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Packet Loss Recovery and Control for VoIP

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Abstract: ***Voice over Internet Protocol (VoIP) is a technology to transfer voice signal over the internet (Network). The transmission of audio data over digital packet switched networks faces lots of issues and challenges basically due to the presence of network impairments such as packet loss, delay, echo, network security and throughput. The biggest issue in the digital voice and video communication is the loss of packets due to network congestion, delay, jitter and other network factors. Different protocols have been standardized to detect and recover the lost packets to overcome the issues and maintain the quality of communication. However, for a real time application it is too late before a lost intermediate packet is retransmitted. This paper evaluates the implementation of agood Packet Loss Concealment (PLC) techniques that must be in action to minimize the effect of packet loss during voice communication. PLC algorithm is an end-to-end bidirectional transmission control scheme that minimizes the effect of data loss to improve sound quality in voice communication.***

Keywords: VoIP, PLC, Recovery, SIP

1. Introduction

Today communication services are offered through the network. It allows people to exchange and retrieve the information. The most popular and widely used service is Voice over Internet Protocol (VoIP). The reason for the popularity of VoIP is that it allows people to communicate with each other freely at low rates [1]. VoIP uses the Internet Protocol (IP) to route packets containing small portions of voice conversations between the callers. In digital communication, the voice quality in VoIP is mostly dominated by the characteristics of packet networks such as delay, jitter, and packet loss. Therefore, it is important for us to take characteristics of IP network into consideration when we design a VoIP application. The transmission technology of VOIP is digital. Hence the caller's voice is digitized. The digitized voice is compressed [2]. Speech quality of VoIP may potentially be degraded by transmission errors such as packet loss, delay, jitter and echo [3]. Major challenges of VoIP network is maintaining quality i.e. packet loss which is a serious and critical issue for voice over Internet Protocol applications. VoIP needs a packet loss concealment (PLC) technique under a congested network as a solution to overcome the effects of packet loss in voice communication [4].

2. Literature Review

Chetty et al. [5] defined VoIP as referring "to a range of protocols designed to send voice over packet switched networks, traditionally the domain of internet traffic." In VoIP voice is sampled at a certain frequency which can be set to any desired value on the devices in use [6]. The sampled voice is then digitized and then finally packaged into packets before being sent over the IP network. VoIP uses different protocols for the call setup and the actual conversation between two communicating device. Signaling protocols like SIP, H.323 are used for the call setup and then the RealTime Transmission Protocol (RTP) is used to carry the voice between the telephones. RTP has been specifically optimized for the transmission of real time data.

There are advantages of VoIP over the Traditional Public Switched Telephone Network. These advantages include the fact that VoIP makes more efficient use of bandwidth by only transmitting when something useful is being sent. This is done by silence compression which cuts out the bits when there is silence in a conversation thereby minimizing the packets sent across a network. Also VoIP unlike the traditional PSTN does not require a dedicated link between two end points since it uses packet switching instead of circuit switching for communication, and this allows the network to be used for multiple conversations concurrently. [6]

W.Yu et al [7] gave a detailed suggestion of how to implement a Point to Point (P2P) network layout for a VoIP network but do not give any comparisons between the P2P network layout and the client/server layout. His experiment demonstrates that the P2P hierarchy model for conferencing applications can achieve better performance than all other VoIP models in terms of minimizing the network bandwidth overhead.

The use of VoIP services is quickly growing up. Although, there are many active corporations that offer and implement VoIP services. VoIP services demand is getting increased everyday as its cost is far less for the customer than the traditional telephone system in particular for international calls. One of the main advantages of VoIP services is: it provides superior management for provider involved and also cost is very less as organization could have single set-up for both data and voice as well. So it is beneficial for both customers and as well as service providers. [8]

Packet Loss Concealment technique is beneficial for all the services that use IP communication. The concern for voice data is that it must be in real time, so it is of maximum priority that communication is robust and lossless. There always exists loss, delayed or dropped voice packet during real time communication. SoPLC technique is required to monitor and interpolate about which voice pattern that are received from the network? In this scenario, there is no any extra bandwidth consumption or additional request to sender.

Receiver can compute and interpolate voice packet by itself and maintain the Quality of service.

2.1 Quality of Service in VoIP

In VoIP, quality means the ability to talk and listen clearly without any unwanted noise. The three major factors that affect the speech quality in VoIP are: Packet Loss, Jitter and Latency.

Packet Loss

Packet loss happens when two users talk on a VoIP phone and the call begins to "break up" during packet transfer over the IP network. This mainly happens when there is a lot of congestion on the network and it results in some part of the conversation being missed at the receiving end. Thus to minimize the effect of packet loss, all VoIP applications should have a module for Packet Loss Concealment

Jitter

In VoIP, when a call is established, the sender sends the VoIP packet at a constant frequency, for example 10ms or 20ms. As the packet travels over the internet it might get delayed or lost, resulting in receiver not receiving the packets with the same frequency. The difference in the expected time of arrival and the actual time of arrival of the packet is called jitter. It is usually experienced in heavily congested networks

Delay or Latency

The total amount of time taken for speech to travel from speaker's mouth to listener's ear is measured as delay or latency. In term of VoIP packets, it is the total time taken by the packet to travel from the source to the destination.

2.2 Packet Lost Concealment Techniques and Algorithms for VoIP

Audio voice is continuous stream of data. Filtering of long sequence voice data is not possible due to limitation of memory size in the digital computer/processor. The input sequence is divided into number of blocks and output blocks are computed for respective input blocks. The input blocks are filtered together to produce the complete output sequence.

Average Magnitude Difference Function (AMDF)

AMDF is the absolute value of the difference between the original signal and the delay signal to calculate the fundamental frequency i.e. Pitch of that signal. AMDF decrease the computation complexity that makes it more suitable for the real-time applications to calculate their pitch

AMDF pitch detection algorithm[9]:

D_{AMDF} (m) =1/N $(\sum_{N=0}^{N-1} |x(n) - x(m+n)|)$

Where x (n) are the samples of speech. For a periodic signal with period T_0, this function is expected to have a strong minimum when the index m equals T_0. The pitch period is, in general, estimated as follows:

T_0 = MIN (D_{AMDF} (m)),   for m = m_{min} to m_{max}

Where the values of m_{min} and m_{max} are chosen to cover the expected pitch-range.

OLA (Overlap and Add)

Input sequence is divided into number of block size.

N=L+M-1   (M-length of impulse response)

Each block consists of first L points from input sequence tailed by M-1 zero.

Using these segmented blocks of input the respective output blocks are computed using Fourier transform. The linear convolution of sequences is always longer than the original sequences. Then output blocks are fitted together to form the complete output. In this paper, OLA with linear convolution is used.

2.1 Recovery Approach used for Quality Control in VoIP

Measurement of Speech Quality

In VoIP, quality simply means being able to listen and speak in a clear and continuous without unwanted noise. QoS (Quality of Service) is essentially a service that prioritizes certain data traffic by slowing down the less important data packets. These important packets then reach their destination as quickly as possible. Speech quality is the measurement of user experience when a VoIP call is established. The measurement of speech quality is divided into two categories, Objective and Subjective. Subjective method are based on user listening tests. Users are told to rate the speech quality. These tests are tedious to perform and the accuracy of speech quality rating relies on the user's mood. These methods are limited, impracticable and too expensive. In order to avoid these problems, new methods that permit the calculation of values representing the different damaging factors combinations of the network have been developed. The quality estimation, providing results as near as possible to Mean Opinion Score (MOS) values. The objective of measurement techniques developed uses an approach where a voice sample represents the input signal to produce a score, representing the original signal produced by the network.[10]

Mean Opinion Score

Mean Opinion Score (MOS) is a numerical value to indicate the quality of the call from the user's perspective of the received call after compression, transmission, and decompression. MOS is a calculation based on the performance of the IP network. The Mean Opinion Score Values Taken in whole numbers, are quite easy to grade

Table 1: Mean Opinion Score (MOS)

Value	*Level*	*Description*
5	Excellent	Perfect. Like face-to-face conversation or radio reception.
4	Good	Fair. Imperfections can be perceived, but sound is still clear.
3	Fair	Annoying.
2	Poor	Very annoying. Nearly impossible to communicate.
1	Bad	Impossible to communicate.

The values do not need to be whole numbers. Certain thresholds and limits are often expressed in decimal values from MOS. For instance, a value of 4.0 to 4.5 is referred to as toll-quality and causes complete satisfaction. Toll Quality is a voice call quality comparable to that of an ordinary long distance call, originally placed over the analog (circuit-switched) Public Switched Telephone Network (PSTN).

In order to analyze calls quality, voice communication systems provide a way for extracting call related data and network information. This type of information is usually provided from VOIP Server in well-defined structures named call detailed records (CDR) and call management records (CMR). CDR files usually contain information regarding the source and destination, the duration but also some network parameters like packets sent, latency, jitter, etc.

3. Research Methodology

A VoIP server with SIP clients were set up as a test scenario to perform voice communication in a network. Also VoIP monitoring software is integrated in same network to view real time network and call status. Monitoring software records audio for real time voice communication, call detailed records (CDR) and call management records (CMR). This files are later used for analysis purpose.

For this test, figure 1 depicts a working lab for VoIP server with SIP clients in a network domain. Call status, jitter buffer status and value of packet loss, are extracted from system CLI (Command Line Interface) during call and from VoIP log file as per requirement

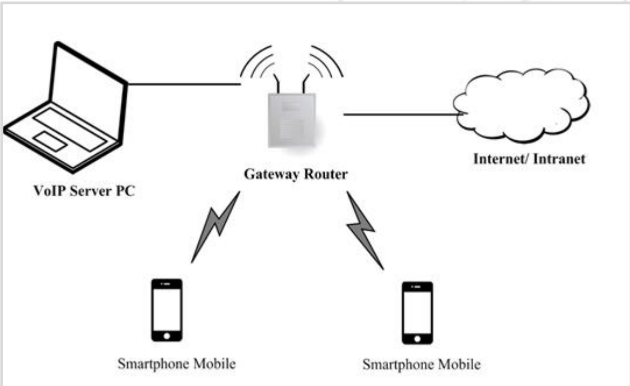

Figure 1: Use Case Scenario

Asterisk [11,12] is an open source framework for building communications applications. Asterisk turns an ordinary computer into a communications server. Asterisk is a software implementation of a telephone Private Branch Exchange (PBX).It allows telephone to make calls to one another and to connect to other telephone services, such as the Public Switched Telephone Network (PSTN) and Voice over Internet Protocol (VoIP) services.Most of the open source VoIP server use Asterisk platform as their background. Different VoIP servers and its application are developed on Asterisk platform to meet their requirements. VoIPmonitor [13] is an open source network packet sniffer with commercial frontend for SIP RTP and RTCP VoIP protocols running on Linux. VoIPmonitor analyze quality of VoIP call based onnetwork parameters - delay variation and packet loss according to ITU-T G.107 E-model which predicts quality on MOS scale. Calls with all relevant statistics are saved to database. Optionally each call can be saved to Pcap file with either only SIP protocol or SIP/RTP/RTCP protocols.

Usually the Quality of Service (QoS) rating of VoIP is carried out using the subjective quality measure called Mean Opinion Score (MOS). MOS scales from 1 (lowest quality) to 5 (highest quality). This type of measurement focuses on the perceived quality provided by users. However, quality data for the study (packet loss rate, delay and jitter values) have been collected using VoIPmonitor, a network packet sniffer software with commercial frontend for SIP RTP and RTCP VoIP protocols. Packet Loss values are obtained from RTP and RTCP responses from SIP client. The packet loss values are represented in the percentage form of the total RTP packets being transmitted. The delay values are obtained by calculating the difference between the RTP packet actual arrival time and the estimated arrival time. On the other hand, the jitter values have been derived from the differences in the inter-arrival time of the RTP packets. MOS is useful when considering the overall end-to-end quality of VoIP communications. For the ideal wireless LAN Environment, the default network setting with no packet loss is used.

4. Result and Analysis

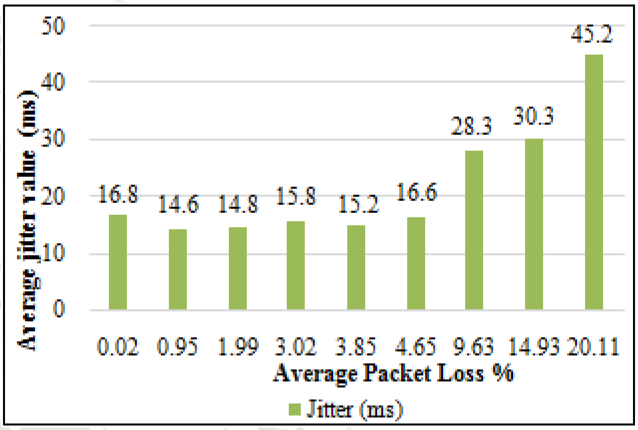

Figure 2: Average value of packet loss and jitter when Jitter Buffer Disable

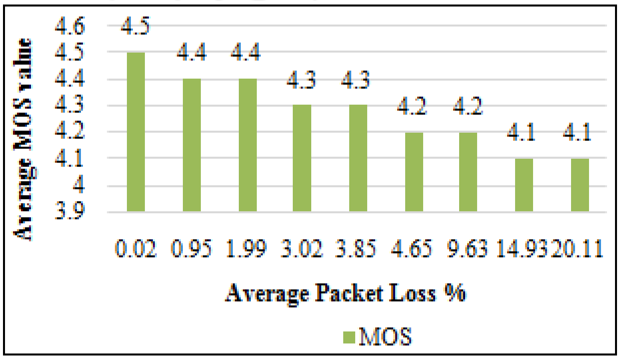

Figure 3: Average value of packet loss and MOS when Jitter Buffer Disable

In Figure 2 and 3, when packet loss is greater than 15%, average jitter value is above 30ms. During RTCP observation it is found that jitter crossed 40ms for some instance. Normal VoIP communication cannot be established for Packet loss greater than 15%.

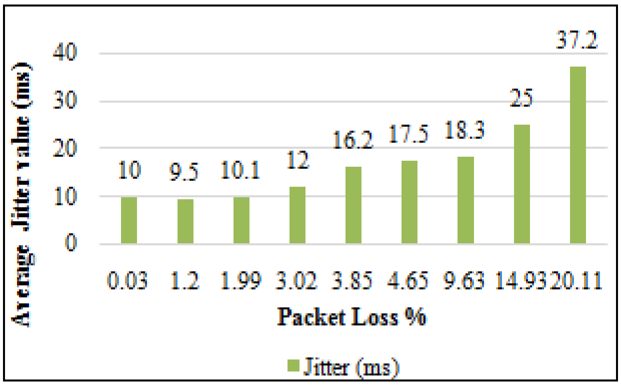

Figure 4: Average value of packet loss and Jitter when Jitter Buffer Enable

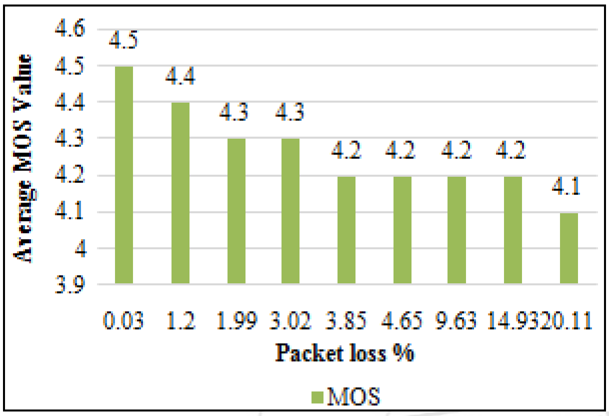

Figure 5: Average value of packet loss and MOS when Jitter Buffer Enable

In figure 4 and 5, when packet loss is greater than 15%, average jitter value is almost 25. Comparing Figure 2 and Figure 3, Average MOS value when Packet Loss Concealment enabled is more than MOS value when Packet Loss Concealment is disabled. Also for Jitter value when Packet Loss Concealment is enabled, the significant difference has been seen.

It is found that average Mean Opinion Score (MOS) for different loss scenarios have slightly been changed from 4.4 to 4.1.Although average value of MOS, is satisfactory when jitter value increases, will affect quality of voice call communication. During Loss period MOS value is 3 or below, in which VoIP call quality is annoying or bad. Jitter value greater than 40ms is not acceptable in VoIP communication.During loss period jitter increases to 40ms or which will affect the VoIP call at that instance.

5. Conclusion

From the experiments and analysis it is observed that the quality of the voice signal is affected from various network parameters as packet loss, delay/latency and jitter. Several objective tests were performed in order to test the proposed PLC technique. Initially different packet loss scenario varying from 1% to 20% were tested without implementing PLC technique. After analysis on results from different scenarios, it is concluded that the loss effect on voice communication increases with the value of packet loss presents in the network. Also delay, jitter value increases with packet loss and voice communication above 10% packet loss condition cannot be established. To mitigate the effect of packet loss in VoIP, PLC technique is implemented for same case scenario as before.

From the results, it was observed that when the value of packet loss is low i.e. 0-1%, network quality is degraded for a short instance of time during voice call communication. After implementing PLC technique it was observed certain percent (1-2%) of packet loss in network that does not affect the overall quality in voice communication. In network with packet loss of 5%, jitter and MOS value is maintained under threshold which allows for normal voice communication with minor lag. For extreme packet loss condition where loss is in between 10 % to 15%, user is able to communicate with delay of 300ms to 600ms. Thus implementing PLC technique improves the Quality of Service of voice even in a network with packet loss.

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