REAL TIME PERFORMANCE EVALUATION OF VOICE OVER IP CALL QUALITY UNDER VARYING NETWORK CONDITIONS

Aladdin Sleem
Olugbenga Olumuyiwa
Khaled Kamel
Department of Computer Science
Texas Southern University
Houston, Texas, USA.

Abstract

Residential as well as business customers have been switching to Voice over Internet Protocol (VoIP) phone services. These new phone services are based on the transmission of voice over packet switched IP networks. VoIP customers use their Internet connection not only to connect to the Internet but also to make phone calls. VoIP service providers have always faced the challenge of providing customers with good call quality even though the IP network that carries call traffic does not provide any Quality of Service (QoS) guarantees. Delays, packet loss, jitter and out of sequence packets are some sources of poor VoIP phone call quality. There are several network design parameters, such as link bandwidth and the router buffer size that can be tuned to improve call quality.There are several VoIP solutions, some of them are based on Peer-to-Peer (P2P) protocols and others use the Session Initiation Protocol (SIP). In this research, we analyzed the performance of VoIP solutions under different network conditions. Experiments were conducted using real networks with different design parameters such as link bandwidth, router buffers size, and the quality of calls was measured and compared for different VoIP solutions. The research concluded by providing a comprehensive analysis of the results of the experiments highlighting the set of network parameters that gives the best call quality for each of the VoIP solutions under investigation.

1 INTRODUCTION

1.1 Background

Voice over Internet Protocol (VoIP) is based on converting voice signals into data packets then transmitting them over an Internet Protocol (IP) network as the name implies. The first form of VoIP was setup when a Vocal Tec released what was then called an internet soft phone that could be used to conduct talk between two computers on the same local network in 1995 [1]. VoIP can also be described as a unique solution which enables the transmission of voice signals over broadband internet connection rather than the traditional telephone lines [2]. In today's VoIP implementations, the voice analog signals are sampled and encoded using "coding-decoding" software (usually called codec) then encapsulated into an IP packet and carried over data cables or the internet infrastructure the same way that data packets are transmitted.

One key advantage of VoIP include making calls to long distance destinations at very cheap prices including calls to other countries with the flexibility of using the same number in different parts of the world. A key disadvantage include drops in line should there be power outages; unlike the traditional phone system that has a separate and parallel self powered infrastructure. Some VoIP solutions operate in hybrid modes which involve the combinational usage of conventional telephone device infrastructure alongside the VoIP infrastructure while others operate in the pure VoIP mode using VoIP soft phones or digital handsets connected directly to an IP network [2].

1.2 VoIP Call Scenarios

When VoIP first started, the calls used to be between two VoIP subscribers each of them using a Personal Computer (PC) connected to the internet and with a software application (called soft-phone) that interfaces with a microphone and a speaker connected to the PC. To provide VoIP subscribers with experience that is closer to Public Switching Telephone Network (PSTN) calls, IP phones were introduced later to allow subscribers to make VoIP calls using physical phones connected to the IP network (instead of the PSTN network) without the need to have a computer.

This IP phone has all the software needed to establish VoIP calls, terminate them when finished, convert the analog voice signal to digital and encode it using a codec that matches the one used at the other end of the call. A more convenient scenario is also available where VoIP subscribers can use a regular phone (not an IP phone) connected to a phone adapter that is connected to the VoIP service provider using the Internet. This phone adapter has all the VoIP settings and is capable of establishing call, terminating them, encoding the voice signals, and generating the IP packets that encapsulate the voice traffic. All the scenarios mentioned above are for calls between VoIP subscribers only and the traffic generated from these calls is transmitted across the IP network. To allow calls between VoIP subscribers and PSTN subscribers, a media gateway is used to link the IP network to the PSTN network. This allows VoIP subscribers to place a call to and receive calls from any PSTN subscriber. In summary, a VoIP call can be one of the scenarios illustrated in Figure 1.

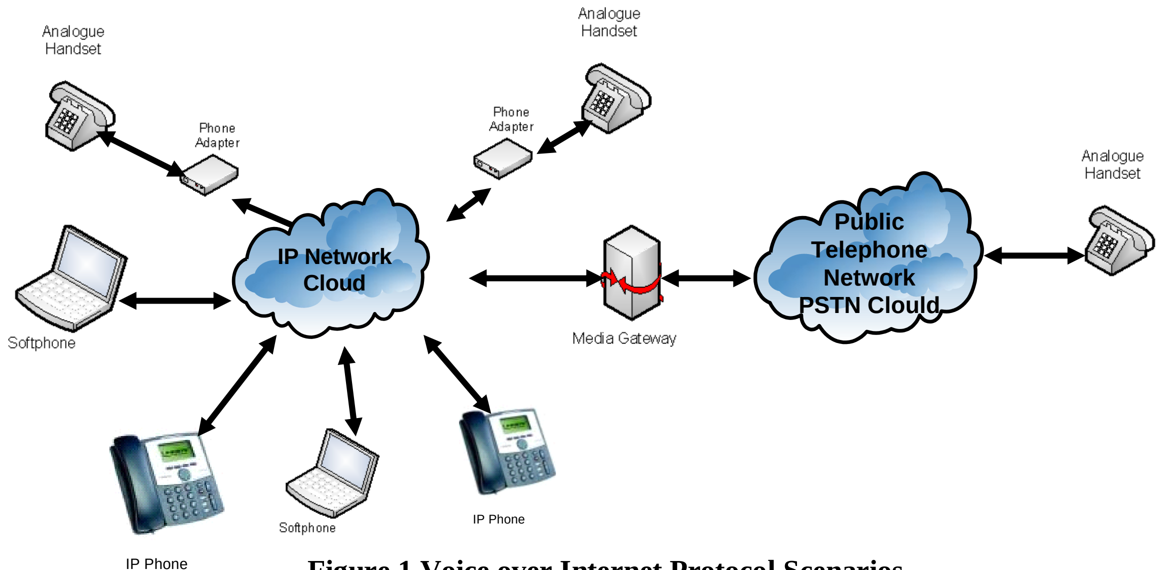


Figure 1 Voice over Internet Protocol Scenarios

1.3 Related Literature

The study and analysis of different methodologies, limitations and performance of VoIP solutions has been the subject of extensive research during the past fifteen years. The work of Yuan Thang [3] provides an in-depth analysis on Session Initiation Protocol (SIP) by describing its protocol stack, summarizing the main protocol features, and illustrating its architecture, message and operation.

M. Karam and F. Tobagi [4] studied and analyzed the delay of voice traffic in situations where voice traffic is separated from other traffic by providing a separate queue. The study concluded that Priority Queuing is the most appropriate scheduling scheme for the handling of voice traffic, while preemption of non-voice packets is recommended for sub-10 Mbit/s links. Several research have also been carried out to study VoIP signaling where different types of signaling protocol were considered alongside their capacity during videoconferencing and ability to perform functions such as the call setup, call tear down and call establishment [5].

An analysis of Peer-to-Peer (P2P) VoIP calls was also presented by Xinyuan Wang et Al [6]. Most internet peer to peer VoIP calls are usually encrypted with different watermarks and anonymous identity and research has already been done to investigate the anonymity of the peer to peer data packets and how to pass them anonymously over the internet and with high level of encryption. Several other efforts on peer to peer have been carried out focusing on how peer to peer solution can impact the internet telephony system by comparing how particular P2P platforms perform call setup, call tear down, route calls and how the Network Address Translation (NAT) or firewalls are bypassed in environments with such facilities [7].

Another study on applying P2P solutions to communication systems is presented in [8]. It proposes a P2P communications system that addresses several of the problems encountered when trying to use a P2P communication system that is based on Distributed Hash Table (DHT) in a real-world setting with combinations of residential, commercial, and academic systems. Resource reservation setup protocol (RSVP) [9] is another area where quality of service research has been addressed.

It enables a particular host to request special quality of service from the network by reserving services along the path flow on the network infrastructure in order to guarantee specific quality of performance. The work of Goode [10] discusses the factors involved in making high quality VoIP calls. It illustrates the tradeoff that must be made between the efficient utilization of bandwidth and delays. It also provides a discussion of the impact of codec selection as well as signaling protocol on the quality of VoIP calls. A real time streaming protocol dedicated for sending real time traffic such as voice and video is presented by Juha Huuhtanen [11] to enhance and guarantee excellent performance of real time traffic over the internetworks. This protocol was designed to control multiple data delivery sessions where the delivery mechanisms are of grave importance. This protocol works in conjunction with the Real Time Protocol (RTP) [12] that directs the end-to-end delivery of the real time data; some of which could affect voice packets traveling over the data network.

Another approach to initiate and manage VoIP calls is based on Peer-to-Peer networks [14]. Peer-to-Peer (P2P) VoIP solutions provide an infrastructure where communication between two parties do not pass through a third party after call setup. In other words, once the call is established, call traffic flows between the peers directly without going through a centralized server as in the SIP case. The following two sections provide an overview of these two approaches.

1.4 SIP-Based VoIP Solutions

Session Initiation Protocol is a signaling protocol used for controlling multimedia communication sessions in the area of voice, data and video. Currently, SIP is used heavily in VoIP services. It has client-server network architecture. Though efforts are going on to make it also work in P2P architectures, but the existing setup and configuration functions with a client-server concept [13].

Figure 2 below gives an example of the SIP major call setup, maintenance and routing architecture.

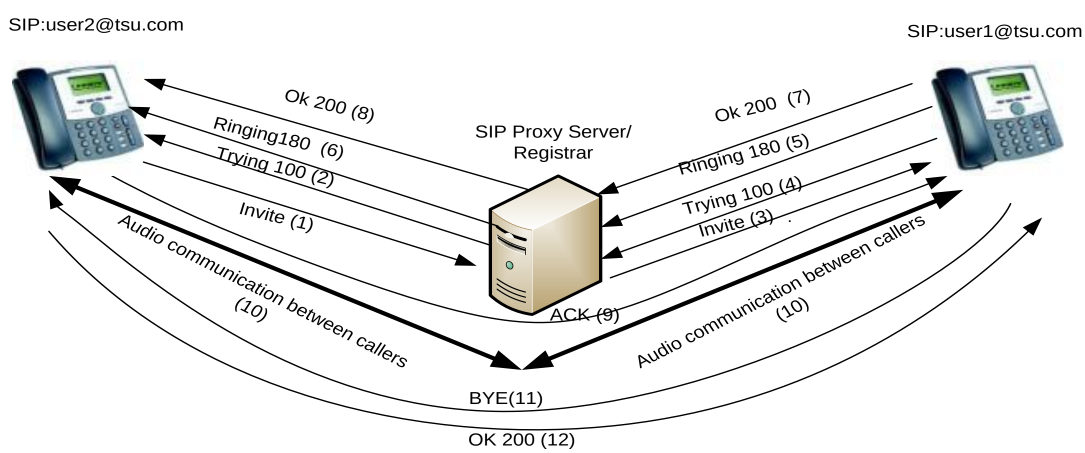


Figure 2 SIP Architecture

SIP solution uses a combination of open source protocols along with the SIP protocol. Though SIP is primarily for setting up and tearing down calls since it was designed to be a signaling protocol, it also permits the functionality of other telephony functions including services such as instant messaging, routing voice mails to mail servers, etc. SIP clients use TCP or UDP to access both the SIP servers and end points. SIP user agents are SIP enabled devices such as handsets, PDA which have the ability to receive SIP calls or make calls to SIP enabled devices. SIP is also known to use the same format as used in the Hypertext Transfer Protocol (HTTP) which also makes it a text based in nature. SIP also has some unique messaging commands which are good to know as they help define the scope and limitations of what the signaling protocol is able to do and what it can't do. SIP also has some important functionality that includes the following:

- User/Call agent – this can function as both a User Agent Client (UAC) that initiates SIP REQUEST calls or (and) a User Agent Server (UAS) which can contact a user client when a request is received and can also respond, if necessary, in place of the user.

- SIP Proxy – This act as both a SIP client and a server and can make SIP requests on behalf of other SIP clients. The server can either be in a state mode to support TCP protocols (or other varieties of services) or stateless which scales better and can also support larger and more volumes of calls.
- Redirect Server – This does the same thing as SIP proxy except it doesn't forward INVITE, but responds with a message notifying the client about where to locate the next hop [3]
- Registrar Server – This usually monitors the valid locations of a client through the acceptance of its registration request information.

1.5 Peer-to-Peer VoIP Solutions

Peer-to-peer (P2P) VoIP solutions provide an infrastructure where communication between parties involved does not pass through a third party after call setup as shown in Figure 3 above. Once the call is established, call traffic flows between the peers directly without going through a centralized server as in the SIP case. It's a technique designed to work in a decentralized server environment for better application sharing distribution. [14]. The P2P client software keeps a database of all the people they can talk to, select from a list, and connect directly to the other clients without going through a third party. Each peer acts as both a server and a client at the same time with full functionalities.

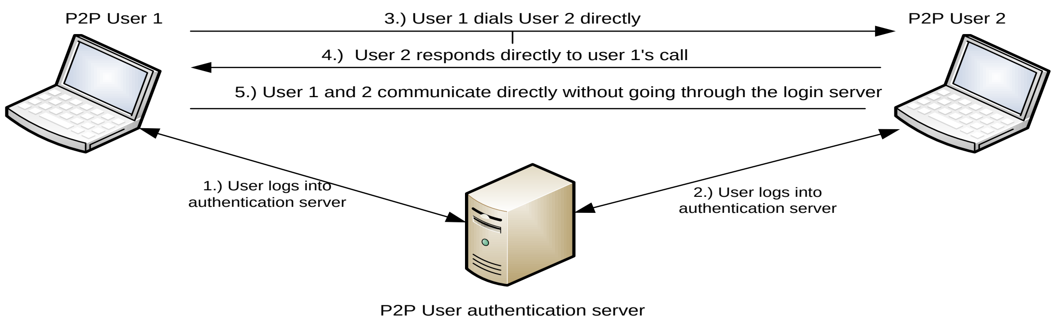


Figure 3 P2P Architecture

The P2P solution stores a list of client database contacts such as phone numbers, the addresses, or other details. P2P VoIP solutions allow their clients to scroll down and select the specific number to dial. The P2P solution has the redundancy of rerouting traffic through different routes on the infrastructure to the congestion on the network infrastructure. Some of the re-routing solutions proposed is the Key-Based Routing (KBR) solution for P2P VoIP communication by Ben Y. Zao *et Al* this involves the usage of a protocol-independent traffic redirection over the mechanism [15]; where nodes with routing tables are used to re-route the information as they travel through the network. Examples of P2P applications are Gnutella, Fast Track, Limewire, PPLive, and Skype. The P2P solution enjoys scalability and has a higher tolerance for redundancy. Unlike the client-server architecture of the SIP, an increase in the number of user does not add to the volume of already congested server, rather this is only added as an extension on a device with full serving capability. They are sometimes called *servents* derived from "SERVer cliENT" [16]. The best form of peer to peer is one where both parties have full information and have same capabilities. The information or content might be different but the capacity and ability of one node must be the same as the other node which is where the advantage of a P2P solution is better utilized.

1.6 Research Motivations

VoIP services have been gaining considerable appeal both from the users' and service providers' point of view and as consumers get used to more and more VoIP services, the demand on higher quality calls is ramping up quickly. However, there are still several challenges that need to be dealt with when VoIP solutions are deployed in IP networks. These challenges are due to the fact that the IP network was originally designed to carry data traffic not multimedia traffic such as video and voice. The characteristics of data traffic and multimedia traffic are totally different and what improves the quality of data traffic might degrade the quality of multimedia traffic. For example, multimedia traffic is more sensitive to time delays than it is to loss while data traffic is more sensitive to loss than to time delays. It has been observed that the voice signals regenerated at the destination, from the received voice packets, are very sensitive to network congestions which lead to delays in the arrival of voice packets.

Sometimes the packets get dropped when router buffers are full or when the network is heavily congested. Voice packets could also arrive at different times and out of the sequence in which they were sent, this is called jitter. This can totally change the original message intended at the source. In this paper we analyze the impact of network configuration parameters on the quality of voice calls in VoIP services. It studies the impact of the network parameters on the quality of voice calls for different types of VoIP solutions and revealing side by side the effects of network parameter variations on the call quality measured by packet loss, jitter, and latency. The rest of this paper is organized as follows: Section 2 includes an overview of the design of the study. Section 3 provides an analysis of the results of the experiment. Section 4 is the conclusion.

2 DESIGN OF THE STUDY

2.1 Study Overview

The purpose of this study is to analyze the impact of network design parameters on the quality of VoIP calls. VoIP applications are different from other applications that generate network traffic. It can be understood that transferring a file from one host to another across the network will be greatly affected by losing a packet which contains a part of the file and unless this packet is retransmitted again, the file will be corrupted at the receiving end. Packet delay, on the other hand, does not affect the correct delivery of the file. It might cause the file transfer to take longer but does not corrupt the file. Contrary to that, VoIP applications are very sensitive to time delays and less sensitive to data loss. Since each VoIP packet contains very short periods of the call (about 20 ms per packet), the person at the receiving end can still understand the conversation even with some packet loss. The impact of packet delay on the other hand can be more severe on the quality of the call. This is due to the real-time nature of VoIP applications.

2.2 Voice Quality Metrics

To study the impact of network parameters on the quality of voice calls, there should be a set of quality metrics used to measure the quality of VoIP calls. There are four key metrics that are commonly used to measure the quality of VoIP calls, these are:

- Latency (end-to-end delay)
- Packet Loss
- Jitter
- Mean Opinion Score (MOS).

These metrics will be used in our study to assess voice quality. The following paragraphs provide some background about these metrics and how they are calculated.

Latency (End-to-End Delay)

It is the measure of the total time delay experienced by voice packets while being transmitted in the network from one end of the call to the other. In this paper the term delay and latency are used interchangeably. Assuming there are N nodes from source to destinations, then the end to end delay is the accumulation of the delays experienced in every node. The following formula can be used to calculate the end to end delay

$$D_{end-to-end} = \sum_{1}^{N} (D_{i\,proc} + D_{i\,queue} + D_{i\,trans} + D_{i\,prop}) \qquad \textbf{(Equation 1)}$$

Where,

$D_{end-to-end}$ is the end-to-end delay; $D_{i\,proc}$ is the processing delay at node *i*

$D_{i\,queue}$ is the queuing delay at node *i;* D_{itrans} is the transmission delay at node *i*

$D_{i\,prop}$ is the propagation delay at node *i*

Jitter

Jitter is defined as the variation in the delay of voice packet from one end to another. It is also sometimes referred to as the variations in the inter arrival time between transmitted packets. Jitter is calculated using the following formula in equation 2.

$$J_i = J_{(i-1)} + \frac{|(R_i - S_i) - (R_{i-1} - S_{i-1})| - J_{(i-1)}}{16} \qquad \textbf{(Equation 2)}$$

Where

J_i, $J_{(i-1)}$ is the current jitter and previous jitter respectively

R_i, R_{i-1} is the arrival time for the current packet and previous packet respectively

S_i, S_{i-1} is the sending time for the current packet and previous packet respectively

Packet Loss

In a regular network infrastructure, queues or buffers are usually configured to hold the data to be transmitted at different stages on the routers or switches. Sometimes, packets arrive to find a full queue and with no place to store this packet, a router will drop this packet. Such packets are referred to as being lost. These packets leave the source but never get to the destination. One of the ways for reducing or controlling the real time voice packet loss is by marking it or by prioritizing the real time voice packet [17]. The greater the amount of packets dropped or lost, the worse the quality of sound. Packet retransmission can congest the network and are better used for top priority data where lost packets can change the whole message. [18].

Mean Opinion Score (MOS) and R-Factor

MOS is a subjective form of deciding voice quality. It is the numerical method of expressing the voice quality and it is measured in a value of 1 to 5 where 5 means a perfect communication synonymous with a face to face quality and 1 means it is impossible to have productive communication [19]. An approach of measuring the voice quality is by using a software model that can be used to measure the value of MOS of the voice quality of a VoIP call. One of these models is the e-model recommended by the International Telecommunication Union (ITU) in the ITU G.107 [20] and [21] documentations. This gives a single scalar output referred to as an R-Factor. Once this R factor is determined, it can be matched to the MOS table to find its equivalent [20]. The R -Factor uses a scale from 100 to 0. The value 100 represents the perfect voice quality and 0; the worst possible quality. The following shows the mapping of the R-Factor to the MOS rating.

R-Factor rating	Likely opinion of human listeners	MOS rating
100-90	Very Satisfied	5.0-4.3
90-80	Satisfied	4.3-4.0
80-70	Some Users Dissatisfied	4.0-3.6
70-60	Many Users Dissatisfied	3.6-3.1
60-50	Nearly All Users Dissatisfied	3.1-2.6
50-0	Not Recommended	2.6-1.0

Table 1: Relationships between the R-factor rating and the MOS rating.

The variables for R factor vary for each codec. e.g I_e which represents the impairment factor for different codec is listed in table 2 below:

Codec	Codec impairment
G.711	0
G.729	11
G.723.1-MPMLQ	15
G.723.1-ACELP	19
G.726	7

Table 2: Codec Impairment

The codec bit rates play a major factor in the quality of the call. Table 3 below provides information about the properties of some commonly used Codec.

Codec	Bit rate(kbps)	Frame time(ms)	Look Ahead(ms)	Codec Impairment
G.711	64	10	0	0
G.729	8	10	5	11
G.723.1-MPMLQ	6.3	30	7.5	15
G.723.1-ACELP	5.3	30	7.5	19
G.726	32	10	0.125	7

Table 3 Codec Properties

In calculating the MOS and R-factor, three of the Codec listed in Table 3 were used. Each one gives what the result would look like the G711, G729, G723, or any of audio codec with sampling rates that fall within the range of the G711, G729 and G723.1 codec. The G711 codec samples at a rate of 64kbits and is better used in environments where the bandwidth is high such as a Local Area Network (LAN) environment. G729 has a sampling rate of 8kbps and is used in low bandwidth environments such as locations with limited internet access. G723.1 is an audio codec used mostly in VoIP environments because it can sample at a rate of 6.3 and 5.3 kbps and works at very low bandwidths introducing more codec impairment in the VoIP system. The MOS values will be calculated using these three codec values for the purpose of knowing how each solution will perform under different circumstances when subjected to codec with sampling bit rates that vary between 64kbps and 5.3kbps. The maximum expected R-factor for G711 is 93.2, while that of G729 is 82.07 and 78 for G723.1.

2.3 Experimental Design

The following paragraph gives an overview of the design of the experiments used in this research. The details of the laboratory setup, network control parameters, and the procedure used for conducting the experiments are explained below.

Experiment Control Variable

In our experiments, we used two network parameters as our control variables. These parameters were modified and the quality metrics discussed above were measured. These two parameters are:

- **Link bandwidth** – It is the transmission rate measured in kbits/second which determines the rate at which voice packets are transmitted from one end to another. This is varied from 23 kb/s to 1024 kb/s.
- **Router buffer size** – Is the size of the memory used by the router as a buffer to hold the packets while they wait to be sent. It is measured in packets and in our experiments it varied from 1 to 100 packets.

Experimental Infrastructure

As shown in Figure 4 above, the infrastructure of our experiments consists of two Local Area Networks (LANs) connected together by a core network. The two LANs are called Houston and Dallas. A SIP server is also available for the users in the two LANS to place VoIP calls using SIP. On each LAN, two desktops with P2P and SIP phones applications installed on them are used to make the calls from on LAN to the other. In each experiment, a call is made from the Houston network to Dallas and the traffic is captured from both networks and analyzed later. To ensure consistency in all experiments and to make sure that any measured quality changes are due to the impact of the control variables only, a recorded voice message that lasts for about two minutes is used for the phone conversation to ensure the same traffic is sent from the caller to the receiver each time. This will ensure that the conversation and duration for all calls made are the same and the only variation from one experiment to another is only the control parameter under investigation.

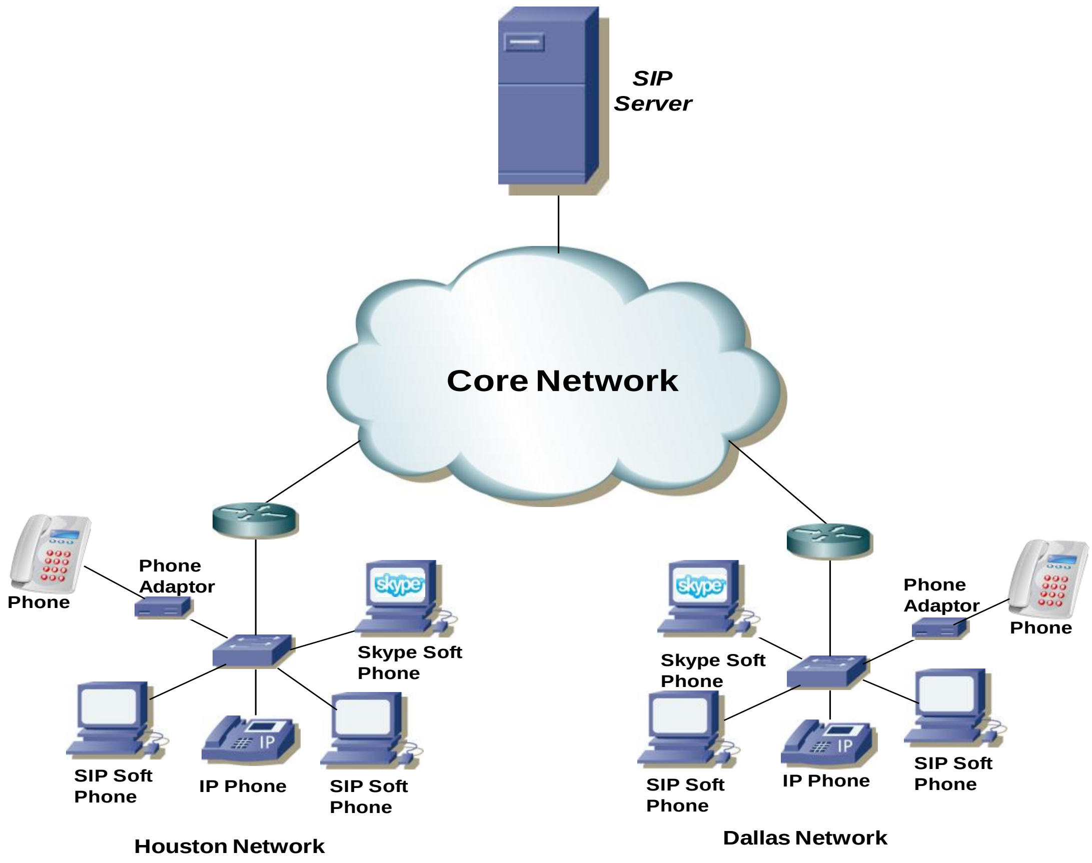


Figure 4 Experimental infrastructure

Experimental Procedure

Our experimental environment is a controlled one in which other sources of traffic over the network are disabled. The only traffic passing through the network is the voice call traffic used for the analysis of VoIP quality. The following are the steps performed to conduct each lab experiment:

1. All network traffic is disabled except VoIP call traffic.
2. The bandwidth and buffer size for the LAN routers are set to a particular value.
3. A VoIP call is placed from the Houston LAN to the Dallas LAN using one VoIP solution (SIP or P2P). The call conversation is one way recorded message that will be played from the caller side once the call is established.
4. Performance metrics such as jitter, delay, packet loss, etc. are measured and recorded during the call.
5. The real time voice signal is recorded at the receiver side.
6. Steps 3 and 4 are repeated for other VoIP solutions.
7. After recording the network performance parameters and the received voice signals for all the VoIP solutions under investigation, we repeat the steps starting from step 2 with new bandwidth and buffer size to perform a new experiment.
8. Once all experiments are finished, the captured traffic parameters are compiled and analyzed.

3 ANALYSIS OF EXPERIMENTS RESULTS

The result of the research experiments carried out in the laboratory is presented in this chapter. The following are the four metrics we collected for every experiment:

- Latency (Delay)
- Jitter
- Packet Loss

3.1 Latency

This is the total amount of end to end delay experienced by the packet from the time it was sent from the source to the time it was received at the destination. In this experiment the overall average delay for both solutions at different bandwidths and buffer size showed similar characteristics. The increase in buffers did not have any impact on the delay. This is expected since the packets do not have to wait due to the fast links.

3.2 Packet Loss

This is the percentage of packets dropped along the path of transmission. Several observations were made during the comparison of the behaviors of packet loss for SIP and P2P. Based on the graphs below, we observed that the amount of packet losses experienced at a bandwidth of 32k was very high while an increase in buffer size didn't make much of a difference even as the bandwidth increased to 64k and above**.** Furthermore an increase in the size of buffer as the bandwidth increased didn't create any major changes in the SIP solution even at high bandwidth especially from the buffer size value of 10 and above. The P2P graph on the other hand showed packet loss for P2P solution was more sensitive to buffer size than SIP solution especially at lower bandwidth. No major differences were observed as the bandwidth increased as seen in the figure below.

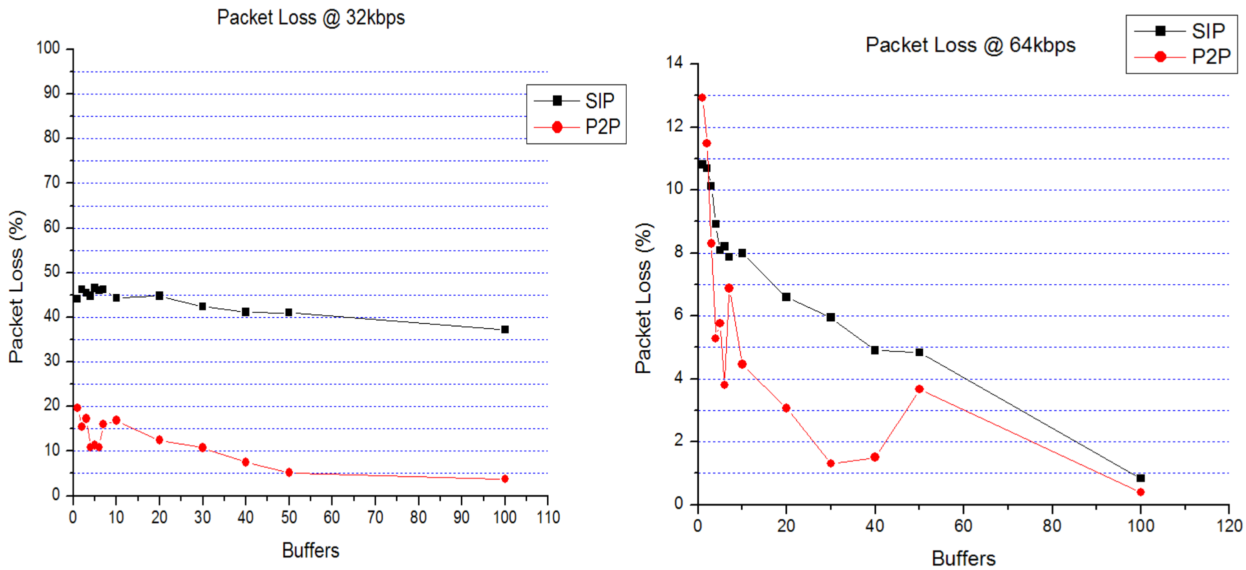


Figure 5 Average delay variations at different bandwidths

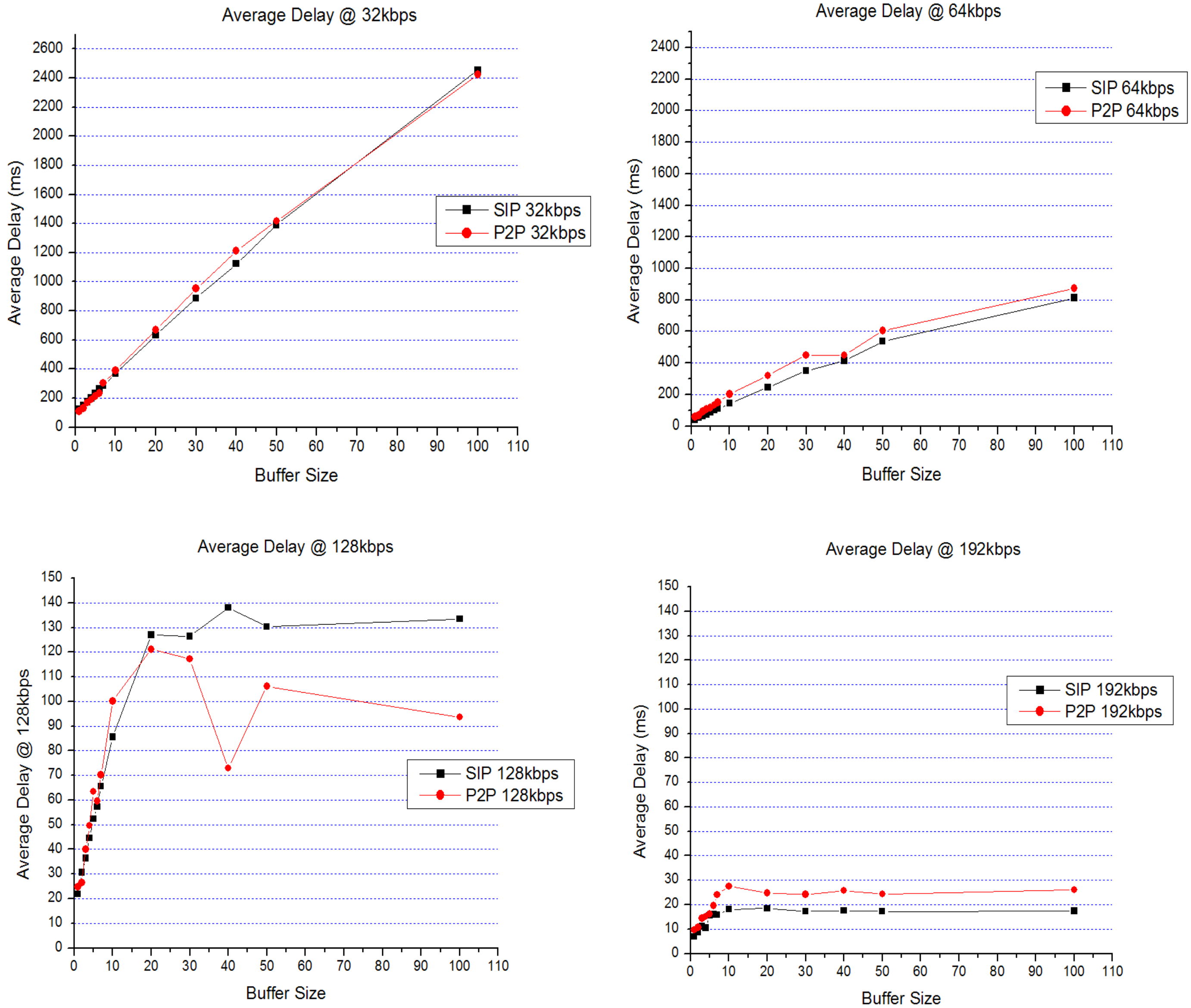


Figure 5 Packet losses at bandwidths of 32k and 64k

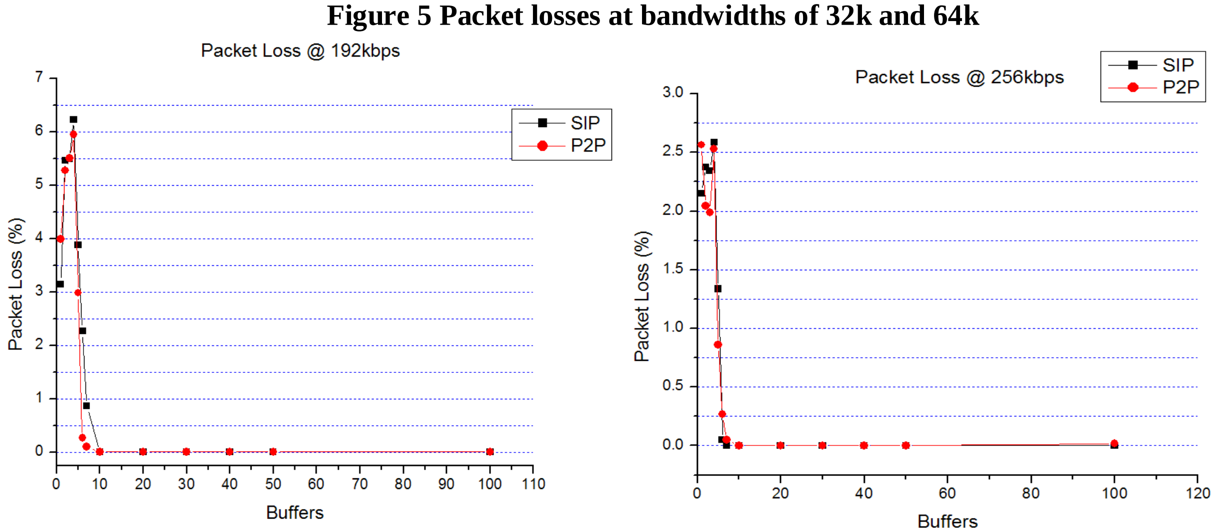


Figure 6 Packet losses at bandwidths of 192k and 256k

3.3 Average Jitter

Average jitter is the average values of the variations in the inter arrival time of packets at the destinations. In the experiment, it was observed that the average jitter value for SIP at the bandwidth of 32k varied intermittently as the buffer sizes were increased and became stable with buffer size above 20.

The jitter value was also observed to be highest at the lowest allocated bandwidth of 32k. Furthermore, the jitter value with a buffer value set to 100 and a bandwidth of 32k was observed to be the same as the value of jitter with a bandwidth size of 64 and buffer size 100. We can deduce that at 32k bandwidth we are able to have a similar sound quality equivalent to what is experienced at a bandwidth of 64k and buffer size of 100. We can also deduce that as the bandwidth increased from 64k and above, an increase in buffer size did not have any significant impact on the average jitter value in the SIP solution. The result looks slightly different for P2P solution. The average jitter measured at 32k was very high compared to the value experienced at 64k. The value experienced at 32k was about 100% above the value experienced at 64k. Furthermore, the value experienced at 64k compared to that experienced at 128k also showed about a 100% decrease as the bandwidth increased from 64k to 128k. As the bandwidth increased above 128k, an increase in bandwidth and buffer size did not have any major significant impact on the P2P solution.

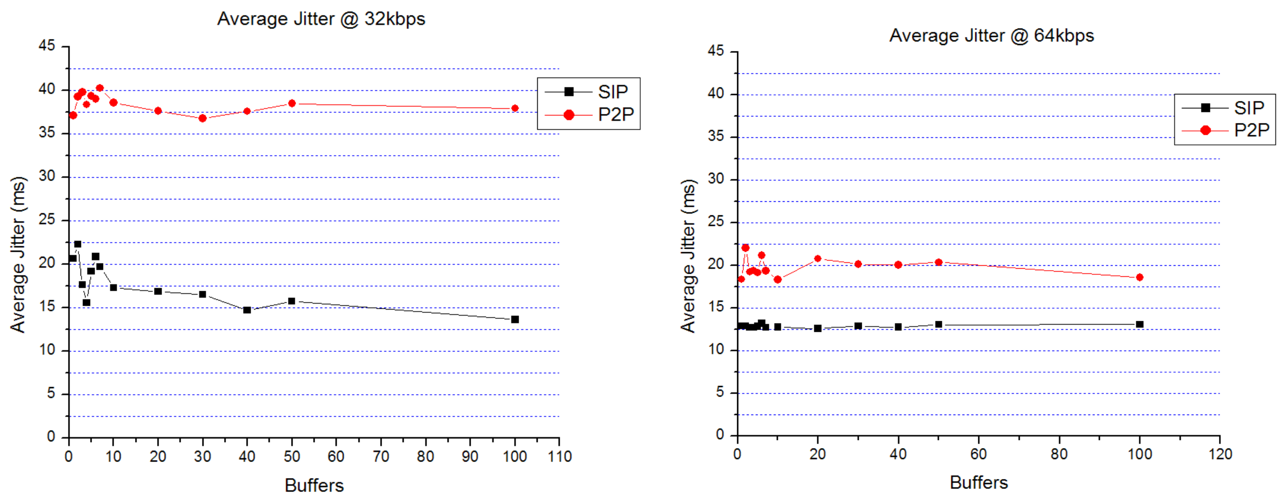


Figure 7 Average Jitter value at 32k and 64k.

As the bandwidth increased, the jitter value differences between P2P and SIP were consistent with SIP always displaying lower jitter values than the P2P. The SIP solution showed more stability with only a slight increase in jitter values at high bandwidth when the buffer sizes were set to values lower than 10.

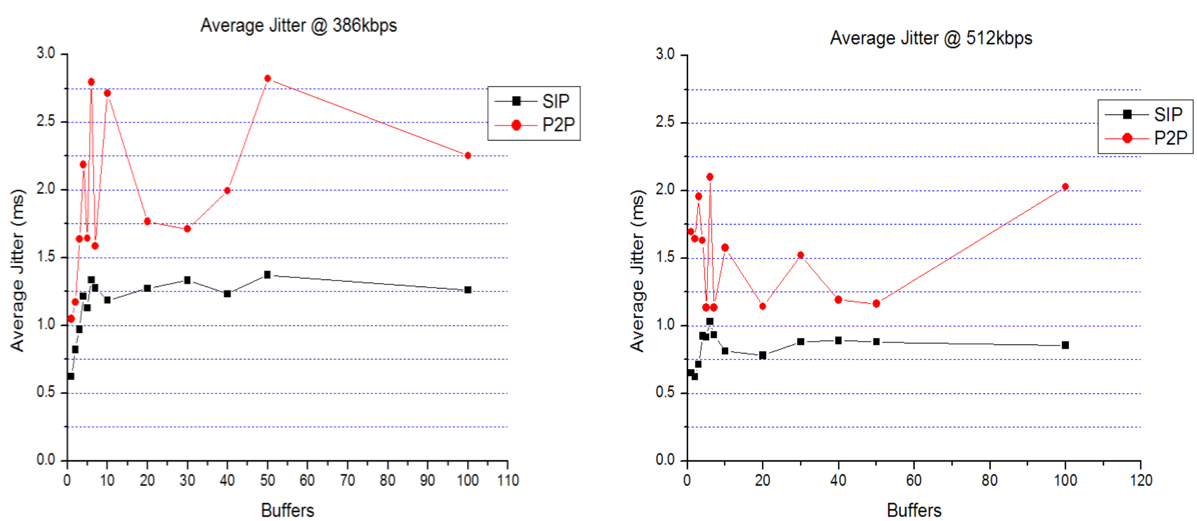
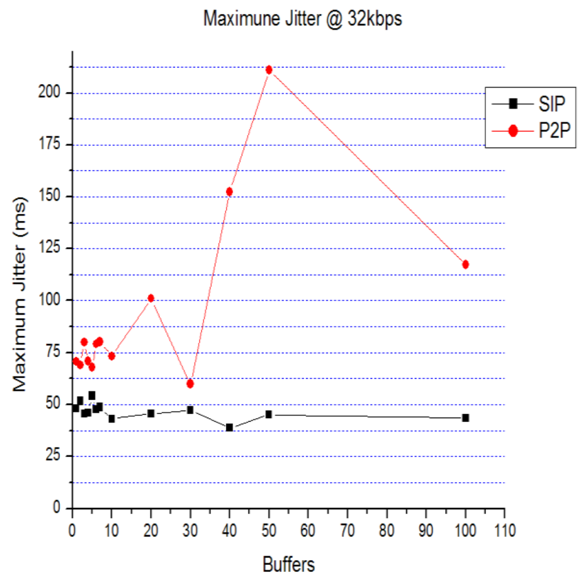


Figure 8 Average Jitter value at high bandwidths.

3.4 Maximum Jitter Value

At a bandwidth of 32k, buffer size variation only began to have significant impacts at the buffer value of 10 and showed a least value when it was set to 30 and peaking at a value of 200 ms when buffer size was set to 50. An increase in the bandwidth from 64k and above showed a decrease in the maximum jitter observed dropping to below 50 ms. Also at a bandwidth of 64k, an increase in the value of buffer size showed no significant impact until the buffer was set to a value above 40.



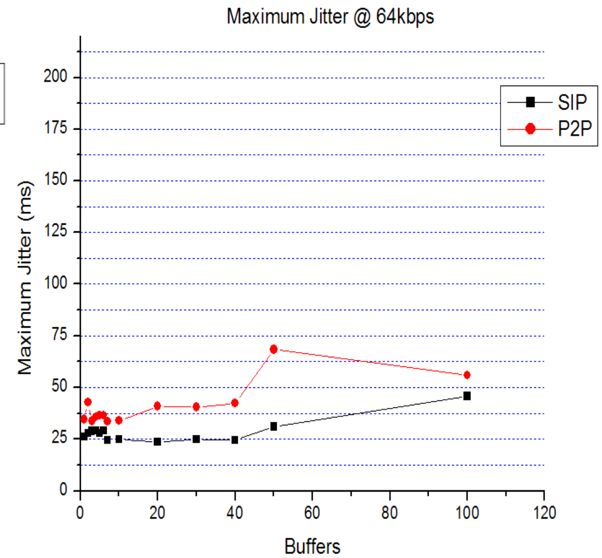
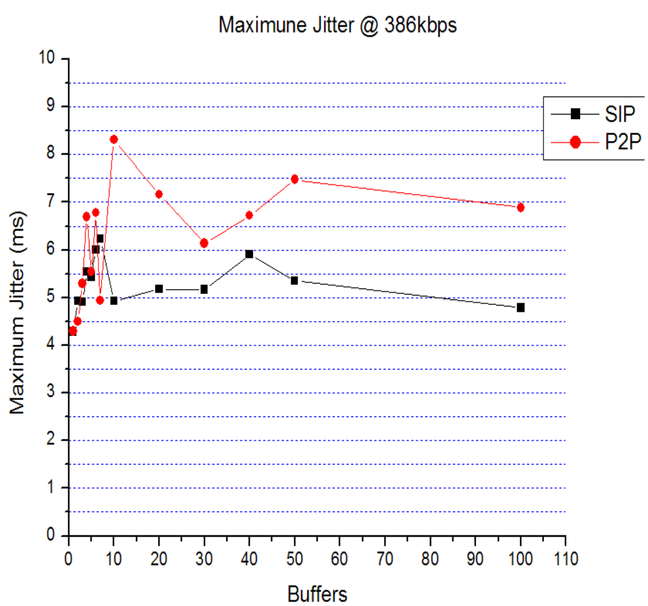


Figure 9 Maximum Jitter value at low bandwidths.

With the bandwidth set to 128k and above, increase in buffer size showed no effect on the maximum jitter values experienced in P2P solution, while the values for the maximum jitter kept decreasing as the bandwidth kept increasing.



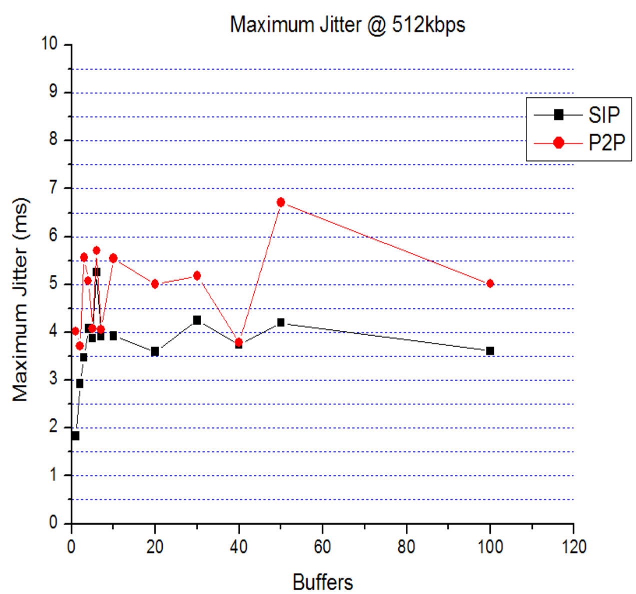


Figure 10 Maximum Jitter value at low bandwidths

4 CONCLUSIONS AND SUMMARY

The results showed that at very low bandwidths, P2P call quality was lower than SIP call quality which is clear from the maximum jitter values measured at different buffer size.

The average jitter recorded by P2P were also constantly higher than that of SIP indicating that P2P experiences more inconsistent delay in packet arrival than SIP at most bandwidths. Furthermore, more packets were lost during SIP calls than during P2P calls at low bandwidth. Both solutions were observed to share the same characteristics at higher bandwidths with respect to packet losses as the bandwidths increased. This indicates that with respect to packet losses, increase in bandwidth did not make or cause much difference. It was also observed that the higher the bandwidth, the lower the amount of packet losses experienced by both solutions, which confirms the traditional results. P2P leveled up with SIP due to buffer size increase and jitter values as the buffer size increased did not have much impact on the SIP solution with respect to both the maximum and average jitter observations.

The average delay observed for both solutions looked similar until the bandwidth value was set to 128k and 192k. SIP seemed to have more average delay at 128k while P2P experienced more average delay at 192k and 256k. The increase in bandwidth had major impact on the average delay experienced by P2P than SIP.

This could be a result of the fact that each packet crossing both ends had more content than that experienced during packet travel in SIP since the SIP server takes care of the packet load off at the end points. Increase in buffer size at the mid-high bandwidth caused a lot more instability in the average delay experienced by P2P solution. Furthermore at high bandwidth, the quality of voice calls for both Peer to Peer and SIP was comparable. The results also show that buffer size plays an important role in reducing packet loss at high bandwidth but this comes at the price of more Jitter as more packets had to wait in the buffer queues.

Further research is underway to study the possibility of creating a P2P-SIP solution such that end points can communicate directly without having to go through a central server once call is established. Also, more investigation on the behavior of VoIP solution will be done while the network has traffic other than VoIP traffic.

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