better quality, intelligibility of speech degraded by narrow bandwidth(NB),researchers have almost concentrate on top of standardize the telephonic networks by introducing wideband (50-7KHz) voice(speech) encodes. Wideband (WB) speech transmission requires the transmission network and terminal devices at both ends to be upgraded to the wideband that turns out to be time-consuming[7]. software cum hardware up-gradation,compatibility,feasibility are major issues to abruptly substitute entire existing NB coding system.[8]. this paper focus on the novel approach based on QMF filter to separate out the wideband speech into LF and HF part and processing it individually so that without wideband coder at transmitter side the near perfect original wideband speech has been recovered and the obtained speech signal can be further processed through source filter model for evaluating the performance of the speech signal.

III. Previous works.

Schitzier (1998) gives a solution to trim down the wideband speech coder bitrates by coding the parameters of wide band voice(speech) by means of noteworthy enlarge in bitrates of NB coders. Makhoul and Benarti (1979), Carl and Heute (1994), Yoshida and Abe (1994), Jax and Vary (2000) were discussed various approach that make use of the WB enhancement source filter model(Linear prediction). This methods synthesis the WB speech from the Pitch, Voicing , and spectral envelope information of Narrowband voice(speech) signal. so many bandwidth extension methods, e.g. codebooks [10,11] , linear mapping [12], Neural Networks etc is used to estimate the missing components. once more Jax and Vary [13,14] bring into being the potential features of speech and evaluate their performance for BWE application. N.S.Bhatt [2014] proposes a novel approach to extend the bandwidth of recovered speech signal at receiver without modification and up-gradation of current NB transmission channels under varying channel condition using CELP based AMR NB coder. In Ref.[16] author has done work on various methods for bandwidth extension.

n Ref.[17] author has designed the Source-filter model based on vocal tract and implemented Bandwidth extension algorithm using LPC coefficients.

IV. Bandwidth Extension based on QMF and evaluation of speech signal through Source filter model

The Bandwidth extension system based on QMF is shown in Fig. 3,which containing Hi band and low band analysis subband coder , down sampler, G.711 encoder at transmitting terminal, G.711 decoder, up sampler, Hi band and low band synthesis subband coder. At user end terminal, from Fig. 1, to start with, original WB speech signal through 16 KHz sampling frequency is place into two channel QMF bank [18]. Then channel bank's outcomes are decimated by two. In this way both HF and LF segments through 8 KHz sampling frequency are acquired. Next, the LF parts are encoded by the narrow band encoder which is G.711 encoder[19]. the HF part is transmitted without changes at the transmitter side to the receiver through transmission channel. At the user end, NB speech signal is decoded through NB speech decoder and HF is restored as it is. finally the signal is synthesized by Hi band and low band synthesis subband coder and near perfect reconstruction of original signal is obtained at output which can be observed on spectrum analyzer and listen speech file played on Audio device writer.

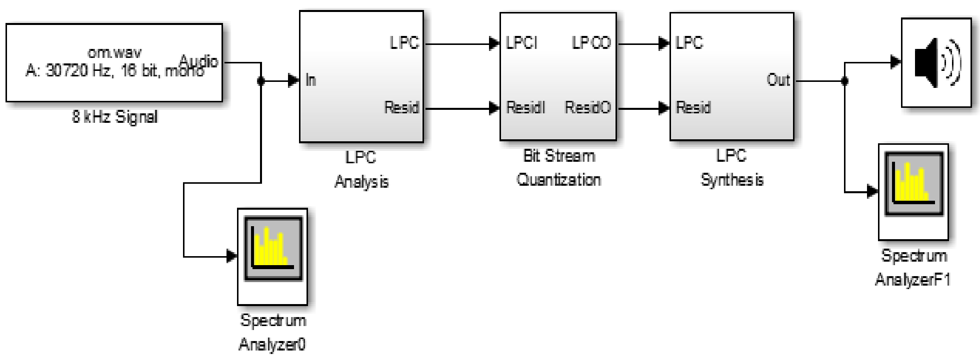

Figure 2. Performance evaluation of speech signal based on Source filter model

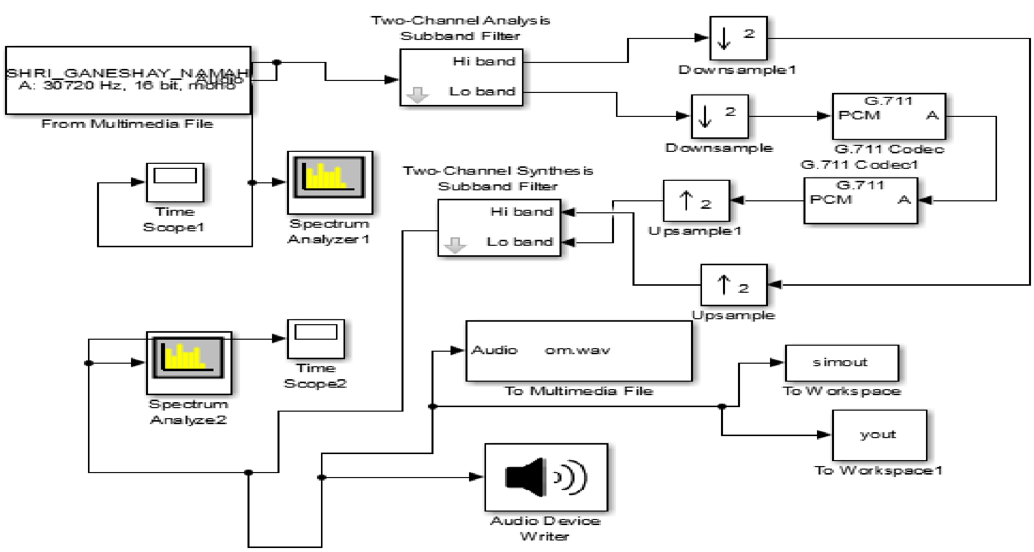

Figure1. Bandwidth extension based on QMF

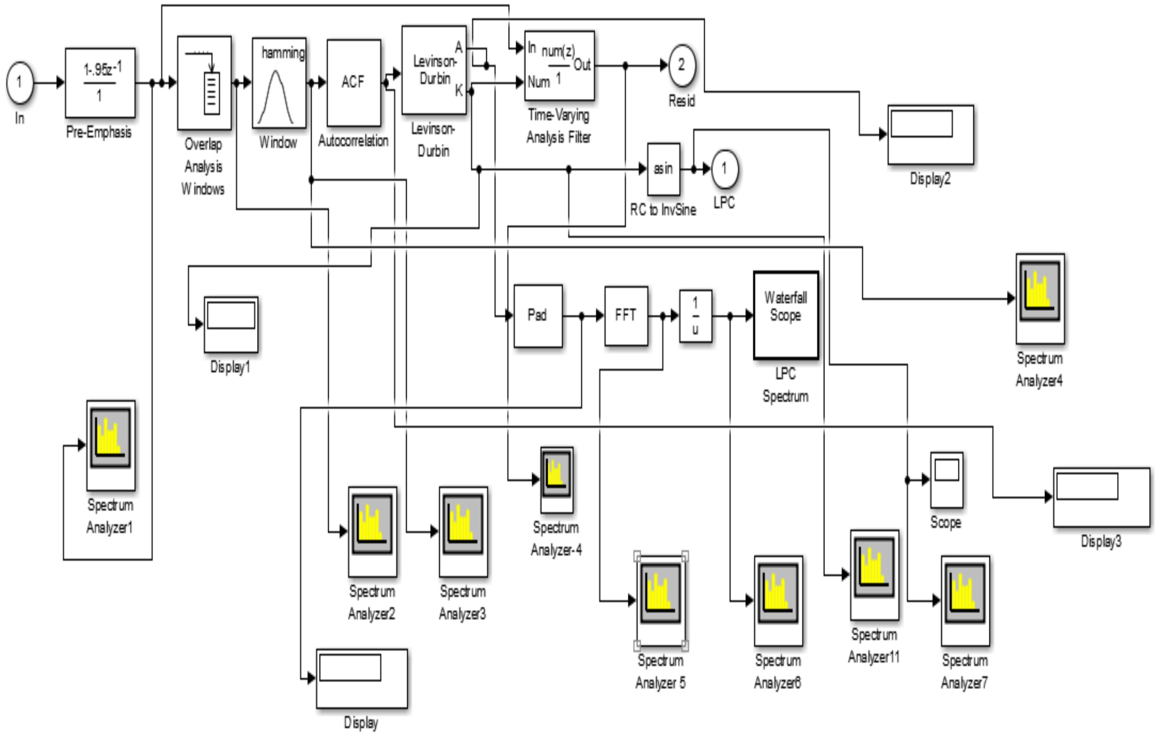

Figure 3. Source Filter model based Analysis

Performance evaluation of speech signal based on Source filter model, Source Filter model based Analysis Source Filter model based Synthesis are shown in Fig. 2, Fig. 3 and Fig. 4 respectively. as shown in figure 2.the original speech signal is pass through LPC analysis block to separate out LPC co-efficient and residual component which is further processed by bit stream quantization to make quantized signal level which is processed by LPC synthesizer to make compressed original speech signal.

As shown in figure 3.the speech signal(wave file) is break up into frames of size 20 ms (160 samples), with an overlap of 10 ms (80 samples). Each frame is windowed using a Hamming window. Then Eleventh-order autocorrelation coefficients are initiate, and then the reflection coefficients are calculated from the autocorrelation coefficients using the Levinson approach. The original speech signal is conceded via an analysis filter, that is an all-zero filter with coefficients as the reflection coefficients obtained above. The output of the filter is the residual signal.

as shown in figure 4. LPC Synthesizer that is time varying filter found in the receiver section of the system, reconstruct the original signal using the reflection coefficients and the residual signal. This is played through the Audio player.

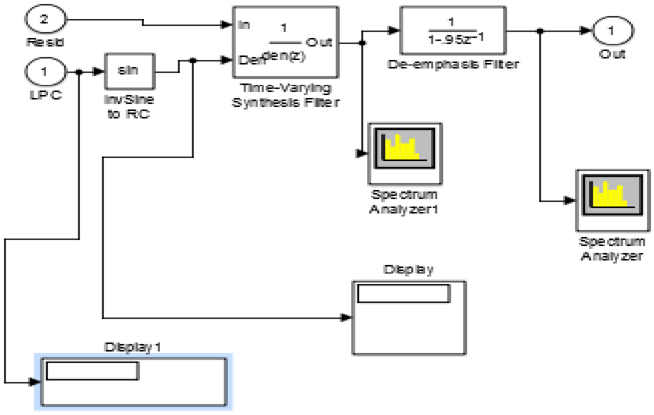

Figure 4. Source Filter model based Synthesis

V Results and Discussion

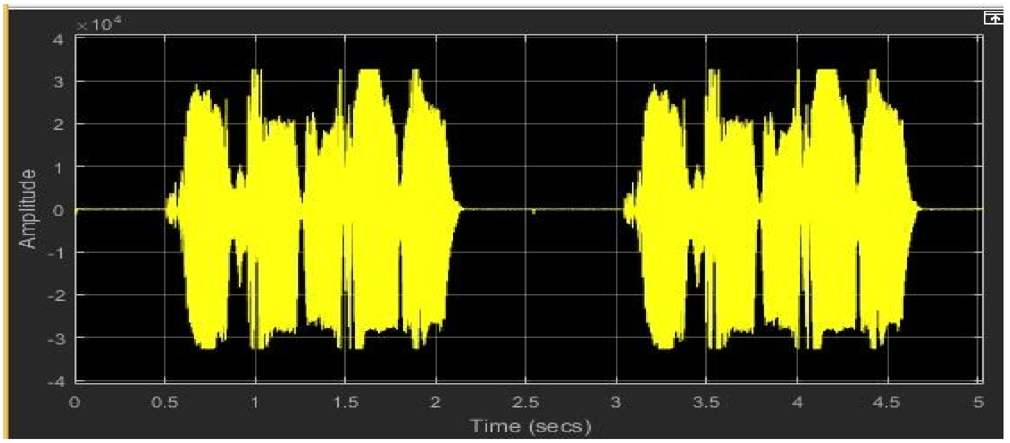

Figure 5. input time domain waveform of wave file "OM SHRI GANESHAY NAMAH"

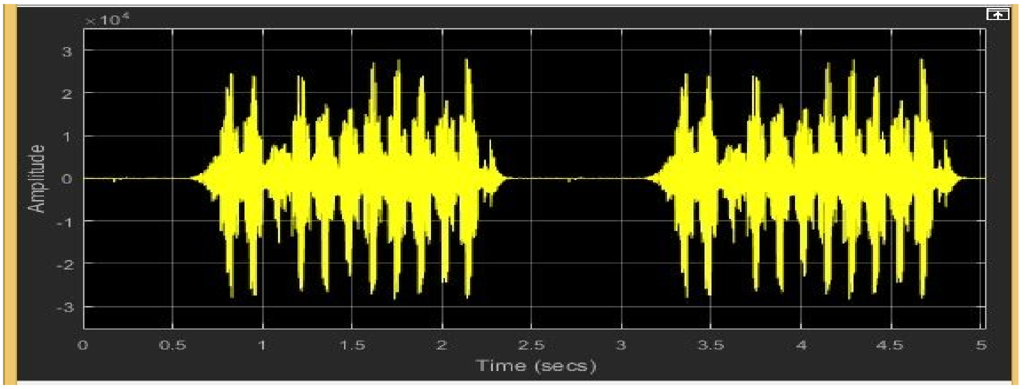

Figure 6. output time domain waveform of wave file "OM SHRI GANESHAY NAMAH"

Figure 7 & 8 shows the input and output time domain waveform of the wave file ''OM SHRI GANESHAY NAMAH'',while Figure 9 & 10 the input and output frequency domain waveform of the wave file ''OM SHRI GANESHAY NAMAH''. from the figure it is clearly identified that the signal looks very different in time domain but in frequency domain it is looks like same because of the nearly same power distribution at the same frequency will be observed in the figure which can be heard on audio player. the result displayed on each stage by using spectrum analyzer or time scope or display is taking some time because in our simulation 1536 samples need to be updated for displaying result at each stage once the required number of samples are acquired there is no issue regarding results. one other noticeable remarks from viewing various time/frequency domain waveform is that at each stage of the source filter model approach the sampling frequency is changing so variation in signal waveform can be obtained.

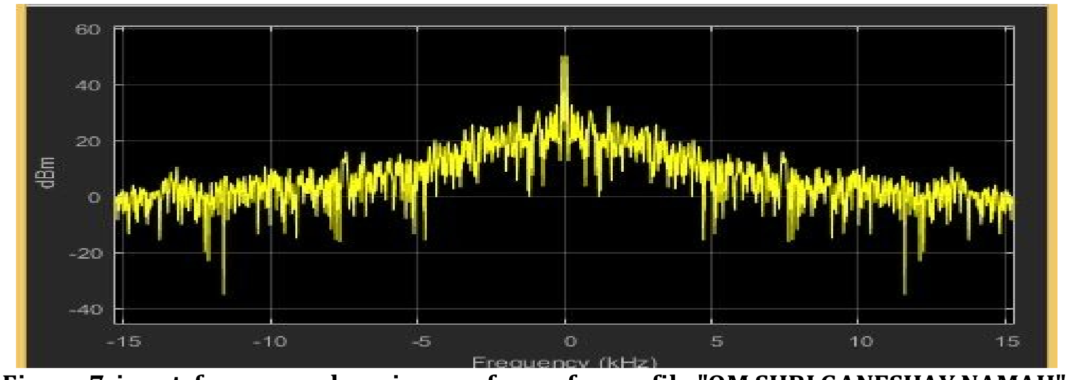

Figure 7. input frequency domain waveform of wave file "OM SHRI GANESHAY NAMAH"

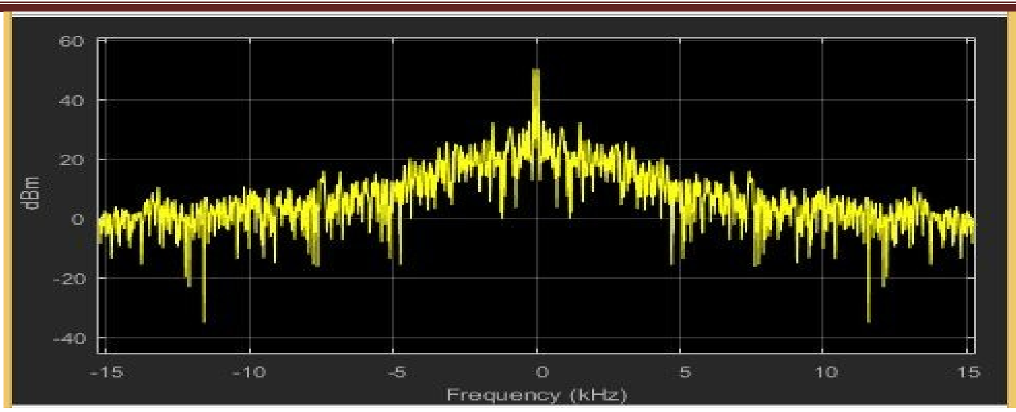

Figure 8. output frequency domain waveform of Wave file "OM SHRI GANESHAY NAMAH"

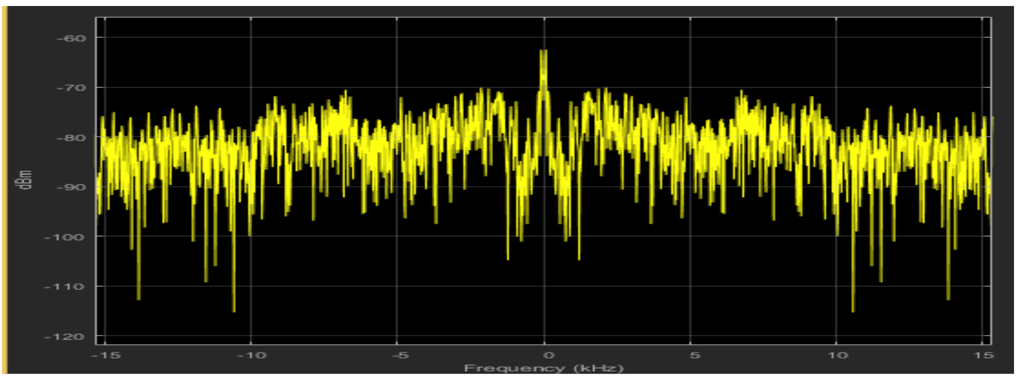

Figure 9. Pre-Emphasized Speech signal

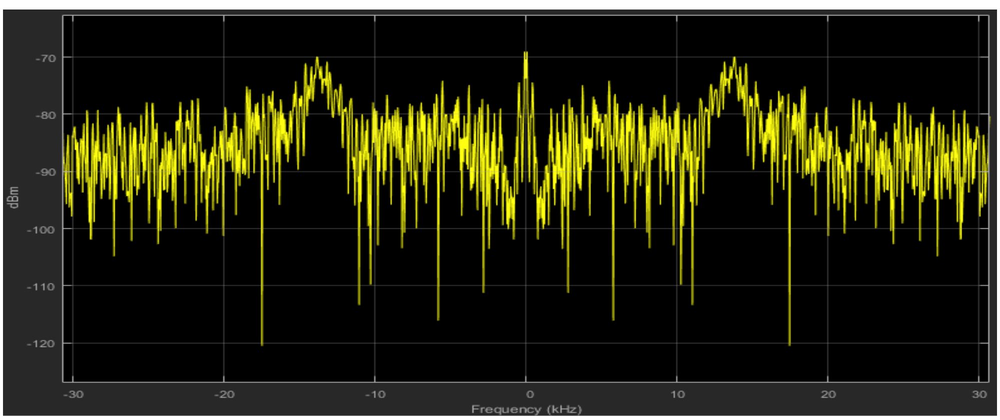

Figure 10. hamming windowed Speech signal

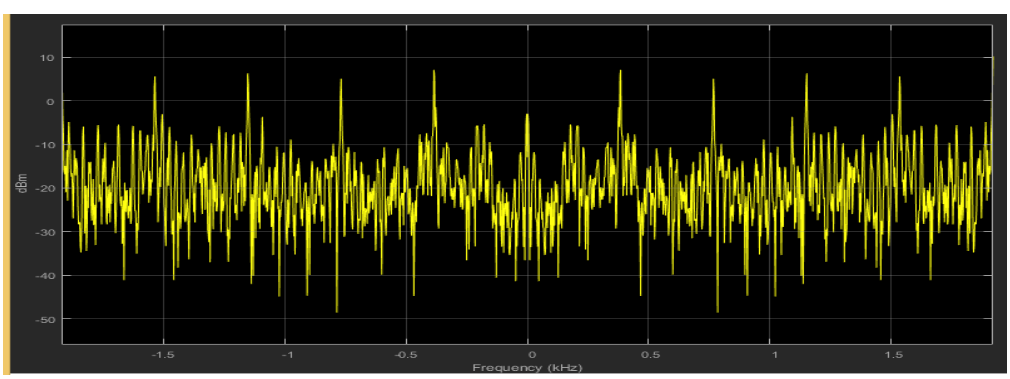

Figure 11.LPC analyzer output

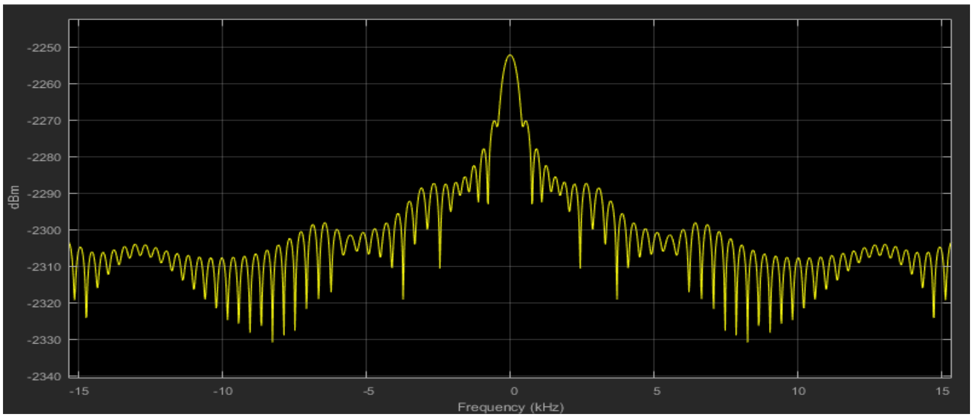

Figure 12.LPC synthesizer output

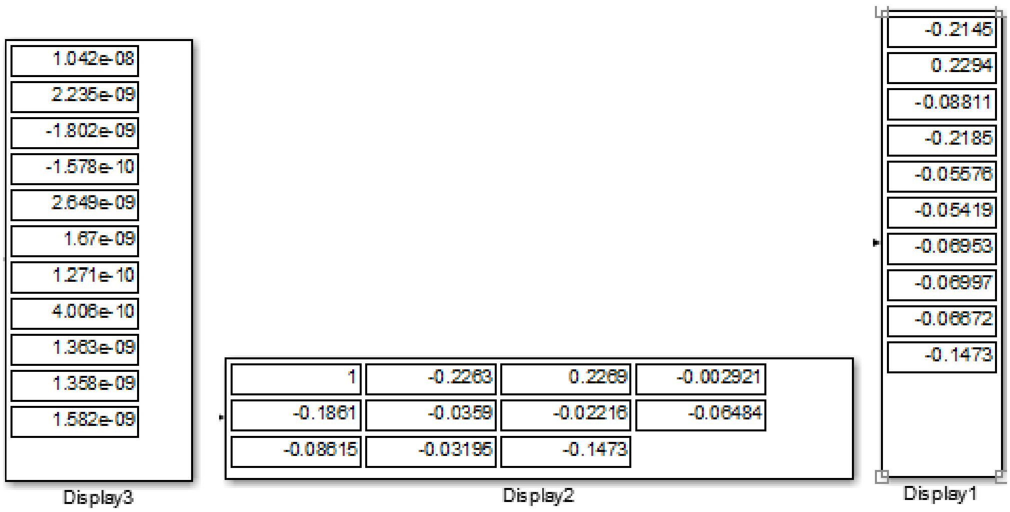

Figure 13.Auto cor-relation co-efficient (ACF), Reflection co-efficient(RC) and LPC co-efficient

VI. Coclusion

The Novel approach presented in the paper can attain the required Bandwidth extension to improve stifled and thin speech quality. without wideband coder at transmitter side the near perfect original wideband speech has been recovered and the obtained speech signal can be further processed through source filter model intended for evaluating signal based on speech quality, frequency domain spectrogram etc.. to judge the speech signal compression. by employing source filter model in place of 160 samples only 10 LPC co-efficient, 01 bit for voiced/Unvoiced Design 01 Bit- for gain are needed so large signal compression achieved which helpful in memory and storage requirement. due to technological advancement the author is target to develop MATLAB code for suprewideband signal from wideband signal and compare the performance analysis of the signal with Real super wideband EVS coder and AMR-WB coder. Also the stability consideration of RC co-efficient is the targeted by author for future work.[22].

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