Efficient Digital Filter Banking Algorithm for Wireless VoIP

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Abstract. Due to advancement in Today's information and communication technology, Voice over IP (VoIP), emerged as one of the most reliable and cost efficient technology over the traditional ones, the same used to transmit the voice data over available computer networks using standardized protocols such as Internet Protocol (IP). Along with the advantages such as scalability and security, VoIP has some threats to deal with such as voice quality and interference. In this research article, an efficient digital filter banking scheme is proposed and the same is implemented after decoding the signal into the system as post-processor. To analysis the performance of the proposed, it is evaluated under various network conditions. Later, the proposed scheme is compared with the traditional one. The proposed scheme shows better results in terms of speech quality with the rational ones.

1 Introduction

Voice over Internet Protocol (VoIP) has emerged as one of the most significant technology in the field of communication and evolved as a substitute to the conventional communication method as the Public Switched Telephone Network (PSTN). Voice over Internet Protocol (VoIP) is the technology used to transmit the voice through internet or local area networks and based on packet switching. The packets in the same are transmitted through the most efficient path from the available ones. The conventional communication system such as public switched telephone network (PSTN) was using circuit-switching for communication which requires a dedicated line [1]. The conventional communication systems was using analog infrastructure. The analog infrastructure lacks in robustness and is not able to deal with the noise occurring from various sources but in digital communication such as VOIP, noise can be handled for example by using repeaters the original signal can be recovered [2]. Singh et al. [3] discussed the evolution of voice communication over the period of the time beginning from packet switched networks to modern VoIP. The authors describe that, for providing voice communication to the users VoIP can be the best alternative of the PSTN telecommunication services. Later, the authors summarized the advantages & disadvantages, compression & measurement schemes for the VoIP system. Han et al. [4], this research work focused on the noise removal and reduction in VoIP transmission. To achieve their research objective of noise reduction in VoIP transmission, authors

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used the modified Weiner filter. The Weiner filter lowered the signal to noise ratio at each frequency. In this research work, authors proposed a system to reduce the noise by using pre-processing of the voice signal. By using the proposed scheme, authors observed better results in terms of MOS score.

Bolot et al. [5] in this research work, authors observed very interesting facts about VoIP transmission. The authors choose the times scales of the interest scales very efficiently from a small amount of milliseconds to couple of minutes. During their experiment, authors recorded rapid fluctuations of queuing delay over small interval of time. Such as, loss of packets are mainly random until and unless they consume a large portion of the existing bandwidth. Later the authors discussed the effects of the said results on the design and implementation of control mechanisms for the internet. For speech applications Noor et al. [6] proposed noise canceller using multirate filter bank by splitting the input signal spectrum and applied the Least Mean Square (LMS) algorithm to control the finite impulse response filter. In the proposed systems, authors concluded that computational power is significantly decreased by implementing the polyphase and the noble identities. Samad et al. [7] focused on the noisy environments and proposed a new feature extracting method for the same. In the proposed method, the speech signals are decomposed into subbands to find out noisiest subband. After decomposing of the speech signal, adaptive filtering methods are applied to the same. In the research work, decomposition of the speech signal is performed with the help low complexity octave filter bank. The adaptive filtering is calculated by applying normalized least mean square algorithm. Ninkovic et al. [8] discussed that there is a need of implementing a loss recovery method to address the VOIP quality challenges where the packet loss is very high. The authors also discussed the disturbing effect of the heavy packet loss in VOIP. The authors emphasized that, the traditional loss recovery methods are dependent on various other factors.

2 VOIP Success Over PSTN

In the section, various factors are discussed which attract the user to migrate towards VOIP from PSTN:

- **Low Cost:** Low cost is one of the most attracting features of the VOIP in comparison with PSTN. VOIP system does not require new infrastructure as it can use the existing one such as computers, laptops etc. to make communication cheaper. The low cost feature also attracts the small and medium sized organization to adopt VOIP.
- **Scalable:** Number of the connecting ports can be added in the PSTN to make it expandable but for the same extra cost is also involved but VOIP is a software based solution where updates can be very easily integrated at no extra cost.
- **Disaster Recovery:** In VOIP system, it chooses the most efficient path from the available ones; if one path is disturbed due to some reason other paths can be chosen. Due to interconnection of various routers and switches, there is more than one path to research the destination as compared to PSTN where there is only dedicated path available.

- **Advance Features:** In VOIP various advance features are very easily integrated without heavy cost such as call forwarding, caller-id and broadcasting of message in comparison to PSTN.
- **Security:** High level of security can be achieved in VOIP than PSTN by using Virtual Private Network (VPN).

3 Demerits of VOIP System

VOIP system is very well implemented due to its advantages; this section is focused on the demerits of the VOIP System, which are as follows:

- **Voice Quality:** The quality of voice signals received by the receiver is not up to the mark, sometimes due to the distortion in the signal over wireless VoIP system. The factors affecting the same are network speed, connectivity and the hardware used for implementation of the wired/wireless VoIP system.
- **No Power No Service:** In the case of power failure, VOIP phone system becomes unreachable as it is needs continuous power supply for working. Although the same is not applicable for the regular telephone system as the power supplied through the telephone line. To make maximum use of VOIP telephone system it requires continuous power supply.
- **Security:** Wireless VOIP uses the computer network for communication and the same is also used by many other users. Wireless VOIP are going to face the same security issues as in computer networks. The major security issues over VoIP are network identity theft of service; phishing attacks, virus's attacks, service denial, tampering of calls.
- **Interference:** Interference can be caused during the transmission of signal into a channel from source to destination. Interference is one of the major limiting factors in the performance of wireless system and the same is also responsible for dropped calls. It also results in crosstalk on voice channels while communicating.

4 Methodology

To enhance the signal quality of the narrowband and wideband wireless VoIP system, which could be degraded due to the loss of voice packets during the communication, the computationally efficient scheme has been proposed. The step by steps plan is as under (Fig. 1):

Step 1: The original speech signal has been fed into the system

Step 2: The speech signal has been encoded with various narrowband wireless VoIP speech encoders at different data rates

Step 3: The compressed version of the speech signal has been packetized into the VoIP packets to transmit over the IP network. To check for network efficiency, the packet size can be varied

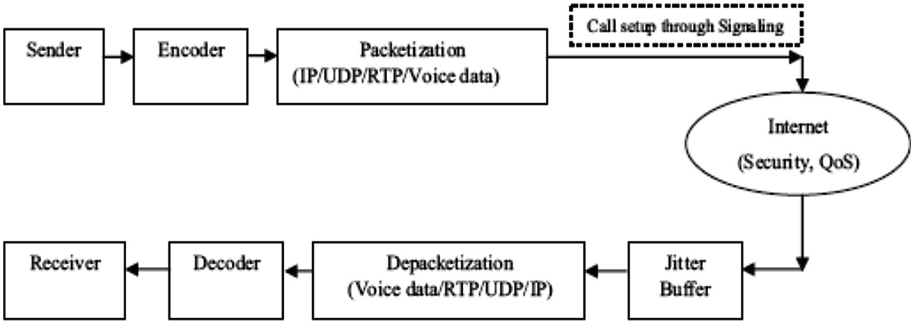

Fig. 1. General architecture of VOIP system

Step 4: The IP network impairments into the speech signals has been introduced through 2-state Gilbert-Elliot model. The signal has been degraded with different packet loss rates
Step 5: The degraded VoIP signal has been depacketized and then decoded with the corresponding decoders
Step 6: The digital filtering schemes has been incorporated into the system as post processor, after the decoder
Step 7: The signal measurement and evaluation had been performed with PESQ

The methodology of the proposed work is presented in Fig. 2.

5 VoIP Simulations

In this section, the VoIP simulation results are discussed. The proposed scheme is implemented on the various modulation schemes such as DBPSK, DQPSK and QAM64. For the experiment simulator is designed according to the available network conditions. The speech signal encoded into the VoIP frames using G.711, G.729 and AMR-NB encoder. The VoIP frames were designed by Gilbert-Elliot mode and then resulting stream decoded into degraded voice. The measurements were carried out through PESQ. The set of speech recordings were taken from [11] for VoIP simulations. The speech signal is degraded with various packet loss rates as described in Table 1. The performance of the proposed filter is analyzed for various loss rates for G.711, G.729A& AMR-NB based wireless VoIP system (Fig. 4).

6 Results and Discussion

Simulations of synthesis and dyadic filtering based VoIP system had been performed for narrowband G.711, G.729A & AMR-NB (7.95 kbps) coders. The variation of the packet loss rates and PESQ-MOS scores for G.711, G.729A and AMR-NB coders in VoIP system are shown in Figs. 3, 5 and Tables 2, 4. The significant increase in

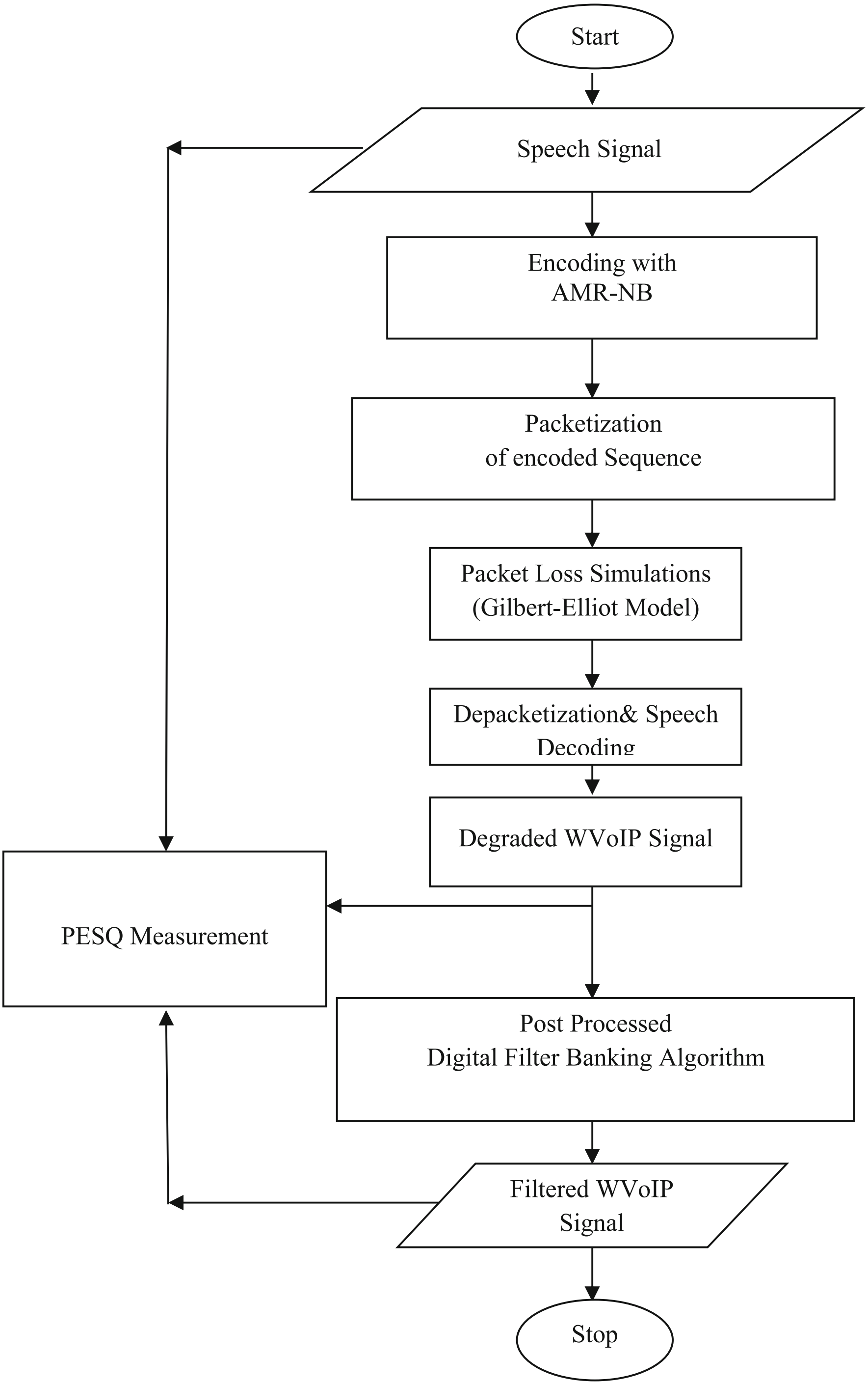

Fig. 2. Flow chart for simulation study of proposed system

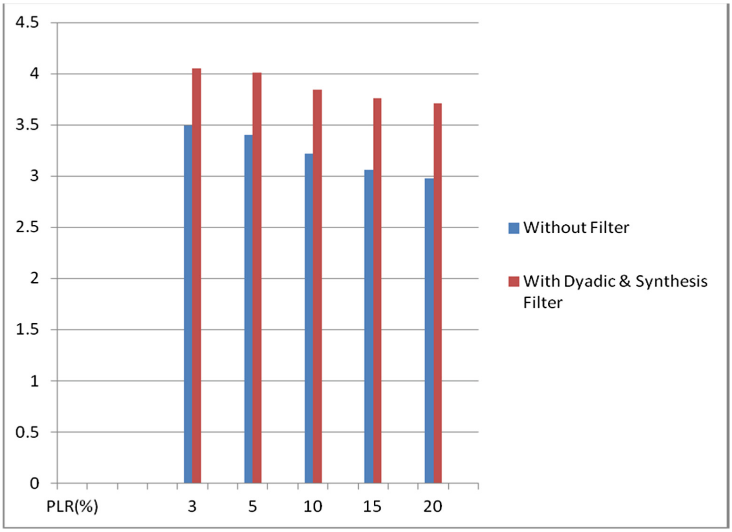

Fig. 3. Dyadic & synthesis filter bank for G.711

Table 1. Simulated loss rates

Packet loss rate (%)	p	q	e
3	0.98	0.032	0.005
5	0.95	0.053	0.005
10	0.91	0.101	0.005
15	0.84	0.151	0.005
20	0.81	0.201	0.005

PESQ-MOS scores is achieved at various packet loss rates with the Dyadic filters. The evaluation results presented in indicated that the DBPSK modulation scheme work well in wireless VoIP system. The proposed scheme is much effective at high packet loss rates as presented in results. The proposed scheme effectively conceals the lost packet during VoIP speech transmission to improve the signal quality (Table 3).

7 Discussion

It is observed form the comparison results that the signal quality of the wireless VoIP signal is significantly increased with Synthesis & Dyadic filtering scheme as compared to the without filters as presented in Figs. 3, 5 and Tables 2, 4. The much improvement in PESQ-MOS scores is observed with Dyadic filter for wireless VoIP speech signal. The results of the G.729A can be compared with recovery by reinitialization (RbR) [12] method where the increase in the signal quality was 0.0065–0.17 (with packet loss rates 10% to 50%), which is very low as compared to the proposed work.

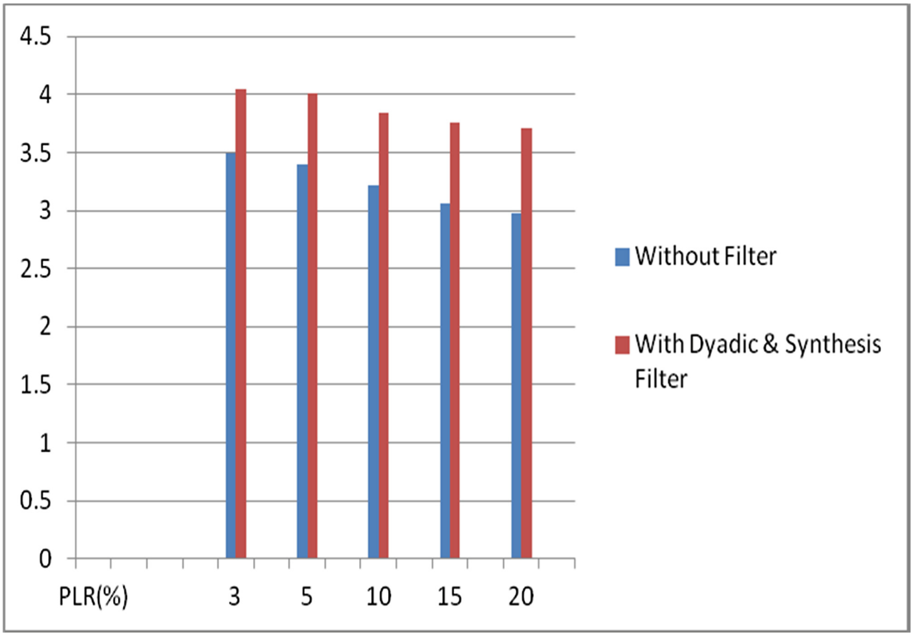

Fig. 4. Dyadic & synthesis filter bank for G.7.29A

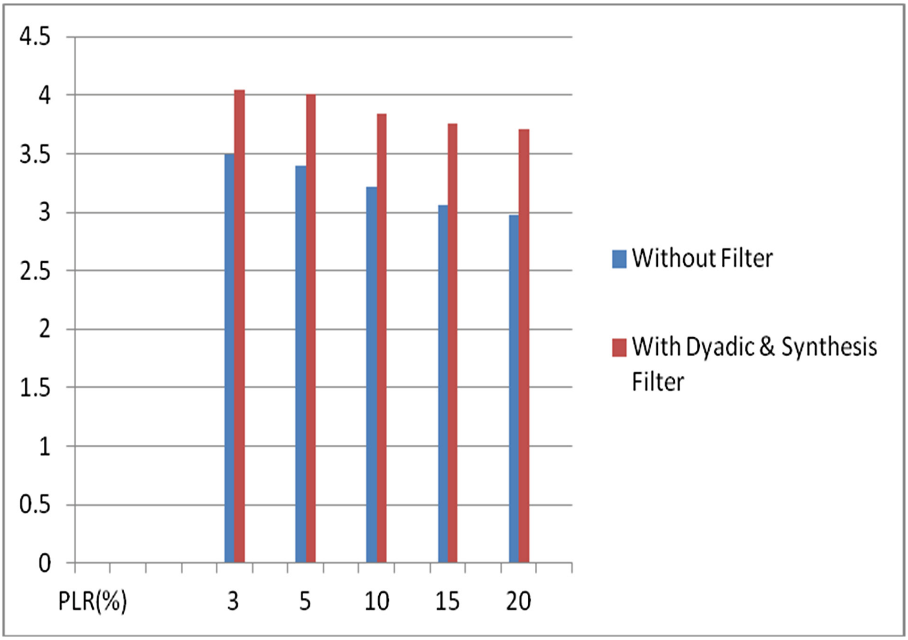

Fig. 5. Dyadic & synthesis filter bank for AMR-NB 7.95 kbps

Table 2. MOS scores for G.711

PLR (%)	MOS scores	
	DBPSK	
	Without filter	With dyadic & synthesis filter
3	2.28	3.01
5	2.20	2.89
10	2.02	2.68
15	2.01	2.48
20	1.89	2.42

Table 3. MOS Scores for G.729A8kbps

PLR (%)	MOS scores	
	DBPSK	
	Without filter	With dyadic & synthesis filter
3	2.39	3.27
5	2.28	3.14
10	2.02	2.92
15	1.96	2.79
20	1.87	2.52

Table 4. MOS scores for AMR-NB 7.95 kbps

PLR (%)	MOS scores	
	DBPSK	
	Without filter	With QMF
3	2.48	3.35
5	2.39	3.23
10	2.25	3.01
15	2.09	2.89
20	2.02	2.68

8 Conclusion

The synthesis and dyadic filtering based VoIP system had been performed for narrowband G.711, G.729A & AMR-NB (7.95 kbps) coders in this work. It had been observed that the proposed scheme works well for both narrowband wireless VoIP systems, since the proposed scheme enhances the signal quality. The enhancement in signal quality was measured through PESQ measurement schemes and it had been found that results of the filtering based wireless VoIP system were better than that of without filtering based system with DBPSK modulation scheme. In future the work can be extended for noisy environment and implemented on TMS320C6713 DSP processor.

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