

# Noise Reduction in Speech Coding Technique Used in GSM

Pratiksha D. Gadge, S.S. Gundal

**Abstract:** *The improvement in speech quality in mobile communication can be made by reducing the noise content. The low bit rate speech coding technique such as RPE-LTP in GSM with noise signal filtering can increase the complexity of complete mechanism. The paper presents noise signal reduction in RPE-LTP speech coding technique by using Zero Crossing Rate (ZCR) and pitch values. By using this technique quality measures of speech coding technique such as SNR, PESQ, MSE are improved as compared to RPE-LTP technique without affecting complexity.*

**Index Terms:** GSM, Pitch, RPE-LTP speech coder, ZCR.

## I. INTRODUCTION

The most widely used mobile communication standards in the world are Global System for Mobile (GSM). GSM is having more services, but wireless speech communication is the most important services. The speech coding techniques used in GSM are GSM half rate (HR), GSM full rate (FR) codecs, GSM adaptive multi-rate codecs and GSM enhanced full rate codecs. In wireless communication, signal travels from a transmitter to the receiver via radio channels. The radio spectrum is a limited, natural resource with radio channels allocated for different communication applications. The demand of wireless communication services has increased tremendously in the last few years. The digital speech coding technology utilizes the knowledge of both speech waveform redundancy and human perception of sounds to represent voice signals in a compact form. In forensic speaker identification, author changes the GSM channel frequency and energy to analyze the influence of the channel. It also describes the different methods of channel compensation using inverse processing. The channel compensation with a proposed method is more transparent, effective and convenient as compare to Cepstral mean subtraction [1]. The technique of modification framework choice methodology in Regular Pulse Excitation areas of Full Rate GSM 06.10 coder is proposed with the intention that there is a general 1.8kbps diminishment in bit-rate to cover high block rate information. GSM 06.10 FR coder and future steganographic codecs are actualized by Simulink in MATLAB [2]. The proposed method works on real-time basis on Android Smartphones. The use of pipeline as in smart phone has suppressed noise and enhanced speech quality [3].

Increasing spectral efficiency and transforming of speech signals into digital form is the main problems highlighted in this technique. There are too many efforts needed to develop a proper speech coding algorithm. To maintain the quality level and decreasing data rate of voice signal is important in transmitting signals over air interference [4]. By using GSM and PCM telephone standard in speech recognition quality is improved. This system is based on automatic telephone call speaker recognition. Full rate GSM coder and GMM are used to represent speech recognition [5]. Modifying codec parameters are discussed to reduce environmental background noise. It is essential to calculate the noise reduction technique in other speech coder to perform well with respect to codec parameters such as bit rate and speech quality [6]. By using discrete wavelet transform technique audio compression technology is designed as well as implemented. The DWT supported by the equal model because of audio compression. The choice of the daubechies wavelet with the DWT yielded comparable improvement within performance parameters with an honest quality psychoacoustic reconstruction of speech signal samples. Different measuring parameters are used to calculate efficiency of audio encoding methods. These parameters are compression ratio, the ratio of signal to noise and ratio of peak signal to noise [7]-[8]. The convex set projection technique is proposed for improvement in recovery of speech spectral parameters [10]. Other techniques such as predictive coding to minimize the RMS error and noise masking before quantization steps LPC excitation, multipulse excitation have been proposed to improve the speech signal quality [9][11]. Speech quality value is better by using voice enhancement device (VEDs) for all calls despite of handset models. Voice quality is improved by using ITU-TG107 and e-model in VEDs. So that GSM network quality index is improved [12]. The fundamental objective of this system is to create procedure that can upgrade the yield nature of irregular transmission speech coding frameworks working in noisy acoustic situations. The focal point of our work will be on outlining GSM speech coding strategy which is utilized to decrease noise in speech signal.

## II. SYSTEM DESCRIPTION

The Fig.1 and Fig.2 shows speech encoder and decoder block diagram of FR GSM coder respectively. The digital communication system is constructed for speech transmission and receiving purpose by using FR GSM.

**Revised Manuscript Received on July 05, 2019**

**Miss Pratiksha D. Gadge**, Dept. of Electronics Engg., Sheti & Shikshan Vikas Sanstha's, Amrutvahini College of Engg., Ghulewadi, Sangamner, India.

**Prof. S. S. Gundal**, Dept. of Electronics. Engg., Sheti & Shikshan Vikas Sanstha's, Amrutvahini College of Engg., Ghulewadi, Sangamner, India.

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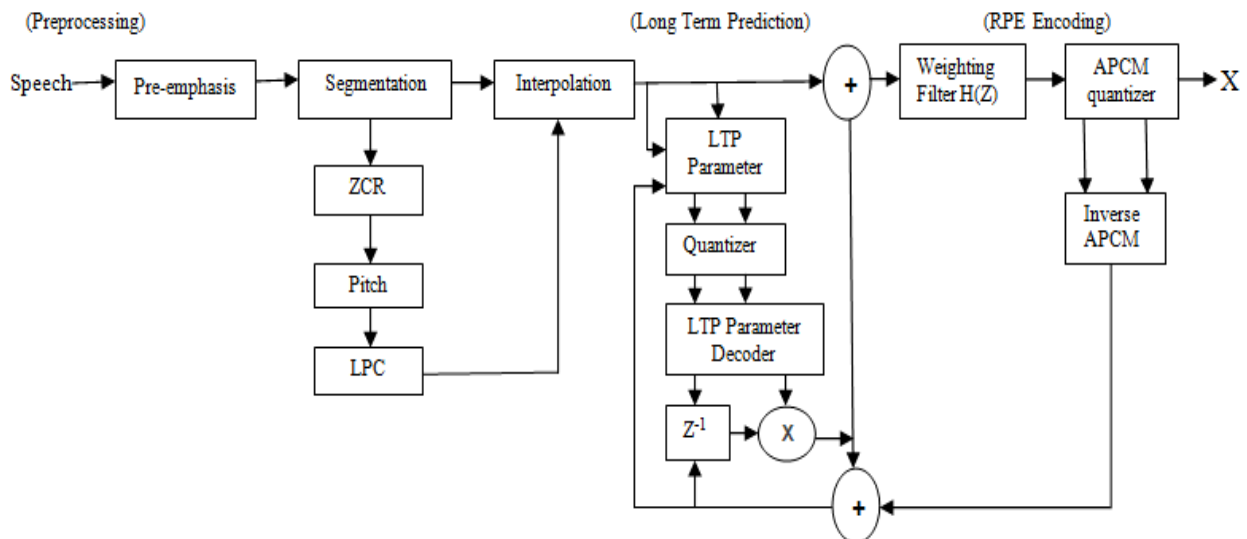


Fig.1. Speech Encoder block diagram of Full Rate GSM coder.

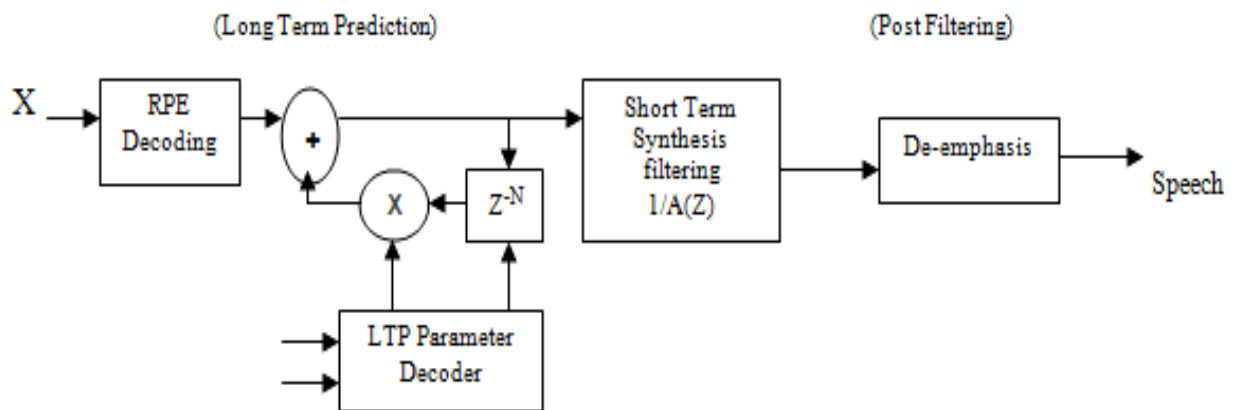


Fig.2. Speech Decoder block diagram of Full Rate GSM coder.

The full rate GSM speech encoder carry thirteen bits uniform pulse code modulation signal as a purpose of input sample. This input is available in audio format. In preprocessing of input speech signal contains 160 samples which is pre-emphasis by using butter worth filter. In LPC analysis section, the ZCR and pitch values are calculated that's why proper segmentation of frames happens. Due to that the measure parameter value is improved and gets better values. In long term prediction of signal samples of LTP lag and LTP gain parameters are analyzed. After that quantization process is happen. The input speech containing 160 sample signal which is divided into four frames of 40 samples in each frame. These 40 sample frame is continued to the RPE encoding part. After that adaptive pulse code modulation process is happen. In that all receiving bits are encoding and which is transmitted to the decoding section. Coding and decoding process of frames is happen with each and every frame of input speech sample.

The encoded sample signal is reconstructed at decoder part. The decoded LTP parameter signal which is shown in Fig.1

and Fig.2 goes to short term synthesis filter (shown in Fig.2) follows de-emphasis filter for the purpose of post filtering and gets original noise free speech signal samples.

### III. RESULT ANALYSIS

Here, FR GSM codec is implemented by using RPE-LTP coding technique. Also coder performance is calculated by using different performance parameters. In Table I Perceptual Analysis (PESQ), MSE and SNR are obtained. SNR values are greater because of that comparatively negligible noise in the reconstructed signal. The original and reconstructed audio signal sample waves are shown in Fig.3. In that First signal shows the plot of original audio signal which having 0.1 maximum amplitude level and second signal shows plot of reconstructed audio file. The average values for MSE is approximately equal to zero. The quality assessment in terms of PESQ is acceptable.

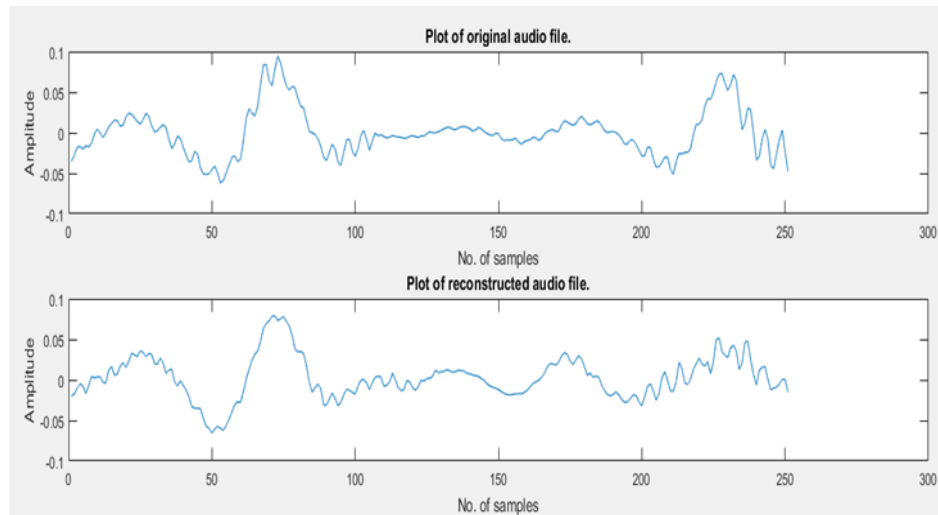


Fig.3: MATLAB output result of speech signal.

Table I: Performance Parameters of Proposed Coder

WAVE FILE	SAMPLES	PESQ ( Perceptual Analysis)	SNR	MSE
1.wav	20880	3.220435567	35.30797093	0.000202956
2.wav	17760	3.230884131	35.33409846	0.00010487
3.wav	16800	3.122959836	36.47366578	0.000246374
4.wav	17520	3.164715824	36.50879504	0.00026465
5.wav	21600	3.354715647	36.49274877	0.000229688
6.wav	19920	3.280039351	36.19742007	0.00012752

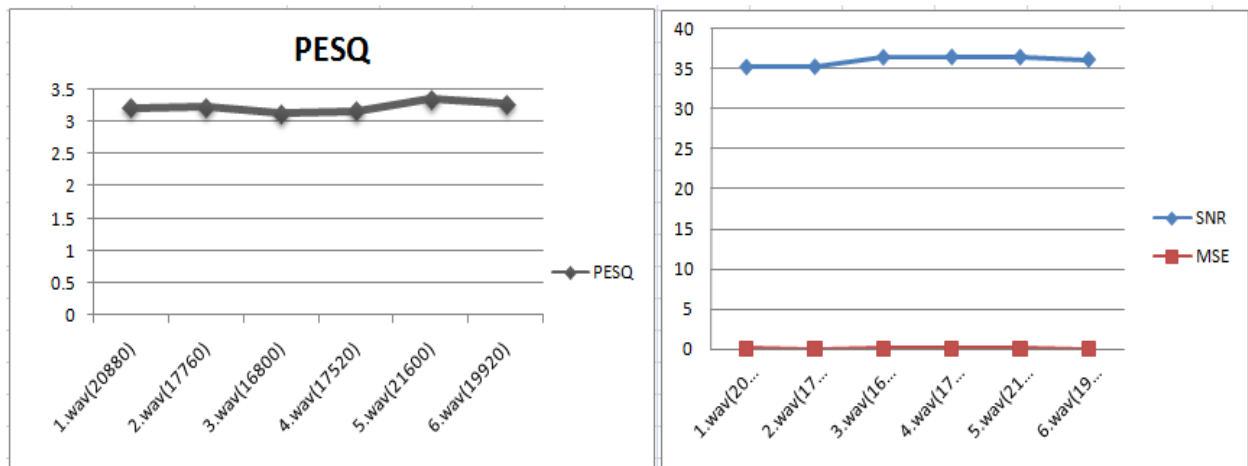


Fig.4. Graphical representation of PESQ, SNR, MSE w.r.t speech wave files

#### IV. CONCLUSION

Speech coding is an important part of our communication services. Our system represents a noise reduction in speech coding technique which is RPE-LTP used in GSM. RPE-LTP technique is depends on objective measures i.e. performance parameter which is measured. The performance of a coder is calculated by using variable objective analysis. In most of the

samples SNR values are greater than 20db so it means that the noise is less or negligible in output speech. The graphical representation of the measured parameter shows the minimum mean square error and improved PESQ values that means error is almost removed and speech signal is recovered. So that by using RPE-LTP GSM coding technique output of speech signal is recovered accurately.

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## AUTHORS PROFILE

**Miss Pratiksha D. Gadge** received her B.E. in Electronics Engg. at Sheti & Shikshan Sanstha's, Amrutvahini College of Engg. (AVCOE), Sangamner, Savitribai Phule Pune University, Pune. She is also pursuing her Master of Engineering and her area of specialization is Digital System.

**Prof. S. S. Gundal** is currently a Associate Professor and HOD of Electronics Engg. at Sheti & Shikshan Sanstha's, Amrutvahini College of Engg. (AVCOE), Sangamner, Savitribai Phule Pune University. She received her M.E. in Electronics Engineering and pursuing Ph.D. She has 19.8 years of teaching experience and her areas of specialization are Speech Processing and Cognitive Radio.