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Parametric Analysis of FIR Digital Filter for Wireless VoIP System

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Abstract: In Voice over Internet Protocol (VoIP) system, the speech signal is degraded when passed through the network layers. The speech signal is processed through the best effort policy based IP network, which leads to the network degradations including delay, packet loss jitter. The implementation of the signal through digital signal processor can improve the quality of the degraded VoIP signal. The work in this paper presents the parametric analysis of the Finite Impulse Response (FIR) filter for speech quality improvement in wireless VoIP system.. The VoIP simulations are conducted with G.711 and G.729A speech coders at different packet loss rates. The performance of the enhanced VoIP signal is evaluated using Perceptual Evaluation of Speech Quality (PESQ) measurement for narrowband signal. The results show significant improvement in the quality of the VoIP signal.

Keywords: VoIP, Signal processing, FIR digital filter, Packet Loss

I. INTRODUCTION

Voice over IP is an advancing technology that is used to transmit voice over the internet or a local area network using internet protocol (IP) [1].This technology provides enhanced features such as low cost compared to the traditional Public Switched Telephone Network (PSTN). VoIP system costs as much as half the traditional PSTN system in the field of voice transmission. This is because of the efficient use of bandwidth requiring fewer long-distance trunks between switches [2]. Packet switched networks like Internet, are based on the Best-effort policy which does not guarantee a minimum packet loss rate and a minimum delay of packet transmission required for VoIP system. This results in harmful effects on the quality of VoIP, since speech packets can be discarded when routers or gateways are congested.

Due to the real time requirement for interactive speech transmission, it is usually impossible for the receivers to request the sender to retransmit the lost packets. When voice packets do not arrive before their playout time, they are considered as lost and cannot be played when they are received. One of the most difficult problems in such networks is the packet loss issue. Even a single lost packet may generate audible distortion in the decoded speech signal. To reduce the effect of packet loss on perceived speech quality, the lost packets have to be regenerated at the receiver using packet loss concealment algorithms.

Waveform substitution algorithms have been used successfully for Pulse Code Modulation (PCM) speech coder [3, 4].The lost packets were also regenerated with the use of time scale modification algorithms [5, 6]. Feng et.al implemented ITU-T G.729 and G.723.1 speech codecs on TMS320C6201 DSP processor. The optimization methods used in the work had reduced the speech processing time. G.729 codec was able to process concurrently 20 voice channels and G.723.1 codec able to process 18 voice channels with single TMS320C6201 chip in IP telephony gateway [7, 8]. Langi [9, 10] had implemented ITU-T G.723.1 voice codec algorithm for VoIP gateways on TMS320C5402 DSP processor and use optimization techniques to improve processing time. Han et.al [11] had raised the issue of noise reduction for VoIP speech codecs. The authors proposed a modified Wiener filter based noise reduction scheme optimized to the estimated SNR at each frequency bin as a logistic function. The proposed system was used as preprocessing before speech encoding.

Since the work done in the literature was only to improve the speech coders and some optimization techniques were applied to reduce the processing time. But the work in this paper differ that window based finite impulse response filters are applied to improve the signal quality in narrowband VoIP system.

The brief description of the VoIP speech coders is presented in Section II. The modeling of the IP network and VoIP simulations are presented in Section III. The section IV describes the design of the digital filters used for VoIP system. The performance analysis results and discussion are presented in Section V. The last section concludes the work and presents the future work.

II. VoIP SPEECH CODERS

The frequently used codec is ITU-T G.711 [12] standard codec which encodes the speech signal at 64 kbps with sampling frequency 8 kHz. This codec is most popular since it is capable of fulfilling all necessary requirement for proper communication such as low delay, low distortion. G.711 is the least processor-intensive codec and provides the toll quality speech signal in VoIP system. But the bandwidth consumption is high in using this codec.

The G.729A coder is based on a Code-Excited Linear Prediction (CELP) coding model. In this model the throat and mouth are modeled as a linear filter and voice is generated by a periodic vibration of air exciting this filter.

The locally decoded signal is compared against the original signal and the coder parameters are selected such that the mean-squared weighted error between the original and reconstructed signal is minimized. The CS-ACELP coder is designed to operate with an appropriately band limited signal sampled at 8000Hz. The input and output samples are represented using 16-bit linear PCM. The coder operates on frames of l0 ms corresponding to 80 samples.

For each frame, the speech signal is analyzed to extract the parameters of the CELP model which include linear-prediction filter coefficients, adaptive and fixed- codebook indices and gains. At the decoder, these parameters are used to retrieve the excitation and synthesis filter parameters. The speech is reconstructed by filtering through the short term synthesis filter. After computing the reconstructed speech, it is further enhanced by post-filter [13].

III. IP NETWORK MODELING

The simulation of VoIP system was performed where each packet contains one frame. Packet losses are not independent on a frame-by-frame basis, but appear in bursts. The packet loss can be approximated by Markovian loss model such as Gilbert model, as discussed in [14-20]. Thus simulation of IP network was performed by using a 2-state Gilbert-Elliot Model. The model has two states reflecting whether the previous packet is received or lost. The state “0” represents that a packet being correctly received and state “1” represents that a packet being lost. The Gilbert-Elliot model is shown in Fig 1. Let p be the transition probability for the network model to drop a packet given that the previous packet is delivered i.e. the probability for network model to go from state “0” to state “1”. Let q is the probability for the network model to drop a packet given that the previous packet is dropped, i.e. the probability for the network model to stay in state “1”. This probability is also known as the conditional loss probability.

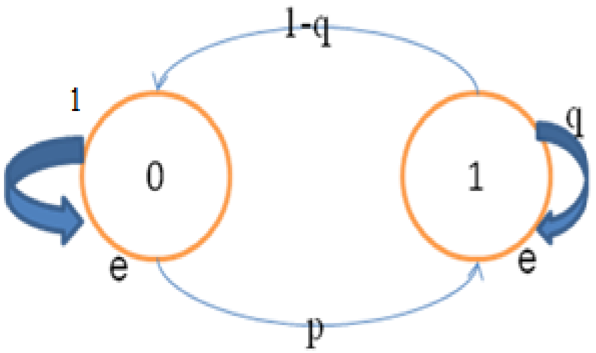

Figure.1 Gilbert-Elliot Model

A. *VoIP Simulations:*

The speech signal is encoded into VoIP frames using G.711 and G.729A for narrowband VoIP system. The speech samples for simulations were taken from [21]. The network impairments were introduced into VoIP frames with the modeling of the IP network through above discussed Gilbert Elliot model. The speech signal of VoIP system is degraded at different packet loss rates (PLR), which are simulated through different combinations of the p and q, as discussed in Table.1. The performance is evaluated with Perceptual Evaluation of Speech Quality (PESQ) measurement defined by ITU-T recommendation P.862 [22] for narrowband VoIP system. After comparing the degraded signal with the original one, the PESQ measurement gives the subjective measurement as Mean Opinion Scores (MOS) value.

Table. 1 Simulated Loss Rates

PLR (%)	p	q	e
3	0.97	0.03	0.005
5	0.95	0.05	0.005
10	0.90	0.10	0.005
15	0.85	0.15	0.005
20	0.80	0.20	0.005

IV. DIGITAL FILTER DESIGN

The window based low pass FIR filter is designed. FIR filter is an all-zero filter in the sense that the zeroes in the z-plane determine the frequency response magnitude characteristic [23, 24, 25]. The basic FIR filter is characterized by:

$$y(n) = \sum_{k=0}^{N-1} h(k)x(n-k) \qquad (1)$$

Where, $x(n)$ is the input sampling sequence, $h(k)$ is the filter coefficients, N is the order of the filter and $y(n)$ is the filter output sequence. The system function can be expressed in terms of the convolution as:

$$y(n) = x(n) * h(n) \qquad (2)$$

$$y(n) = x(0) * h(n) + x(1) * h(n-1) + \cdots . + x(n) * h(0)$$

The discrete time Fourier transform of a finite sequence impulse response $h(n)$ is given by

$$H(e^{j\omega}) = \sum_{n=0}^{N-1} h(n)e^{-j\omega nt} = |H(e^{j\omega})|e^{j(\omega)} \qquad (3)$$

The magnitude and phase responses are given by

$$M(\omega) = |H(e^{j\omega})| = \sqrt{Re[H(e^{j\omega})]^2 + Im[[H(e^{j\omega})]^2}$$
$$(\omega) = \tan^{-1}\frac{Im[H(e^{j\omega})]}{Re[H(e^{j\omega})]} \qquad (4)$$

The z transform of an N-point FIR filter is given by:

$$H(z) = \sum_{n=0}^{N-1} h(k)Z^{-k} \qquad (5)$$

The major advantages of using window method are their relative simplicity as compared to other methods and ease of use. The fact that well defined equations are often available for calculating the window coefficients has made this method successful. The Kaiser window is used to design the FIR filter. The Kaiser window with parameter β is given as

$$W(n) = \begin{cases} \frac{I_o(ß)\sqrt{1-2(n+1)(N+1))^2}}{I_o(ß)} & n = 0,1 \ldots N \\ 0, & otherwise \end{cases} \qquad (6)$$

The Bartlett window reduces the overshoot in the designed filter but spreads the transition region considerably. The Hanning, Hamming and Blackman windows use progressively more complicated cosine functions to provide a smooth truncation of the ideal impulse response and a frequency response that looks better. The best window results probably come from using the Kaiser window, which has β, which allows adjustment of the compromise between the overshoot reduction and transition region width spreading [26, 27]. The proposed FIR scheme for VoIP speech signal improvement is designed using the MATLAB. The performance of FIR scheme is analyzed for VoIP system. The low pass FIR filter is designed for narrowband speech coder with 3100 Hz cutoff and 8000 Hz sampling frequency. The passband and stopband ripples for the designed filters are 0.001 and 0.001 respectively.

V. RESULTS

The performance of the designed filter is analyzed in wireless VoIP system for varying filter length, varying Kaiser Window beta factor and for various packet loss rates.

A. Effect of filter length and Kaiser Window Beta factor:

The performance of the proposed filter is analyzed at various values of the filter length and Kaiser Window beta factor, β, for VoIP system at various values of the packet loss rates. The Kaiser Window beta factor is ranged from 0 to 3.0, since beyond this value the average gain in MOS scores falls rapidly. The variation of filter length with average gain in PESQ-MOS scores for all loss rates, at fixed values of Kaiser Window beta factor, is plotted through Fig. 2(a) - Fig.2 (f). The performance of G.711 and G.729A based VoIP system is analyzed with the proposed filtering scheme. The best results for VoIP system was obtained with filter length of 115 and Kaiser Window beta factor 1.5. The significant increment of 0.51 and 0.55 in average gain in PESQ-MOS scores is obtained with G.711 and G.729A coders respectively. The magnitude response and impulse response of the proposed FIR filter is presented in Fig.3 and Fig.4.

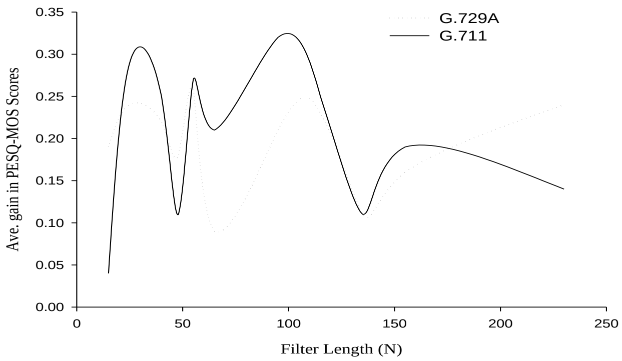

Figure 2 (a) β=0.5

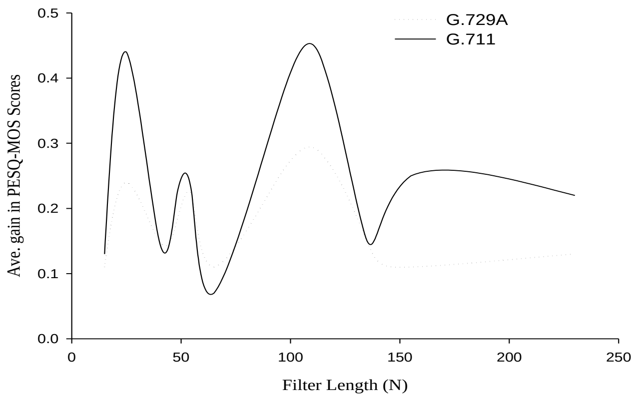

Figure 2 (b) β=1.0

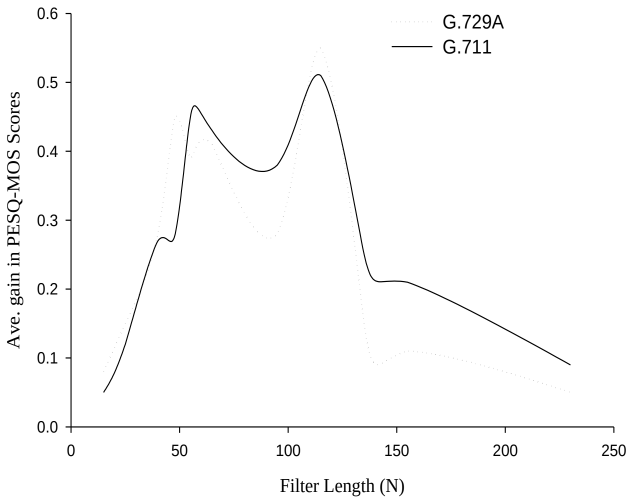

Figure 2 (c) β=1.5

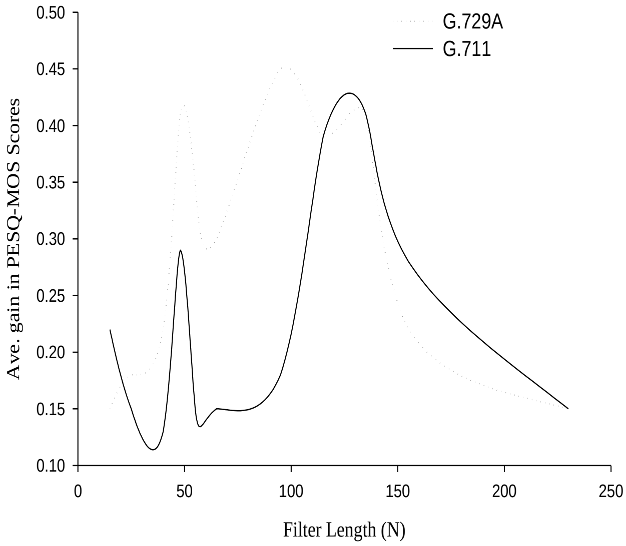

Figure 2 (d) β=2.0

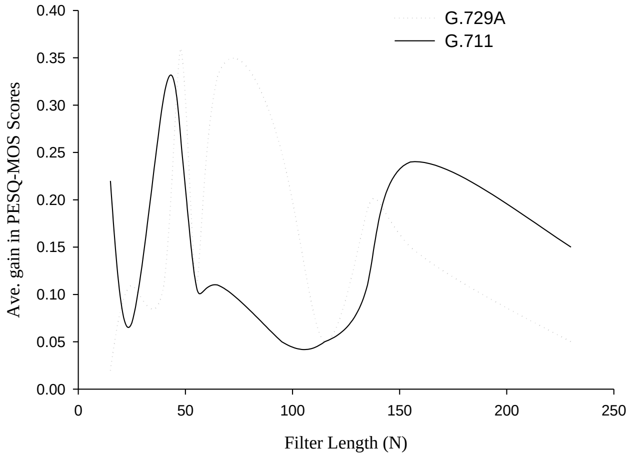

Figure 2 (e) β=2.5

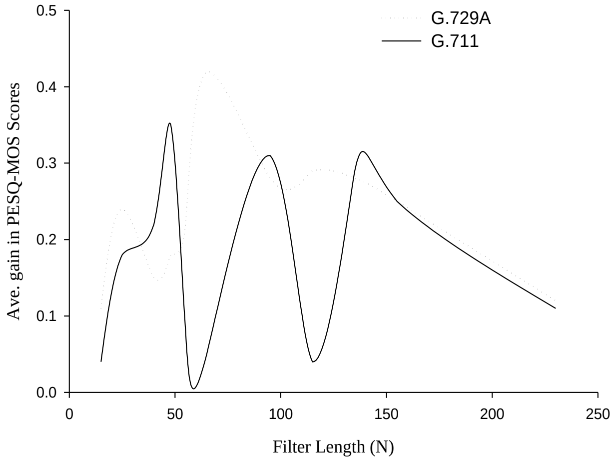

Figure 2 (f) β=3.0

Fig. 2 Effect of filter length (N) and Kaiser Window beta, β for G.711 and G.729A based VoIP system

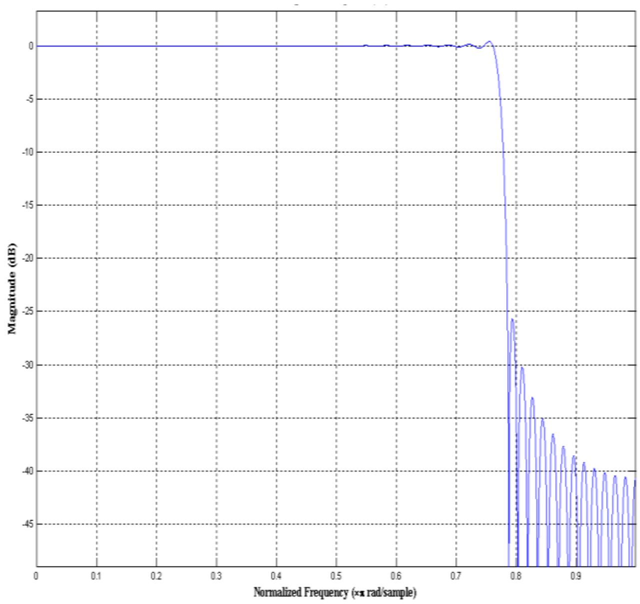

Figure. 3 Magnitude Response of proposed FIR filter

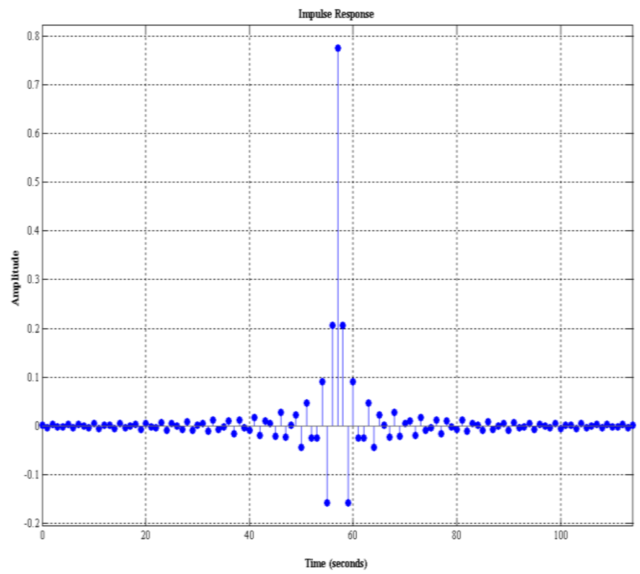

Figure. 4 Impulse Response of proposed FIR filter

B. *Effect of Packet Loss in VoIP system:*

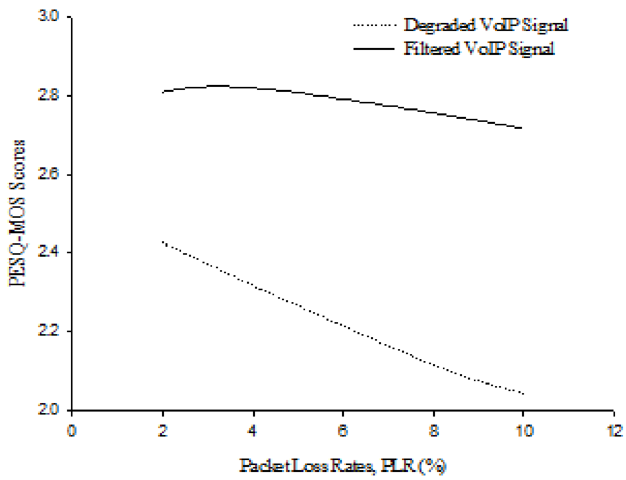

Figure: 5 (a) G.729A Packet Loss Rates vs. PESQ-MOS Scores

The speech signal is degraded with various packet loss rates at described in Table.1. The performance of the proposed FIR filter is analyzed for various loss rates for G.711 and G.729A based VoIP system. The variation of the packet loss rates and PESQ-MOS scores for G.729A and G.711 coders in VoIP system is shown in Fig. 5 (a) and Fig.5 (b). The significant increase in PESQ-MOS scores is achieved at various packet loss rates with the proposed scheme. The proposed scheme is much effective at high packet loss rates as presented in Fig. 5. The proposed scheme effectively conceals the lost packet during VoIP speech transmission to improve the signal quality.

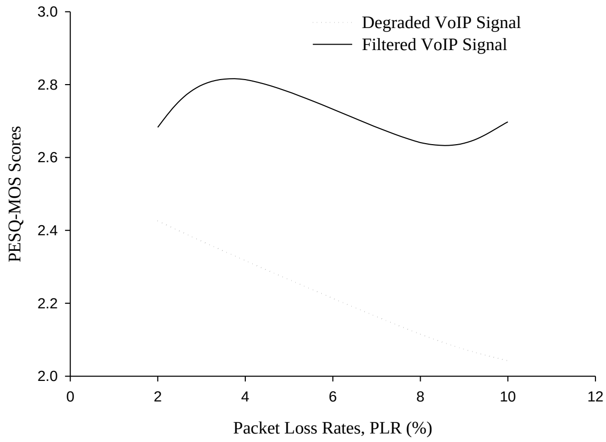

Figure: 5 (b)G.711 Packet Loss Rates vs. PESQ-MOS Scores

VI. CONCLUSION AND FUTURE WORK

The parametric analysis of FIR filtering scheme was performed on degraded narrowband VoIP speech in this work. The implementation of proposed filter on VoIP speech signal not only improves the speech quality but also try to retain the spectral shape of original signal. The comparison results presented in this paper proposed the suitable design parameters for FIR filter to improve the signal quality of the VoIP speech signal. In future the study can be used for improving the speech quality using various signal processing algorithms performed at higher frequency digital signal processors.

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