International Journal of Electrical, Electronic and Communication Sciences
ISSN: 2517-9438
Vol:6, No:11, 2012

Computationally Efficient Signal Quality Improvement Method for VoIP System

H. P. Singh, and S. Singh

Abstract**—**The voice signal in Voice over Internet protocol (VoIP) system is processed through the best effort policy based IP network, which leads to the network degradations including delay, packet loss jitter. The work in this paper presents the implementation of finite impulse response (FIR) filter for voice quality improvement in the VoIP system through distributed arithmetic (DA) algorithm. The VoIP simulations are conducted with AMR-NB 6.70 kbps and G.729a speech coders at different packet loss rates and the performance of the enhanced VoIP signal is evaluated using the perceptual evaluation of speech quality (PESQ) measurement for narrowband signal. The results show reduction in the computational complexity in the system and significant improvement in the quality of the VoIP voice signal.

Keywords**—**VoIP, Signal Quality, Distributed Arithmetic, Packet Loss, Speech Coder.

I. Introduction

VOICE over IP is a technology that enables the users to transmit voice over the internet or a local area network using internet protocol (IP).This technology provides benefits such as low cost compared to the traditional Public Switched Telephone Network (PSTN). With the efficient use of bandwidth requiring fewer long-distance trunks between switches, VoIP system costs as much as half the traditional PSTN system in the field of voice transmission [1]. Packet switched networks like the 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 the 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 [2].

H P Singh is with Department of Physics, Dr B R Ambedkar National Institute of Technology, Jalandhar, Punjab, India (phone: 9101812690301; e-mail: harjit_nit@yahoo.co.in).
S Singh is with Department of Physics, Dr B R Ambedkar National Institute of Technology, Jalandhar, Punjab, India (phone: 9101812690301).

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 differs that window based finite impulse response filters are applied to improve the signal quality in narrowband VoIP system through distributed arithmetic algorithm to improve the signal quality and reduce the system complexity. The brief description of the related work is presented in Section II. The section III describes the design of the digital filters using distributed arithmetic algorithm for the VoIP system. The modeling of the IP network and VoIP simulations are presented in Section IV. The performance analysis results and discussion are presented in Section V. The last section concludes the work and presents the future work.

II. Related Work

The lost packets were also regenerated with the use of time scale modification algorithms [3, 4]. Feng et.al implemented ITU-T G.729 and G.723.1 speech codecs on TMS320C6201 DSP processor and used the optimization methods used to reduce the speech processing time. With the optimizations applied, G.729 codec was able to process concurrently 20 voice channels and G.723.1 codec able to process 18 voice channels with a single TMS320C6201 chip in IP telephony gateway [5, 6]. Han et.al [7] had raised the issue of noise reduction for VoIP speech codecs and proposed a modified Wiener filter based noise reduction scheme and applied the proposed system as preprocessing before the speech encoding. Several multiplier-less schemes had been proposed. These methods can be classified in two categories according to how they manipulate the filter coefficients for the multiply operation. The first type of multiplier-less technique is the conversion-based approach, in which the coefficients are transformed to other numeric representations whose hardware implementation or manipulation is more efficient than the traditional binary representation. The example of such techniques are the Canonic Sign Digit (CSD) method, in which coefficients are represented by a combination of powers of two in such a way that multiplication can be simply implemented with adder/subtractors and shifters [8], and the Dempster-Mcleod method, which similarly involves the representation of filter coefficients with powers of two and in this case arranging partial results in cascade to introduce further savings in the usage of adders [9]. The second type of multiplier-less methods involve the use of memories (RAMs, ROMs) or Look-Up Tables (LUTs) to store pre-computed values of coefficient operations. These memory-based methods involve Constant Coefficient Multiplier method and the very-well known Distributed Arithmetic method [10].

III. Distributed Arithmetic for 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 [11-13]. 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 z transform of an N-point FIR filter is given by:

$$H(Z)=\sum_{n=0}^{N-1}h(k)Z^{-k} \tag{4}$$

The major advantages of using the 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

$$W(n)=\begin{cases}\dfrac{I_o(\beta)\sqrt{1-(2(n+1)(N+1))^2}}{I_o(\beta)} & n=0,1\ldots N\\ 0, & otherwise\end{cases} \tag{5}$$

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 [13]. The proposed FIR scheme for VoIP speech signal improvement is designed using the MATLAB. The performance of FIR scheme is analyzed for the 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.

Alternatively, Distributed Arithmetic (DA) is an important algorithm used in computing the sum of the products:

$$y[n]=<h,x>=\sum_{n=0}^{N-1}h[n]x[n] \tag{6}$$

DA system, assumes that the variable $x[n]$ is represented by:

$$x[n]=\sum_{b=0}^{B-1}x_b[n]\times 2^b,\; x_b[n]\in[0,1] \tag{7}$$

If $h[n]$ are the coefficients of the FIR filter, then the output of FIR filter in the bit level form is:

$$y=\sum_{n=0}^{N-1}h[n]\times\sum_{b=0}^{B-1}x_b[n]\times 2^b \tag{8}$$

In distributed arithmetic form

$$y=\sum_{b=0}^{B-1}2^b\times\sum_{n=0}^{N-1}f(h[n],x_b[n]) \tag{9}$$

The output in distributed arithmetic form:

$$y=-(\sum_{n=0}^{N-1}h[n]x_b[0])+\sum_{b=1}^{B-1}(\sum_{n=0}^{N-1}h[n]x_b[n])2^{-b} \tag{10}$$

The modified filter coefficients are obtained by rounding the filter coefficients to the nearest integer after being multiplied with a constant integer value. The multiplied constant was chosen to be a power of 2, in such a way that the new modified filter confidents must contain at least 93 % of the signal power [14]. The similar time and frequency responses were obtained with modified coefficients as were obtained with the original coefficients. The frequency, phase and impulse responses of the original and modified filter coefficients are presented in the Fig.1- Fig.6. The filter coefficients obtained with the traditional scheme were 115 but with the proposed DA algorithm, the number of filter coefficients became 49. Since the FIR filter was designed with multiplier-less distributed arithmetic scheme, therefore the number of adders and addition operations must be a crucial point in the system computational complexity. The similar filter responses were obtained at the reduced computational complexity in the system.

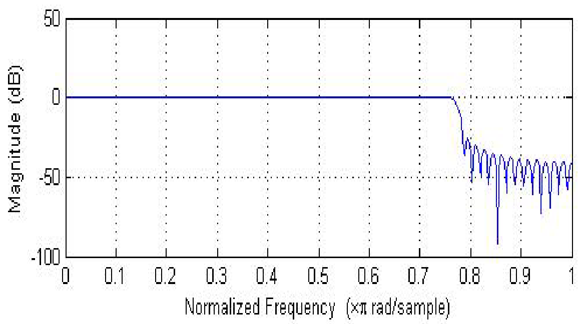

Fig. 1 Frequency Response of FIR filter with original Coefficients

International Journal of Electrical, Electronic and Communication Sciences
ISSN: 2517-9438
Vol:6, No:11, 2012

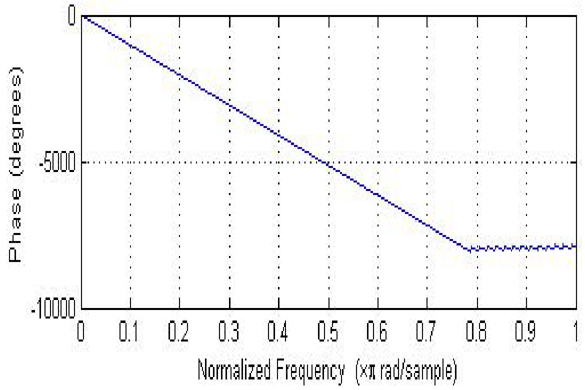

Fig. 2 Phase Response of FIR filter with original Coefficients

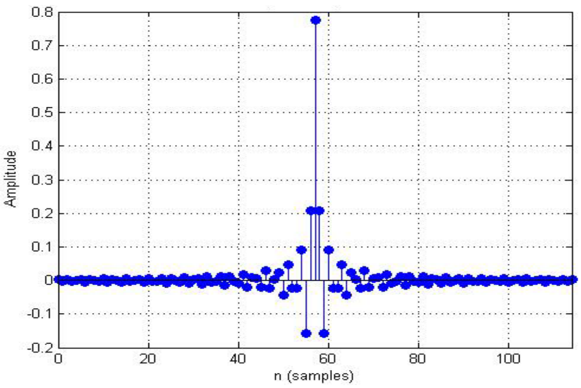

Fig. 3 Impulse Response of FIR filter with original Coefficients

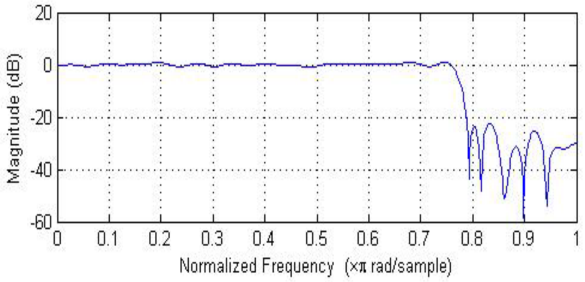

Fig. 4 Frequency Response of FIR filter with Modified Coefficients

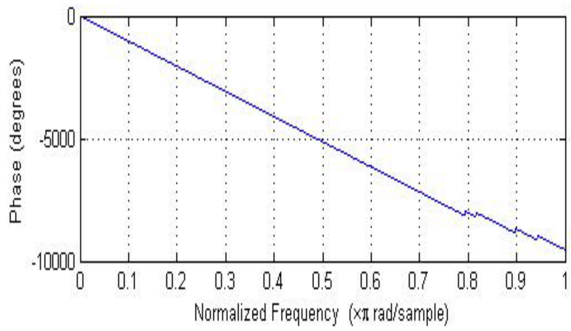

Fig. 5 Phase Response of FIR filter with Modified Coefficients

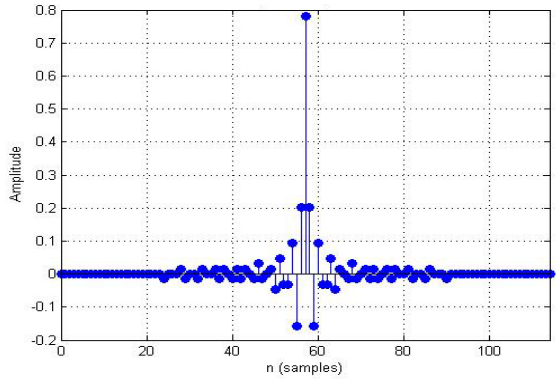

Fig. 6 Impulse Response of FIR filter with Modified Coefficients

TABLE I
SIMULATED LOSS RATES

PLR (%)	p	q
2	0.0032	0.15
4	0.012	0.25
8	0.025	0.25
10	0.10	0.85

IV. IP NETWORK MODELING

To analyze the performance of the proposed scheme in the VoIP system, the system simulations were performed in varying conditions. The simulation of VoIP system was performed where each packet contains one frame. The packet losses are not independent on a frame-by-frame basis, but appear in bursts. The packet loss can be approximated by a Markovian loss model such as Gilbert model, as discussed by Bolot in [15]. Most research for VoIP networks uses a Gilbert Model to represent packet loss characteristics, since the complexity is increased through the higher order Markov models [16]-[21]. Thus the simulation of IP network was performed by using a 2-state Gilbert Model. The model has two states reflecting whether the previous packet is received or lost.

International Journal of Electrical, Electronic and Communication Sciences
ISSN: 2517-9438
Vol:6, No:11, 2012

TABLE II
PESQ RESULTS FOR AMR-NB 6.70 KBPS FOR VOIP SYSTEM IN NOISY ENVIRONMENT

SNR (dB)	PLR (%)	Noise Type								
		Babble Noise			Car Noise			Street Noise		
		None	FIR Filter	DAFIR Filter	None	FIR Filter	DAFIR Filter	None	FIR Filter	DAFIR Filter
0 dB	2	1.40	1.91	1.89	1.41	1.96	1.93	1.37	2.02	1.98
	4	1.22	1.71	1.70	1.24	1.78	1.75	1.42	2.03	2.00
	8	1.05	1.53	1.52	0.87	1.38	1.36	0.91	1.46	1.43
	10	0.85	1.31	1.30	0.96	1.44	1.42	0.91	1.40	1.39
5 dB	2	1.66	2.17	2.15	1.49	2.01	1.98	1.68	2.30	2.26
	4	1.45	1.97	1.96	1.41	1.93	1.92	1.42	1.97	1.96
	8	1.23	1.69	1.67	1.09	1.56	1.53	1.18	1.66	1.64
	10	1.08	1.52	1.50	0.88	1.29	1.27	1.18	1.60	1.59
10 dB	2	1.80	2.35	2.31	1.82	2.37	2.33	1.89	2.41	2.38
	4	1.62	2.16	2.13	1.67	2.22	2.15	1.67	2.15	2.13
	8	1.31	1.85	1.80	1.40	1.89	1.85	1.43	1.88	1.88
	10	1.21	1.70	1.69	1.21	1.66	1.65	1.31	1.69	1.66
15 dB	2	2.05	2.66	2.63	1.88	2.47	2.43	2.02	2.49	2.57
	4	1.83	2.41	2.38	1.73	2.28	2.28	1.80	2.49	2.26
	8	1.50	2.01	1.96	1.34	1.88	1.88	1.61	2.06	2.05
	10	1.43	1.91	1.88	1.28	1.74	1.74	1.42	1.83	1.80

TABLE III
PESQ RESULTS FOR G.729A FOR VOIP SYSTEM IN NOISY ENVIRONMENT

SNR (dB)	PLR (%)	Noise Type								
		Babble Noise			Car Noise			Street Noise		
		None	FIR Filter	DAFIR Filter	None	FIR Filter	DAFIR Filter	None	FIR Filter	DAFIR Filter
0 dB	2	1.54	2.15	2.12	1.44	2.03	2.01	1.45	2.06	2.04
	4	1.18	1.77	1.73	1.37	1.94	1.92	1.37	1.91	1.91
	8	1.19	1.74	1.72	1.29	1.84	1.82	1.06	1.64	1.62
	10	1.11	1.60	1.59	1.16	1.64	1.64	0.99	1.63	1.57
5 dB	2	1.67	2.29	2.28	1.63	2.21	2.20	1.63	2.25	2.22
	4	1.49	2.04	2.02	1.48	2.02	2.01	1.57	2.18	2.15
	8	1.44	1.92	1.89	1.39	1.88	1.86	1.40	1.96	1.99
	10	1.29	1.73	1.71	1.25	1.71	1.70	1.30	1.99	1.95
10 dB	2	1.87	2.42	2.40	1.92	2.51	2.51	1.89	2.54	2.53
	4	1.80	2.28	2.26	1.78	2.33	2.31	1.78	2.37	2.37
	8	1.50	1.95	1.93	1.57	2.06	2.04	1.60	2.14	2.14
	10	1.39	1.83	1.80	1.49	1.94	1.94	1.48	1.99	1.99
15 dB	2	2.03	2.64	2.62	2.02	2.66	2.63	2.02	2.69	2.67
	4	1.84	2.33	2.32	1.85	2.43	2.40	1.94	2.53	2.53
	8	1.72	2.17	1.17	1.76	2.30	2.29	1.68	2.20	2.20
	10	1.56	2.00	1.99	1.61	2.06	2.04	1.61	2.12	2.12

The state "0" represents that a packet being correctly received and state "1" represents that a packet being lost. The Gilbert model is shown in Fig 7.

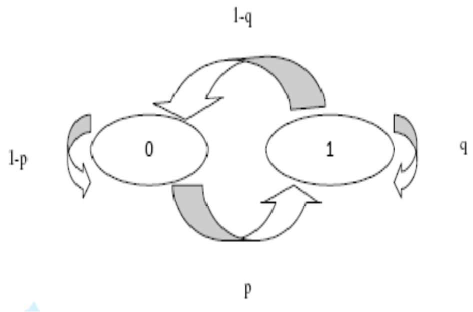

Fig. 7 Two State Gilbert Model

Let p be the transition probability of the network model to drop a packet given that the previous packet is delivered i.e. the probability for the 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. Let $p0$ and $p1$ denote the probability of the network model to be in state 0 and 1.The probability for a packet to be dropped regardless whether the previous packet is delivered or dropped i.e. the unconditional loss probability is exactly the probability for the network model to be in state 1 $(p1)$.

$$p0 = \frac{q}{p+q}, \quad p1 = \frac{p}{p+q} \tag{11}$$

The transition matrix is given as

$$P = \begin{pmatrix} 1-p & p \\ q & 1-q \end{pmatrix} \tag{12}$$

V. Simulation Results in Noisy Environment

The simulation for IP system is performed in noisy environments with different types of noise (babble, car and street) at SNR of 0, 5, 10, 15 dB. The speech samples for simulations were taken from [22]. The noisy speech signal is encoded into VoIP frames using AMR-NB 6.70 kbps [23] and G.729a [24] for narrowband VoIP system. The network configurations were introduced into VoIP frames with the modeling of the IP network through Gilbert model. The VoIP packet is dropped if it is in the "Error" state otherwise it is retained. 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 I. Then resulting stream of frames decoded using G.729A decoder. The performance is evaluated with Perceptual Evaluation of Speech Quality (PESQ) measurement defined by ITU-T recommendation P.862 [25] 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.

A. Discussion

The FIR filter with original and modified filter coefficients was implemented on the degraded VoIP speech signal. The results of the simulation for the proposed scheme in the VoIP system with AMR-NB 6.70 kbps and G.729a coders in noisy environments are presented in Table II, Table III. The average increase of 0.51 and 0.55 was achieved in PESQ MOS scores of the degraded VoIP signal with the implementation of the FIR filter with original coefficients. Comparatively, the average increase of the 0.49 and 0.53 was obtained in PESQ MOS scores of the degraded VoIP speech signal with the implementation of DAFIR filter, which has the reduced computational complexity, for AMR-NB 6.70 kbps and G.729a coder respectively. Approximately, the same level of signal quality improvement had been achieved with the implementation of the DAFIR on degraded VoIP speech signal as was achieved with a traditional FIR filter, but at the reduced computational complexity of the system.

VI. Conclusion & Future Work

The work presented in the paper proposed the use of the distributed arithmetic algorithm in FIR filter for signal quality improvement of the VoIP speech signal. The results revealed that the improvement in the VoIP speech signal was achieved almost up to the same quality level as obtained with the implementation of the FIR filter with original coefficients, but at the advantage of the reduced system computational complexity. The implementation of the proposed filter on VoIP speech signal not only improves the speech quality but also try to retain the spectral shape of the original signal. In future, the study can be performed on faster digital signal processors such as TMS320C6713 to improve the signal quality of VoIP system.

References

[1] Goralski, J.W., Kolon, C.M., IP Telephony, 1st ed., McGraw-Hill, 2000.
[2] Varshney, U., Snow, A., Mcgivern, M., Howard, C., "Voice over IP", Communications of the ACM, 2002, vol.45,no.1, pp.89-96
[3] Verhelst, W., Roelands, M., "An overlap-add technique based on waveform similarity (WSOLA) for high quality time-scale modification of speech", In Proceedings of IEEE International Conference on Acoustics, Speech and Signal Processing, 1993, vol.2, pp. 554-557.
[4] Stenger, A., Younes, K.B., Girod, B., Sanneck, H., "A new technique for audio packet loss concealment", In Proceedings of IEEE Global Telecommunication Conference, Communications: The key to Global Prosperity, 1996, pp. 48-52.
[5] Feng, H.Y., Ling, Z.J., "Implementation of ITU-T G.729 speech codec in IP telephony gateway", Wuhan University Journal of Natural Sciences, 2000, vol.5, no. 2, pp. 159-163.
[6] Feng, H.Y., Yang, J., Zhang, S., Zhang, J.L., "Implementation of ITU-T G.723.1 dual rate speech codec based on TMS320C6201 DSP", In Proceedings of 5th International Conference on Signal Processing, 2000, vol. 2, pp.679-682.
[7] Han, S., Jeong, S., Yang, H., Kim,J., "Noise reduction for VoIP speech codecs using modified Weiner filter", Advances and Innovations in Systems, Computing Sciences and Software Engineering, 2007, pp. 393-397
[8] Yamada, M. and Nishihara, A., "High-Speed FIR Digital Filter with CSD Coefficients Implemented on FPGA", in Proceedings of IEEE Design Automation Conference, 2001, pp. 7-8.
[9] Soderstrand, M.A., Johnson, L.G., Arichanthiran, H., Hoque, M. and ELANGOVAN, R., "Reducing Hardware Requirement in FIR Filter Design", in Proceedings IEEE International Conference on Acoustics, Speech, and Signal Processing 2000, Vol. 6, pp. 3275 – 3278

[10] Martinez-P. J., Valls, T. Sansaloni, Pascual, A.P. and Boemo, E.I.., "A Comparison between Lattice, Cascade and Direct Form FIR Filter Structures by using a FPGA Bit-Serial DA Implementation", in Proceedings of IEEE International Conference on Electronics, Circuits and Systems, 1999, Vol. 1,pp. 241 – 244.
[11] Proakis, J.D., Manolakis, D.G., Digital Signal Processing: Principles, algorithm and applications, 3rd ed., 2000.
[12] Ifeachor, E.C., Jervis, B.W., Digital Signal Processing: A Practical Approach, 2nd., Pearson Education, 2003.
[13] Li, C., "Design and realization of FIR digital filters based on MATLAB", In Proceedings of International Conference on Anti-Counterfeiting Security and Identification in Communication, 2010, pp. 101-104.
[14] Mahdi, A.M., and Othman, M.B., "An algorithm proposed for FIR filter coefficients representation", International Journal of Applied Mathematics and Computer Sciences, Vol.4, No.1, 2007, pp: 24-30
[15] Bolot, J.C., "End-to-end packet delay and loss behavior in the internet", In Proceedings of ACM Symposium on Communications Architectures, Protocols and Applications, 1993, pp.289-298
[16] Sannech, H., Le, N.T.L., "Speech property-based FEC for Internet Telephony applications", In Proceedings of the SPIE/ACM/SIGMM Multimedia Computing and Networking Conference, 2000, pp.38-51.
[17] Hohlfeld, O., Rudiger.G, Halblinger, G., "Packet loss in real-time services: Markovian Models generating QoE impairments", In Proceedings of the 16th International workshop on quality of service, 2008, pp: 239-248.
[18] Jelassi, S., Youssef, H., Hoene, C., Pujolle, G., "Voicing aware parametric speech quality models over VoIP networks", In Proceedings of the Second international conference on Global Information Infrastructure Symposium, 2009, pp. 120-127.
[19] Wu, C.C., Chen, K.T., Huang, C.Y., Lei, C.L., "An Empirical evaluation of VoIP playout buffer dimensioning in Skype, Google Talk and MSN messenger", In Proceeding of 19th International Workshop on Network and Operating System Support for Digital Audio and Video, 2009, pp-97-102.
[20] Singh, H.P., Singh, S., Singh, J., "Computer modeling and performance analysis of VoIP under different strategic conditions", in Proceedings of 2nd IEEE International Conference on Computer Engineering and Applications, 2010, vol.1, pp. 611-615.
[21] Singh, H.P., Singh, S., Singh, J., "Digital signal processing approach for performance improvement in Voice over Internet Protocol (VoIP) network", International Journal of Information and Communication Technologies, 2011, vol.4, no. 1-2, pp. 85-90.
[22] http://www.utdallas.edu/~loizou/speech/noizeus/.
[23] 3GPP.TS 26.090 : Mandatory speech codec processing functions: Adaptive Multi-Rate (AMR) speech codec: Transcoding functions, 2009
[24] ITU-T Recommendation G.729 Annex A. Reduced complexity 8 kbit/s CS-ACELP speech codec, 1996.
[25] ITU-T Recommendation P.862: Perceptual evaluation of speech quality (PESQ); an objective method for end-to-end speech quality assessment of narrowband telephone networks and speech codec, 2001.