Analysis of VoIP Signal Processing for Performance Enhancement

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Abstract

The present paper outlines the voice over internet protocol service, various features and services provided by VOIP technology, The quality of voice over IP is evaluated based on packet-loss rate, delay, jitter and echo compensation. In this paper we discuss viability of real time processing of adaptive voice over IP using adaptive rate control algorithms. Finally, performance analysis of LMS, Sign-Error LMS algorithms for AVOIP is presented in this paper.

Keywords: AVOIP, LMS, NLMS, Sign-Error LMS

1. Introduction

Voice communication over internet not be possible without a reliable data network, this was first available when distributed network topologies were used in conjunction with data packets. Early network used single centre node network in which a single workstation (Server) is responsible for the communication. This posed problems as if there was a fault with the centre node, (workstation) nothing would work. This problem was solved by the distributed system in which reliability increases by spreading the load between many nodes. The idea of packet switching & distributed network were combined, this combination were increased reliability, speed & responsible for voice communication over internet. Voice-over-IP (VoIP) These data packets travel through a packet-switched network such as the Internet and arrive at their destination where they are decompressed using a compatible Codec (audio coder/decoder) and converted back to analogue audio. [1]

2. Adaptive VoIP

The aim of voice transmission over IP is to find the technical solution of the problems which affects the QoS of VoIP network like delay, jitter, delay variation, packet loss, speech compression and offer a QoS under most network conditions. So our overall objective is to reach the high quality level of service provided by VoIP. To achieve this quality level a new approach is presented known as 'adaptive VoIP' , whose basic idea is the adaptation of source coder to the present state of the network if network is congested the speech coded at lower bit rates and when network is lightly loaded then speech is coded at higher bit rates or simply we can say the adaptive nature of the VoIP network. So adaptive VoIP relies on variable bit rate speech coders to generate required bandwidth. Advantage of this adaptive approach is the efficient use of resources of the present VoIP network. To achieve adaptively, network congestion needs to be estimated, a rate control algorithm developed and a variable bit rate speech coder integrated into the system. [2]

Network State Estimation. the source rate depend on the state of the network, requires some way of estimating such state, since the IP service model does not offer congestion notification and the detection of temporary congestions. Based on such measures, the rate control algorithm will select bit rates compatible with the estimated capacity of the network.

Rate control algorithm. Give estimates of the state of the network, appropriate ways to adapt the source rate are then needed. In the case of IP telephony, typical networking objectives, such as maximum throughput and stability, must be considered together with perceptual constraints linked to the desired perceptual quality of service. An increase in bit rate, for instance, generally results in increased speech quality, but only if the network can sustain the increased traffic. If it cannot, the quality increase will be attenuated, or even transformed into a decrease, by greater average delays and more frequent packet losses.

Variable Bit Rate Speech Coding. Although most speech coding standards are fixed rate, several variable bit rate speech coders are available. A recent example of a variable bit rate solution is the new GSM Adaptive Multi-Rate speech coding standard [3]. In GSM-AMR one of 8 rates ranging from 4.75 kb/s to 12.2 kb/s is selected, depending on the instantaneous condition of the wireless channel. Another example is the new ISO MPEG-4 audio standard, which includes a variable bit rate CELP speech coder operating at bit rates between 3.85 kb/s and 12.2 kb/s [6].

3. Adaptive Algorithmic Analysis of VoIP

Adaptive algorithms become the effective part of DSP and its plays a vital role in communication. Least mean squares (LMS) algorithms is a type of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). An adaptive algorithm is used to estimate a time varying signal. There are many adaptive algorithms like recursive least square (RLS), Kalman filter, least mean square (LMS) etc. the LMS algorithm is most commonly used for various applications.[5]

The LMS algorithm was developed by Windrow and Hoff in 1959. The algorithm uses a gradient descent to estimate a time varying signal. The gradient descent method finds a minimum, if it exists, by taking steps in the direction negative of the gradient. It does so by adjusting the filter coefficients so as to minimize the error. The LMS algorithm approaches the minimum of a function to minimize error by taking the negative gradient of the function. [6]

The Normalized least mean squares filter (NLMS) is a variant of the LMS algorithm that solves this problem by normalizing with the power of the input.Some adaptive filter applications require to implementation of adaptive filter algorithms. This purpose requires a simplified version of the LMS algorithm. Applying sign function to the standard LMS algorithm results three types of LMS algorithm as Sign-error LMS algorithm, Sign-data LMS algorithm, Sign-Sign LMS algorithm In the present paper we analyze only standard LMS and Sign-Error LMS algorithm.[7-8]

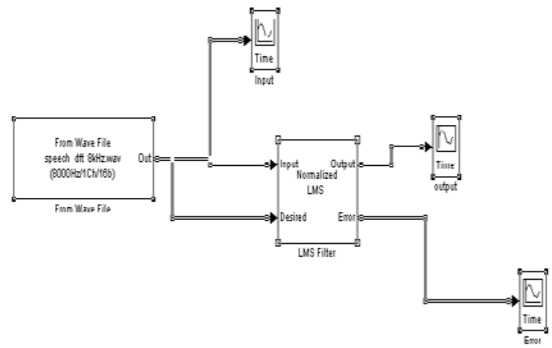

Fig. 1 Model for analysis of adaptive filtering algorithms

Results and Discussion

The spectrum given below shows the analysis of various adaptive filtering algorithms based on VoIP simulink proposed here.

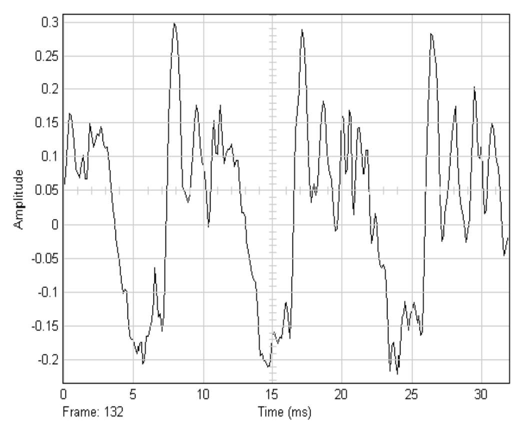

Fig. 2 LMS Input

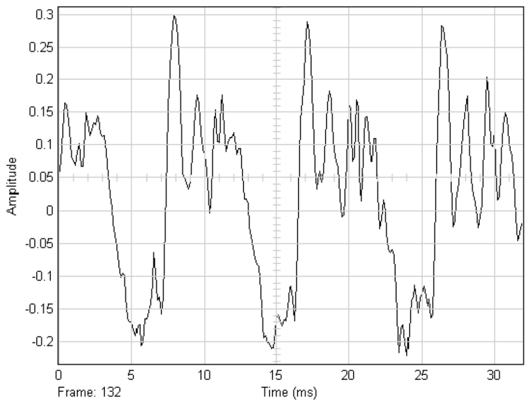

Fig. 3 LMS Output

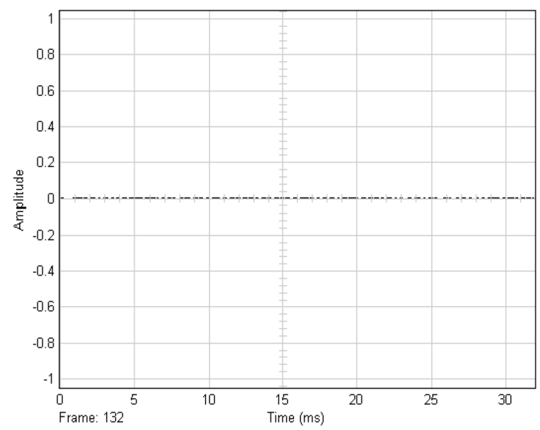

Fig.4 LMS Error

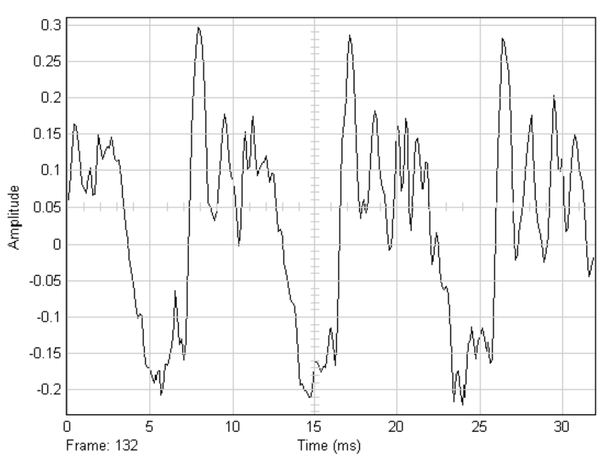

Fig.5 Sign- Error LMS Input

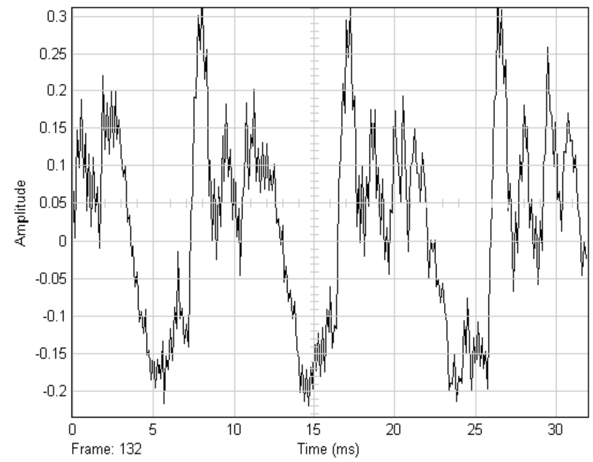

Fig.6 Sign-Error LMS Output

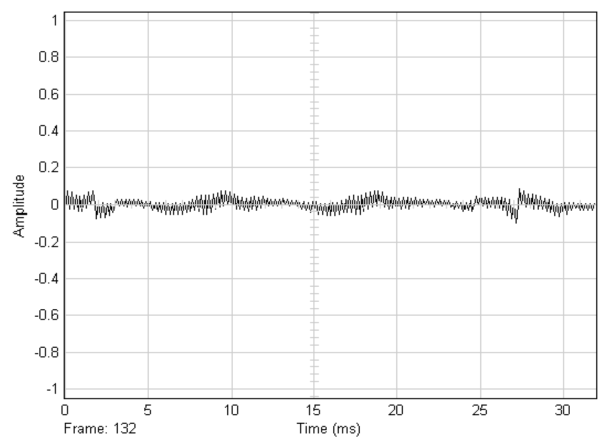

Fig.7 Sign-Error LMS Error

From the above spectrums for LMS, Sign-Error LMS algorithms we analyze that for LMS algorithm there is zero error in the analysis, this is because here we are using same signal for both the input as well as desired signal and using distorted signal.

4. Conclusion

The present paper has highlighted the role of adaptive filtering in VoIP system the different adaptive algorithms such as LMS, NLMS, Sign-Error LMS algorithms have been attempted in the VoIP based simulink model. The work is in progress to incorporate influencing parameters for the performance improvement of VoIP system.

5. Future work

In this paper a simulink model for VoIP is proposed and on the basis of this model our future objective is to analyze the distortion and degradation of the signal. Our ultimate aim is to quality of VoIP signal and reduces degradation, for this work is in progress.

6. References

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