Trends in
Applied Sciences Research

ISSN 1819-3579

Academic
Journals Inc.

www.academicjournals.com

Trends in Applied Sciences Research 7 (5): 382-391, 2012
ISSN 1819-3579 / DOI: 10.3923/tasr.2012.382.391

Performance Analysis for VoIP System using Finite Impulse Response Algorithm in Noisy Environment

[1]H.P. Singh, [1]S. Singh and [2]J. Singh
[1]Department of Physics, Dr. B.R. Ambedkar National Institute of Technology, Jalandhar, India
[2]Department of Electronics Technology, Guru Nanak Dev University, Amritsar, India

Corresponding Author: H.P. Singh, Department of Physics, Dr. B.R. Ambedkar National Institute of Technology, Jalandhar, India

ABSTRACT

The speech signal is degraded due to delay, jitter and packet losses, when passed through different network layers of internet protocol based Voice over Internet Protocol (VoIP) system. The speech quality of the VoIP system can be improved by reducing the background noise, codec distortions and various network impairments such as packet loss, delay and jitter. The work in this paper proposes the Finite Impulse Response filter (FIR) based scheme for speech quality improvement of the VoIP signal. The lab experiment is performed to realize the VoIP system and to obtain the degraded database. For various noisy conditions, the performance of the enhanced VoIP signal is evaluated using Perceptual Evaluation of Speech Quality (PESQ) measurement for narrowband signal. The results show much improvement in speech quality with proposed method.

Key words: VoIP, signal processing, WANem, packet loss, FIR digital filter, speech coder

INTRODUCTION

Voice over Internet Protocol (VoIP) is an increasingly popular service for voice calls over IP based internet or local area networks (Ismail and Abusin, 2006; Goralski and Kolon, 2000). The current communication technology is shifting from circuit to packet switched voice communications (VoIP) due to additional features such as voice and video conferencing, text chat, caller ID, voice mail, call forwarding. The most favorable benefit of the VoIP system is that it costs as much as half the traditional Public Switched Telephone Network (PSTN) system in the field of voice transmission and this is because of the efficient use of available bandwidth for data packets (Altalhi *et al*., 2012; Lin *et al*., 2006; Varshney *et al*., 2002). Since the congestion in the network degrade the signal quality, to reduce the congestion in the communication network queuing algorithms (Alwakeel and Almansour, 2011; Baklizi *et al*., 2012; Karim *et al*., 2007) and congestion control algorithms (Babainejad *et al*., 2010; Sharma *et al*., 2006) were proposed in literature. But the quality of speech signal is one of the main problems in the implementation of the voice over IP, since the speech signal is processed through the internet protocols and internet protocols work on best effort based policy and lead to delay, jitter and packet loss during the communications. Due to these network impairments, the voice packets are lost during communication and degrade voice signal at the receiver side. Sanneck *et al*. (1996) proposed time scale modification algorithms to enhance the signal quality by regenerating the lost packets. Goodman *et al*. (1986) proposed the waveform substitution algorithms for signal quality improvements for Pulse Code Modulation (PCM) speech coders. The ITU-T G.729 and G.723.1 speech codecs were implemented on TMS320C6201 DSP processor and the optimization methods had used to reduce the speech processing time and with the optimizations, 20 and 18 voice channels concurrently could be processed for G.729 and G.723.1 codec respectively, with single TMS320C6201 chip implemented in IP telephony gateway (Yong-Feng and Jiang-Ling, 2000; Zhang *et al.*, 2000). The issue of noise reduction for VoIP speech coders using Wiener filter based noise reduction scheme was raised by Han *et al.* (2007) and the scheme 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.

PROPOSED SYSTEM

The noisy voice signal is when processed through the internet based network; the quality of the VoIP signal is further reduced due to delay, jitter and packet losses. The VoIP system is realized through the lab experiment (Radhakrishnan and Hadi, 2011). The speech signal is degraded with various network impairments at different rates. To enhance the signal quality, FIR filter based scheme is proposed. The FIR filters are implemented as post-processor on the degraded VoIP speech signal. The proposed system is presented in Fig. 1.

The simulation for VoIP system is performed under various network conditions. The basic steps in derivation of the proposed system are:

- **Step 1:** The noisy speech signals are fed in to the system. The speech samples for both male and female are taken from (http://www.utdallas.edu/~loizou/speech/noizeus/)
- **Step 2:** The speech signal is then encoded with G.711a, iLBC and Speex speech coders, which is the compressed version of the input signal. G.711 is the standard used for the communication purpose and is a high bit rate Pulse Code Modulation codec. It works at sampling rate of 8 kHz and compresses the 16 bit audio samples into 8 bits (ITU-T Recommendation G.711, 1988). The iLBC is also used for the packet network communications and it work at dual bit rates, 13.3 kbps with frame length of 30 msec and 15.20 kbps with frame length of 20 msec (Andersen *et al.*, 2004). The Code Excited Linear Prediction (CELP) based Speex codec is an open source codec developed for the packet network and VoIP applications (Valin, 2007). The Speex supports three different sampling rates narrowband (8 kHz), wideband (16 kHz) and ultra-wideband (32 kHz)
- **Step 3:** The compressed signal is then packetized into VoIP packets to transfer it to the IP network

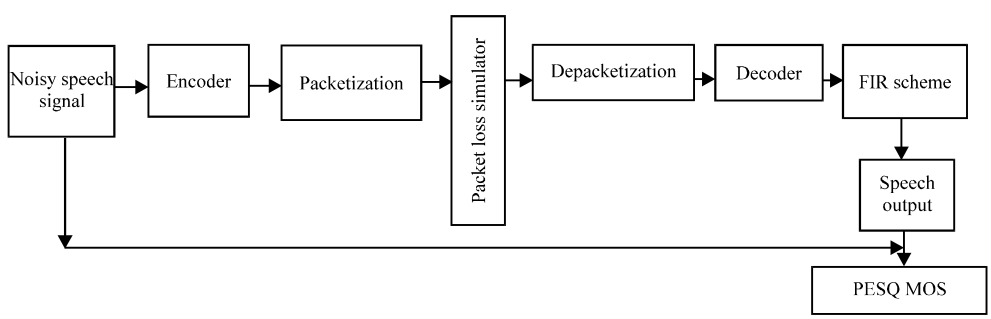

Fig. 1: Conceptual diagram of proposed system

- **Step 4:** The speech signal is degraded due to the various network impairments including delay, jitter and packet loss during VoIP communications. The network impairments are introduced through the lab experiment performed using WANem emulator (TCS WANem v 2.0, 2008)
- **Step 5:** The degraded VoIP signal is depacketized and then decoded with G.711a, iLBC and Speex decoders
- **Step 6:** The FIR scheme is incorporated into the system as post processor, after the decoder
- **Step 7:** The performance is evaluated with Perceptual Evaluation of Speech Quality (PESQ) measurement defined by ITU-T Recommendation P.862, 2001. The degraded signal is compared with the original signal and then PESQ measurement gives the subjective measurement as Mean Opinion Scores (MOS) value from -0.5 to 4.5

VoIP LAB EXPERIMENT

The lab experiment is performed to realize the VoIP system and different types of background noises including babble, car and street noise at SNR of 0, 5, 10, 15 dB are used as voice samples. The speech signal is further degraded due to the various network impairments including delay, jitter and packet loss during VoIP communications. The recommended one way delay by ITU-T is 0-150 msec and this amount of delay is acceptable for most of the user applications (ITU-T Recommendation G.114, 2003). The different values 0, 50, 100, 150 and 200 msec are used for delay in the performed lab experiment. Jitter is the variation in the delay content and can be created by queuing delays on the WAN link across network (Jelassi *et al.*, 2009). The lab experiment is performed at five different values 0, 5, 7, 10 and 15 msec for jitter. The packet loss is the percentage of the lost packet during the transportation due to various network conditions such as buffer overflow, network congestion etc. The delay and jitter also contribute to the packet losses and these results in harmful effects on the quality of VoIP signal. 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. Even a single lost packet may generate audible distortions in the decoded speech signal (Singh *et al.*, 2010). The different values of the Packet Loss Rate (PLR) 3 to 10% are used during the lab experiment. Since packet loss is the most difficult problem in VoIP system, the present work has been focused on the packet loss issue. To reduce the effect of packet loss on perceived speech quality, the lost packets have to be regenerated at the receiver. The WANem emulation software is used to emulate the WAN traffic. The voice traffic is degraded with various network conditions such as delay, jitter and packet loss as discussed. The X-Lite IP softphone (X-lite, 2010) is used to establish VoIP calls between two computers. 3CX phone system (3CX, 2010) is used as the SIP server to provide VoIP services. The performance is evaluated with PESQ measurement defined by ITU-T recommendation P.862.

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 (Jackson, 1996). The basic FIR filter is characterized by:

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

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:

$$y\ (n) = x\ (n)^*\ h\ (n)$$
$$y\ (n) = x\ (0)\ h\ (n) + x\ (1)\ h\ (n-1) + x\ (2)\ h\ (n-2) + \ldots + x\ (n)\ h\ (0) \tag{2}$$

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

$$H(e^{j\omega}) = \sum_{n=0}^{N-1} h(n)\, e^{-j\omega nt} = \left|H(e^{j\omega})\right| e^{j\Phi(\omega)} \tag{3}$$

The magnitude and phase responses are given by:

$$\begin{aligned} M(\omega) &= \left|H(e^{j\omega})\right| = \sqrt{\mathrm{Re}[H(e^{j\omega})]^2 + \mathrm{Im}[H(e^{j\omega})]^2} \\ \Phi(\omega) &= \tan^{-1} \frac{\mathrm{Im}[H(e^{j\omega})]}{\mathrm{Re}[H(e^{j\omega})]} \end{aligned} \tag{4}$$

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

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

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:

$$W(n) = \begin{cases} \dfrac{I_O\left[\beta\sqrt{1-(1-\frac{2n}{M})^2}\right]}{I_O(\beta)} & n = 0, 1, \ldots, M \\ 0, & \text{otherwise} \end{cases} \tag{6}$$

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 (Li, 2010; Reddy *et al*., 2008). The proposed FIR scheme for

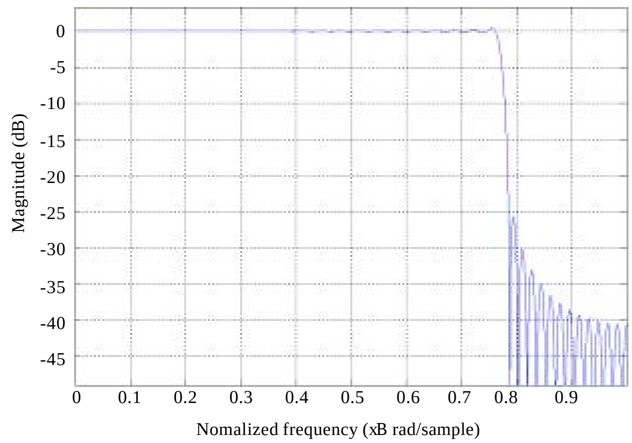

Fig. 2: Frequency response of FIR filter

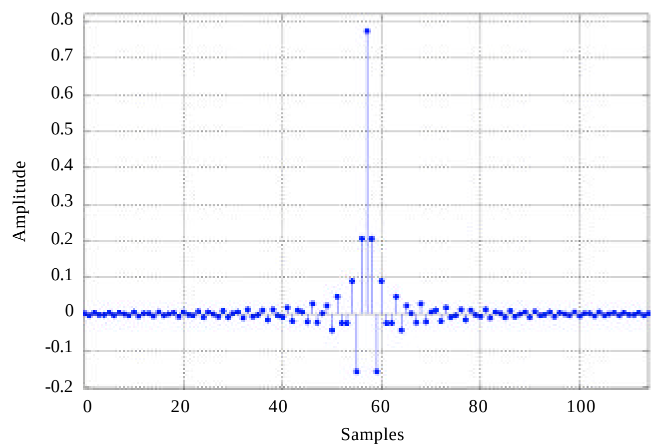

Fig. 3: Impulse response of FIR filter

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 frequency response and impulse response of the proposed filter is presented in Fig. 2 and 3, respectively.

RESULTS

The performance results of the proposed system are evaluated for packet loss rates, Kaiser Window beta factor and for various noise types used in this study and are discussed here.

Evaluation of proposed scheme at variation of packet loss rates: The performance of the proposed system is evaluated for VoIP system at different packet loss rates varying from 3 to 10%. The average PESQ-MOS scores are taken at different packet loss rates for G.711a, iLBC and Speex coders and plotted in Fig. 4. The MOS scores of VoIP signal with G.711a coder decreases as the packet loss rate increases. The more specific coders for VoIP communication, iLBC and Speex gives much better performance than the G.711a coder at high packet loss rates. The proposed filtering

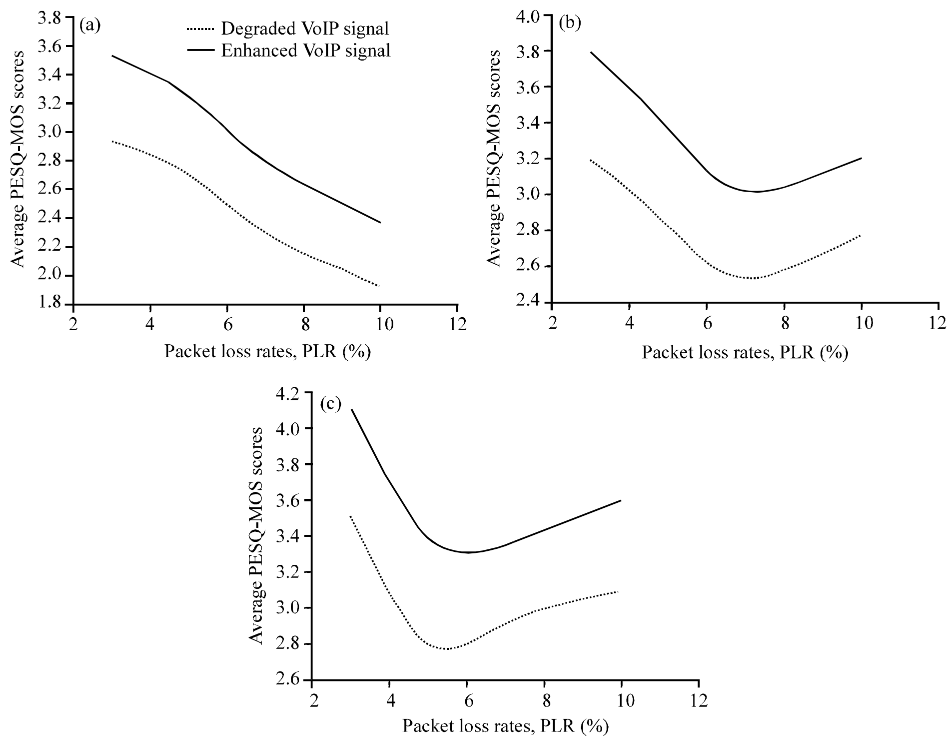

Fig. 4(a-c): Performance evaluation of FIR filtering at varying loss rates; (a) G.711a, (b) iLBC and (c) Speex

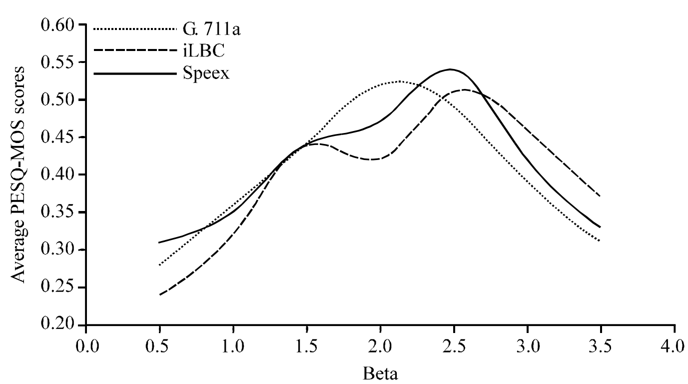

Fig. 5: Effect of beta factor on VoIP system

scheme is very much effective for all codec used in this work. With the application of the proposed scheme, the significant increment in MOS scores is achieved in VoIP system, even at higher loss rates too.

Study of effect of beta factor (β) for VoIP system: At the optimum value of the filter length of 115, the performance of the proposed filter is analyzed at various values of the Kaiser Window beta factor (β) ranging from 0 to 3.5, since beyond this value the average gain in MOS scores falls rapidly. The variation of β factor with average gain in PESQ-MOS scores for all speech codecs is plotted in Fig. 5. The proposed scheme for VoIP system gives much better results in beta range of

Table 1: Comparison results for G.711a

		Noise type								
		Babble noise			Car noise			Street noise		
SNR	PLR (%)	None	Proposed	Gain in PESQ	None	Proposed	Gain in PESQ	None	Proposed	Gain in PESQ
0 dB	3	2.5	3.05	0.55	2.8	3.42	0.62	2.2	2.85	0.65
	5	2.3	2.81	0.51	2.5	3.07	0.57	2.1	2.71	0.61
	7	1.9	2.39	0.49	2.1	2.61	0.51	1.8	2.35	0.55
	10	1.5	1.96	0.46	1.7	2.18	0.48	1.5	1.99	0.49
5 dB	3	2.8	3.37	0.57	3.1	3.68	0.58	2.6	3.22	0.62
	5	2.6	3.12	0.52	2.8	3.32	0.52	2.4	2.95	0.55
	7	2.1	2.62	0.52	2.2	2.67	0.47	2.1	2.58	0.48
	10	1.8	2.28	0.48	1.9	2.31	0.41	1.7	2.12	0.42
10 dB	3	3.1	3.74	0.64	3.4	3.99	0.59	2.9	3.42	0.52
	5	2.9	3.49	0.59	3.1	3.65	0.55	2.6	3.08	0.48
	7	2.5	3.04	0.54	2.8	3.29	0.49	2.2	2.65	0.45
	10	2.1	2.59	0.49	2.4	2.85	0.45	2.1	2.48	0.38
15 dB	3	3.4	4.01	0.61	3.3	3.94	0.64	3.1	3.67	0.57
	5	3.1	3.68	0.58	3.1	3.68	0.58	2.9	3.39	0.49
	7	2.6	3.11	0.51	2.8	3.34	0.54	2.4	2.85	0.45
	10	2.2	2.68	0.48	2.2	2.68	0.48	1.8	2.21	0.41

Table 2: Comparison results for iLBC

		Noise type								
		Babble noise			Car noise			Street noise		
SNR	PLR (%)	None	Proposed	Gain in PESQ	None	Proposed	Gain in PESQ	None	Proposed	Gain in PESQ
0 dB	3	2.8	3.42	0.62	2.9	3.51	0.61	3.0	3.59	0.59
	5	2.6	3.17	0.57	2.4	2.95	0.55	2.9	3.45	0.55
	7	2.6	3.11	0.51	2.4	2.89	0.49	2.1	2.61	0.51
	10	2.7	3.18	0.48	2.7	3.11	0.41	2.5	2.96	0.46
5 dB	3	2.7	3.28	0.58	2.8	3.42	0.62	3.2	3.77	0.57
	5	2.4	2.92	0.52	2.6	3.15	0.55	2.9	3.42	0.52
	7	2.5	2.97	0.47	2.1	2.58	0.48	2.2	2.72	0.52
	10	2.8	3.25	0.45	2.4	2.79	0.39	2.6	3.08	0.48
10 dB	3	3.6	4.19	0.59	3.4	3.92	0.52	3.3	3.94	0.64
	5	3.2	3.75	0.55	3.2	3.65	0.45	2.8	3.39	0.59
	7	2.4	2.89	0.49	2.4	2.82	0.42	2.4	2.91	0.51
	10	2.8	3.25	0.45	2.9	3.28	0.38	3.1	3.56	0.46
15 dB	3	3.7	4.34	0.64	3.6	4.17	0.57	3.3	3.91	0.61
	5	3.6	4.18	0.58	2.9	3.39	0.49	2.4	2.98	0.58
	7	3.2	3.74	0.54	3.2	3.61	0.41	2.8	3.31	0.51
	10	2.9	3.31	0.41	2.7	3.08	0.38	3.2	3.61	0.41

2.0-2.5. For G.711a, the maximum average gain in MOS scores is found at beta factor 2.0. The peak value of average gain in MOS scores is found at 2.5 beta factor for both iLBC and Speex coders.

Study of effect of noise type for VoIP system: The average PESQ-MOS gain for all speech coders at various values of SNR for different types of noise is described in Table 1-3. The proposed

Table 3: Comparison results for Speex

		Noise type								
		Babble noise			Car noise			Street noise		
SNR	PLR (%)	None	Proposed	Gain in PESQ	None	Proposed	Gain in PESQ	None	Proposed	Gain in PESQ
0 dB	3	3.3	3.89	0.59	3.1	3.69	0.59	3.4	3.97	0.57
	5	3.1	3.67	0.57	2.7	3.27	0.57	2.9	3.44	0.54
	7	2.8	3.35	0.55	2.8	3.31	0.51	2.2	2.78	0.58
	10	2.6	3.09	0.49	2.7	3.18	0.48	3.6	4.24	0.64
5 dB	3	3.4	4.02	0.62	3.3	3.88	0.58	3.4	4.02	0.62
	5	2.9	3.45	0.55	2.3	2.82	0.52	2.9	3.51	0.61
	7	3.3	3.78	0.48	2.4	2.87	0.47	3.4	3.96	0.56
	10	3.1	3.52	0.42	2.8	3.25	0.45	3.1	3.79	0.69
10 dB	3	3.6	4.12	0.52	3.8	4.39	0.59	3.6	4.25	0.65
	5	2.8	3.28	0.48	2.7	3.25	0.55	2.9	3.49	0.59
	7	2.6	3.05	0.45	3.2	3.69	0.49	3.2	3.74	0.54
	10	3.2	3.58	0.38	3.4	3.85	0.45	3.1	3.61	0.51
15 dB	3	3.9	4.47	0.57	3.6	4.24	0.64	3.7	4.37	0.67
	5	2.7	3.19	0.49	2.9	3.48	0.58	2.8	3.39	0.59
	7	3.2	3.65	0.45	2.6	3.14	0.54	3.3	3.82	0.52
	10	3.4	3.81	0.41	3.2	3.61	0.41	2.9	3.41	0.51

FIR scheme is very much effective for street noise at low SNR for VoIP signal coded with G.711a coder and at high SNR values, scheme is better for babble and car noises as described in Table 1. For iLBC coder, the proposed scheme gives much better results for all types of noise used in this work at low SNR and at high SNR, the MOS scores are increased for babble & street noises as predicted in Table 2. The significant increment is observed for street noise at all SNR values with the proposed scheme for Speex coder as presented in Table 3.

CONCLUSION

The work in this study proposed the use of finite impulse response filter for noise reduction in VoIP system. The proposed FIR filter was applied as post processor in VoIP system. The results showed that the proposed FIR filtering scheme was very much effective in all type of noise used in this work at different SNR and much better results were observed in case of the VoIP impairments such as packet loss. The performance evaluation study of the proposed system showed that the average increase of 0.52 in PESQ-MOS was observed for G.711a codec based VoIP system in noisy environment. For experiment with iLBC and Speex coders based VoIP system, the average gain of 0.51 and 0.54 in PESQ-MOS scores was achieved. The proposed filtering scheme significantly reduced the distortions in the VoIP signal and enhanced the signal quality. Thus the proposed scheme can be efficiently used for VoIP applications. In future the study can be conducted for different other VoIP wideband speech coders. The study can be used for improving the speech quality using various signal processing algorithms performed at faster digital signal processors.

ACKNOWLEDGMENT

The authors are thankful to Ministry of Human Resource and Development of India for carrying out this study.

REFERENCES

3CX, 2010. Software based PBX. http://www.3cx.com/

Altalhi, A.H., M.S.B.M. Azmi, A.M. Al-Kharasani and S.A. Ali, 2012. A modified packet marking algorithm to improve bandwidth fairness in diffserv networks. Res. J. Inform. Technol., 4: 1-11.

Alwakeel, S.S. and H.M. Almansour, 2011. Modeling and performance evaluation of message-oriented middleware with priority queuing. Inform. Technol. J., 10: 61-70.

Andersen, S., H. Astrom, R. Hagen, W. Kleijn and J. Linder, 2004. Internet Low bit rate codec (iLBC). http://tools.ietf.org/pdf/rfc3951.pdf

Babainejad, S., S. Babainejad, N. Bigdeli and K. Afshar, 2010. Setting up a virtual laboratory for evaluation of congestion control algorithms in TCP/IP networks. Trends Applied Sci. Res., 5: 177-187.

Baklizi, M., H. Abdel-Jaber, S. Ramadass, N. Abdullah and M. Anbar, 2012. Performance assessment of AGRED, RED and GRED congestion control algorithms. Inform. Technol. J., 11: 255-261.

Goodman, D., G. Lockhart, O. Wasem and W.C. Wong, 1986. Waveform substitution techniques for recovering missing speech segments in packet voice communications. Speech, Signal Process., 34: 1440-1448.

Goralski, J.W. and M.C. Kolon, 2000. IP Telephony. McGraw-Hill Inc., New York, USA., ISBN-13: 9780071352215, Pages: 468.

Han, S., S. Jeong, H. Yang and J. Kim, 2007. Noise Reduction for VoIP Speech Codecs using Modified Wiener Filter. In: Advances and Innovations in Systems, Computing Sciences and Software Engineering, Elleithy, K. (Ed.). Springer, New York, USA., ISBN-13: 9781402062636, pp: 393-397.

ITU-T Recommendation P.862, 2001. Perceptual Evaluation of Speech Quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codec. http://www.catr.cn/radar/itut/201007/P020100707479400910965.doc

ITU-T Recommendation G.114, 2003. One way transmission time. http://www.cs.columbia.edu /~andreaf/new/documents/other/T-REC-G.114-200305.pdf

ITU-T Recommendation G.711, 1988. Pulse Code Modulation (PCM) of voice frequencies. http://www.analytic.ru/articles/lib228.pdf

Ismail, M.N. and A. Abusin, 2006. Prepaid and postpaid VoIP service enhancements and hybrid network performance measurement. Inform. Technol. J., 5: 386-400.

Jackson, L.B., 1996. Digital Filters and Signal Processing: With MATLAB Exercises. 3rd Edn., Spronger, New York, USA., ISBN-13: 9780792395591, Pages: 502.

Jelassi, S., H. Youssef, C. Hoene and G. Pujolle, 2009. Parametric speech quality models for measuring the perceptual effect of the network delay jitter. Proceedings of the 34th IEEE Conference on Local Computer Networks, October 20-23, 2009, Zurich, pp: 193-200.

Karim, S., I. Islam, A. Rab, R. Hasan and M.R. Amin, 2007. Development of deterministic service time traffic model for packet communication. Inform. Technol. J., 6: 160-165.

Li, C., 2010. Design and realization of FIR digital filters based on MATLAB. Proceedings of IEEE International Conference on Anti-Counterfeiting, Security and Identification, July 18-20, 2010, Chengdu, Sichuan, China, pp: 101-104.

Lin, J.C., S.K.C. Lo, H.C. Keh, S.J. Wu and Y.H. Lin, 2006. The intelligent agent of the billing service between the PSTN and VoIP. Inf. Technol. J., 5: 51-56.

Radhakrishnan, K. and L. Hadi, 2011. Evaluating perceived voice quality on packet networks using different random neural network architectures. Perform. Eval., 68: 347-360.

Reddy, G.H., Y.V. Reddy and S.N. Reddy, 2008. The effect of β in Kaiser window on the SNR of MST radar signals. Asian J. Scientific Res., 1: 203-212.

Sanneck, H., A. Stenger, K.B. Younes and B. Girod, 1996. A new technique for audio packet loss concealment. Proceedings of the IEEE Global Telecommunication Conference Communications, November 18-22, 1996, London, UK, pp: 48-52.

Sharma, S., R.C. Jain and S.S. Bhadauria, 2006. Simulation study of congestion adaptive routing algorithm for mobile Ad hoc network. Trends Applied Sci. Res., 1: 368-378.

Singh, H.P., S. Singh and J. Singh, 2010. Computer modeling and performance analysis of VoIP under different strategic conditions. Int. Conf. Comput. Eng. Appl., 1: 611-615.

TCS WANem v 2.0, 2008. WANem: Wide area network emulator. http://sourceforge.net /projects/wanem/files/Documents/

Valin, J.M., 2007. The Speex codec manual version 1.2 beta 3. http://www.speex.org /docs/manual/speex-manual.pdf

Varshney, U., A. Snow, M. McGivern and C. Howard, 2002. Voice over IP. Commun. ACM, 45: 89-96.

X-lite, 2010. CounterPath X-lite. http://www.counterpath.com/x-lite.html

Yong-Feng, H. and Z. Jiang-Ling, 2000. Implementation of ITU-T G.729 speech codec in IP telephony gateway. Wahan Univ. J. Nat. Sci., 5: 159-163.

Zhang, Y., F. Huang, Y. Juan and S.Z. Jiang, 2000. Implementation of ITU-T G.723.1 dual rate speech codec based on TMS320C6201 DSP. Proceedings of International Conference on Signal Processing, August 21-5, 2000, Beijing, China, pp: 679-682.