An Experimental Study of Audio Transmission and Performance in a Mesh Network

Dhivya Padmanabhan* Amit Jardosh Elizabeth M. Belding
Department of Computer Science
University of California, Santa Barbara
*Texas A&M University
dhivya12@neo.tamu.edu
{amitj, ebelding}@cs.ucsb.edu

Abstract
With the growing popularity of voice communication over the Internet and the advancement of wireless technology, there arises a need to evaluate the performance of an audio traffic over a network - more specifically, a wireless mesh network. We establish a point-to-point communication between computers and evaluate the performance of voice packets transmitted between them over the UCSB MeshNet. This enables the analysis of the various factors that are crucial to a satisfactory voice communication like packet loss, latency etc. We believe that this study and understanding of audio performance over our mesh network will provide a basis for future research involving the improvement of voice and other multimedia communication over wireless networks.

1. INTRODUCTION
With the increased deployment of wireless technology, there is a need to test the performance of a variety of useful applications on wireless networks. These applications require a robust network support. For instance, voice communication requires real-time data delivery, minimum packet losses and latency for its applications. Without such efficiency, voice applications and other audio tools over a network would be extremely poor, unintelligent and not deployable amongst common users. Further, both commercially and practically voice communication is one of the most important and common applications. Hence, we need to study the transfer of voice packets over a wireless network and provide a basis for further research in improving the quality of audio transmission. Drawing from the study of multimedia traffic performance conducted on the UCSB testbed in [1], this project will focus on the performance of voice communication on the same. This study involves the setting up of two clients (machines) that initiate a voice communication, a server that connects the two clients, an audio tool that enables the actual audio transmission, a codec to encode the voice packets and a wireless network on which the packets are sent and received. After initiating an audio communication, software to capture the traffic across the network is required to study the various factors that determine the quality of the communication.

Ultimately, this study will enable us to determine the metrics involved in voice communication and how the packet transfer varies across different routes and at different times. Using this information, one can develop codecs and audio tools that function in the most efficient manner over a network. For instance, there is scope for future research to develop intelligent audio tools that pick the shortest routes through a network or employ the best codecs available etc.

2. BACKGROUND

For the study of audio transmission over a wireless network, it is necessary to first understand wireless networking and the mesh network in UCSB.

2.1 Wireless Networking

To understand the transmission of voice packets over a network one must learn the various stages a packet takes to reach its destination - the 'packet' being a formatted block of data to be transmitted over a network. The process of communication over a network includes the five stages or layers (TCP/IP model): Application layer, Transport layer, Network layer, MAC layer and the physical layer.

The application layer is most accessed by programs for communication. The data is created here by various applications like the web, email etc and then encapsulated in a transport layer protocol before transferring the message to the transport layer. Two transport layer protocols are the TCP and UDP. The transport layer is responsible for the successful transmission of the message safely with minimum errors, controlled flow and fragmentations.

The transport layer also sends messages or packets depending on the factors most important to it. For instance, some messages need to be transmitted in real time and can afford packets being dropped and some with assured reliability and order. The connection-oriented Transport Control Protocol ensures reliability; data order and correctness while the connection-less User Datagram Protocol enables real-time delivery. The UDP is best suited for applications like VoIP. The UDP addresses reliability through error detection with a checksum algorithm. The two protocols work on different ports.

The Network layer gets the packet across a single network. The Internet Protocol (IP) (Part of the Network layer) performs the function of routing the packets to its right destination from the source. The different protocols the IP can carry data for include the ICMP and IGMP. All the routing protocols that perform the rerouting functions are part of the network layer as well.

The data link layer or the MAC layer is where the system identifies the route to be taken by a packet for transmission. It provides addressing and channel access control mechanisms making it possible for several terminals to communicate. The MAC layer also provides the MAC address or hardware address which is a unique serial number assigned to each network adapter making it possible to transfer data to the intended receiver.

The physical layer is responsible for the encoding and transmission of the data packets over a network. Some devices of this layer are cables, hubs etc. This layer is studied and explained better mostly by hardware, electric and electronic engineers and scientists.

In short, every packet transmitted carries with it certain header information such as the source IP address, Destination IP address, port number etc which enables it to be routed in the right direction. At first, the message is created in the Application layer. Next at the transport layer is where the port number of the source and destination application is attached to the message as a header. The source and destination IP addresses are added to the header at the Network layer. The MAC layer is where the system identifies the path to be taken for successful transmission and the fifth layer, the physical layer, where the information is physically transmitted.

2.2 UCSB MeshNet Testbed

The University of California, Santa Barbara MeshNet on which the various experiments to analyze voice communication are performed is an experimental wireless mesh network deployed on the campus of UC Santa Barbara. The network consists of 25 nodes distributed on the five floors of the Harold Frank Engineering Building. Every node is equipped with the IEEE 802.11b wireless radio. The 802.11b radios operate in ad hoc mode and connect the various mesh nodes. The purpose of this test bed is to conduct different experiments and research to evaluate the operation of multi hop wireless networks. The mesh gear includes the WRT54G Mesh Router and the mesh gateway. The mesh router is in fact two Linksys WRT54G devices held together. The Mesh Gateway is a small form-factor Intel Celeron based Linux machine. It has a PCMCIA 802.11b radio and Ethernet interface to provide Internet access to the machines on the mesh network and to allow out-of-band management of the mesh gateway [2].

While conducting our experiments, our machines are connected to the network and the various access points are routed to transmit voice packets in between two client machines through the intended paths. (increasing number of hops)

3. EXPERIMENTAL SETUP

In this section, we describe the experimental setup, including the Robust Audio Tool, Client-Server and other useful Perl scripts written and the Network Topology. We also explain the different experiment metrics and why they were chosen.

3.1 Robust Audio Tool (RAT)

RAT is an open-source audio conferencing and streaming application that is user friendly and extremely practical for experimental use for audio communication [3]. It features a point-to-point communication between two participants directly or a group of participants on a common multicast group. RAT only requires a network connection and a sound card. It employs an IP multicast for group conferencing with all participants on a multicast capable network. RAT is based on IETF standards and uses the Real-Time Transport Protocol (RTP) as its transport protocol. RAT also provides a variety of features to counteract various previously observed problems. For instance, it uses Redundancy to resolve packet loss, silence detection for acoustic issues, a range of different rate and codecs for efficient packet encoding and a higher quality sound system relative to the network situations. It uses encryption to assure the confidentiality of the user's audio transmissions. For the purpose of our experiments, RAT proves extremely efficient. The version of RAT available for a Linux platform was installed on the client machines. RAT is only an audio application and does not feature the means to provide services such as finding the user location or IP address etc. The command line usage of RAT is: 'rat (IP address of receiver) / (port number)'.

3.2 Network Topology

We added our two client machines onto the Mesh Network in ad hoc mode. Using the two clients and 3 other nodes placed on various rooms of the second floor of the Harold Frank Building on the campus of UC Santa Barbara, we created a 4-hop path for our voice packets to be transmitted between the

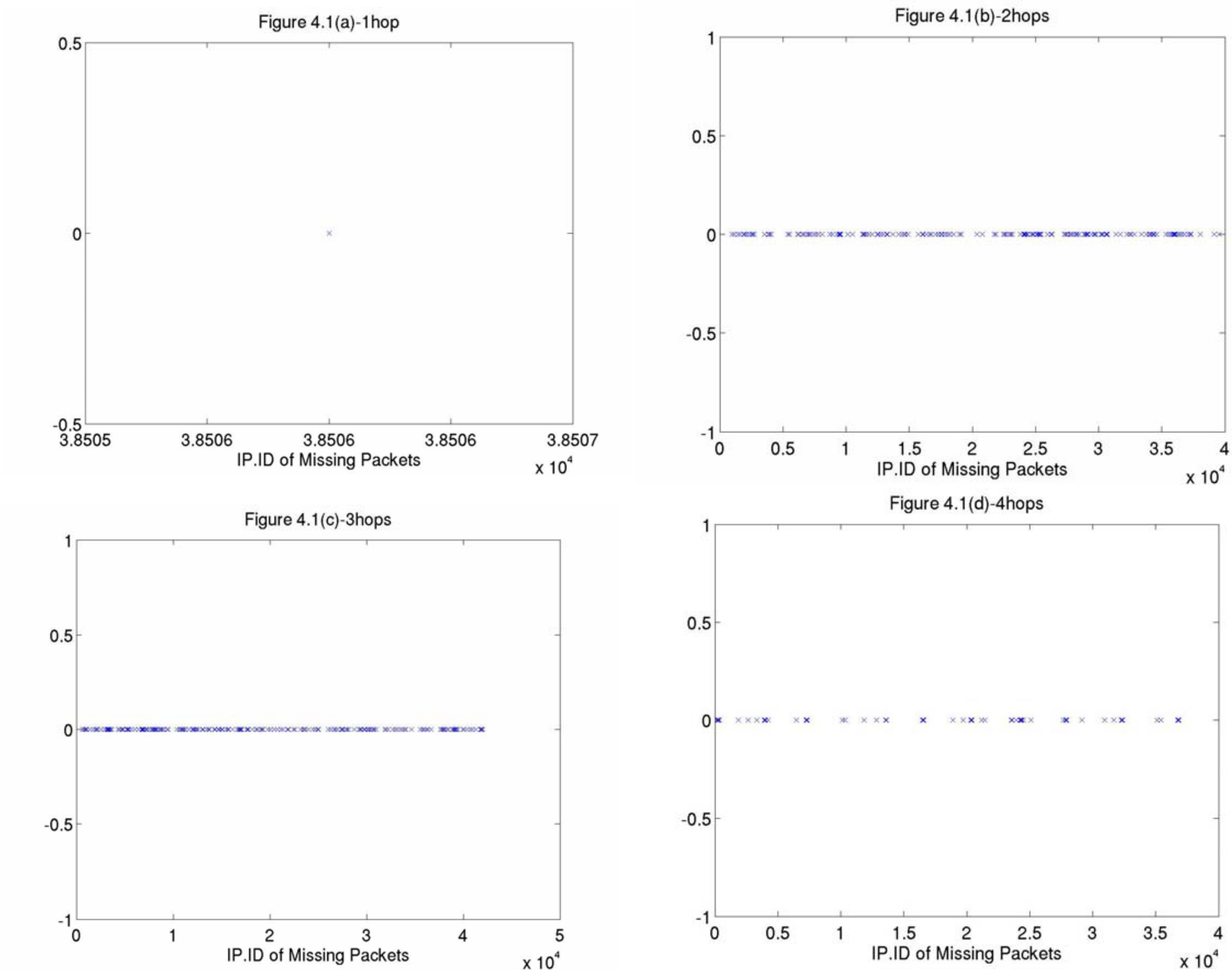


Figure 4.1(a-d): Packet Loss vs. 0

clients. Depending on the experiment, these paths were re-routed to a 3-hop, 2-hop and 1-hop path.

3.3 Perl Scripts and Tethereal

RAT being an audio application only, does not initiate a conversation with a client machine. To resolve this issue we created a low-key SIP (Session Initiation Protocol) that would perform function similar to a directory service on a lower scale. Two scripts client.pl and server.pl were created and saved in the client and server machines respectively. The gateway of our mesh network served as the server machine. The scripts used socket programming to open a connection in between the client computers and the server, to request the IP address of the intended call recipient. The server performs a binary search of the directory listing the IP addresses and names of all registered client computers and provides the correct IP address of the call receiver to the caller (client). The client.pl script initiates RAT with the provided IP address on a priorly selected port (10,000). Once RAT is initiated between the two clients, an Audio CD is run to the length of 3 songs on one client, transmitting the audio to the other. Using tethereal, we capture the flow of voice packets between the clients

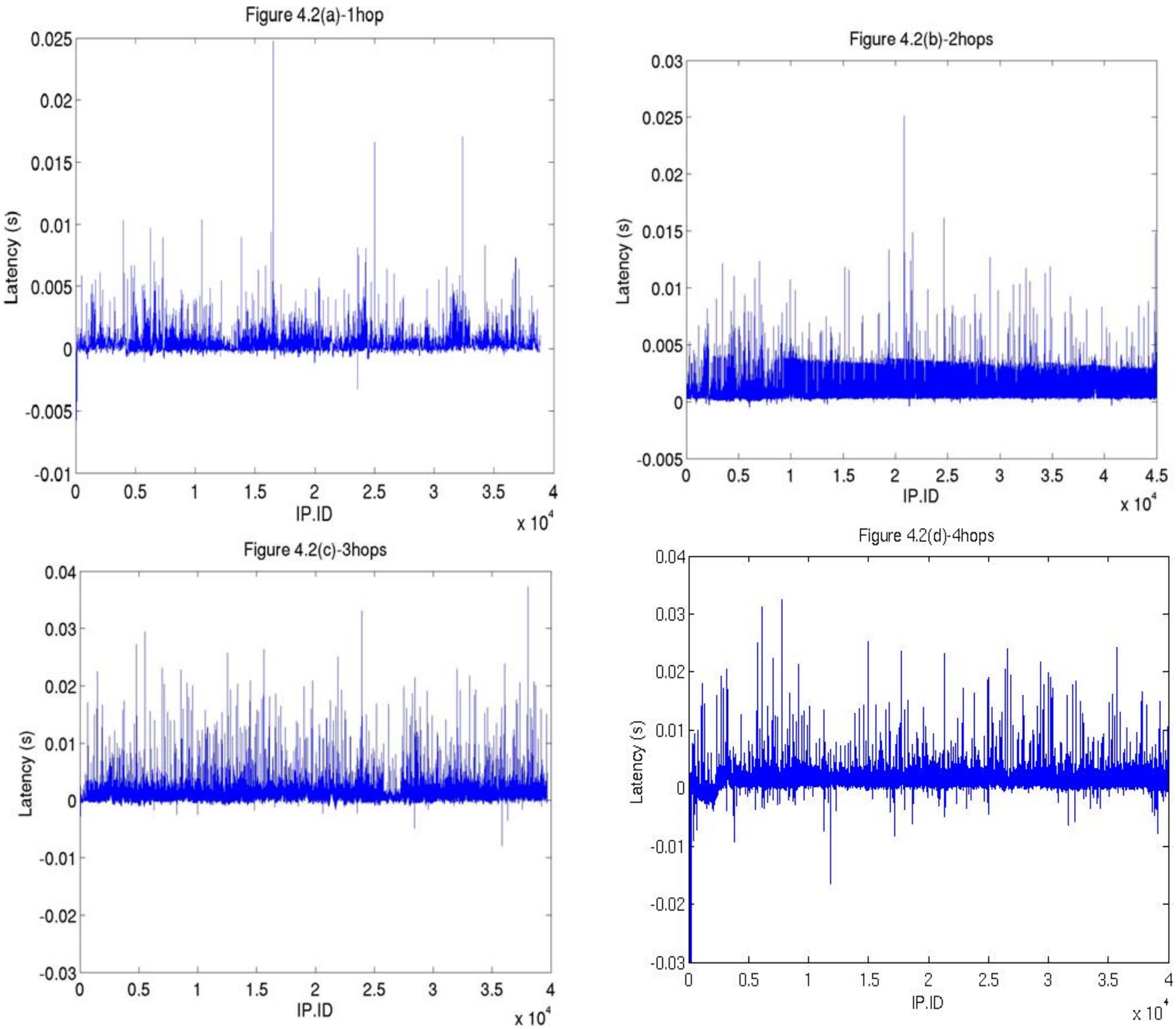


Figure 4.2(a-d): IP.ID vs Latency

in a TCPdump file to further analyze the communication patterns and flaws. The dump file is converted to a more readable format using a parser and this file is analyzed using two scripts get-stats.pl and analyzer.pl. These scripts help obtain the various experiment metrics of the audio transmission.

3.4 Experiment Metrics

The most important feature of a voice transmission is real-time delivery with minimum packet loss. Excessive packet losses would create a discrepancy in the audio received and will degrade the quality of the transmission. Also, delay in packet reception and transmission would create difficulties in real-time voice conversations with applications such as Internet phones etc. Hence the metrics to evaluate the performance during this study are:

- **Packet latency**: the end-to-end (source to receiver) packet transmission delay.
- **Packet loss**: number of missing packets on the receiving end.
- **Inter-Packet delay:** the time difference between successive transmissions.

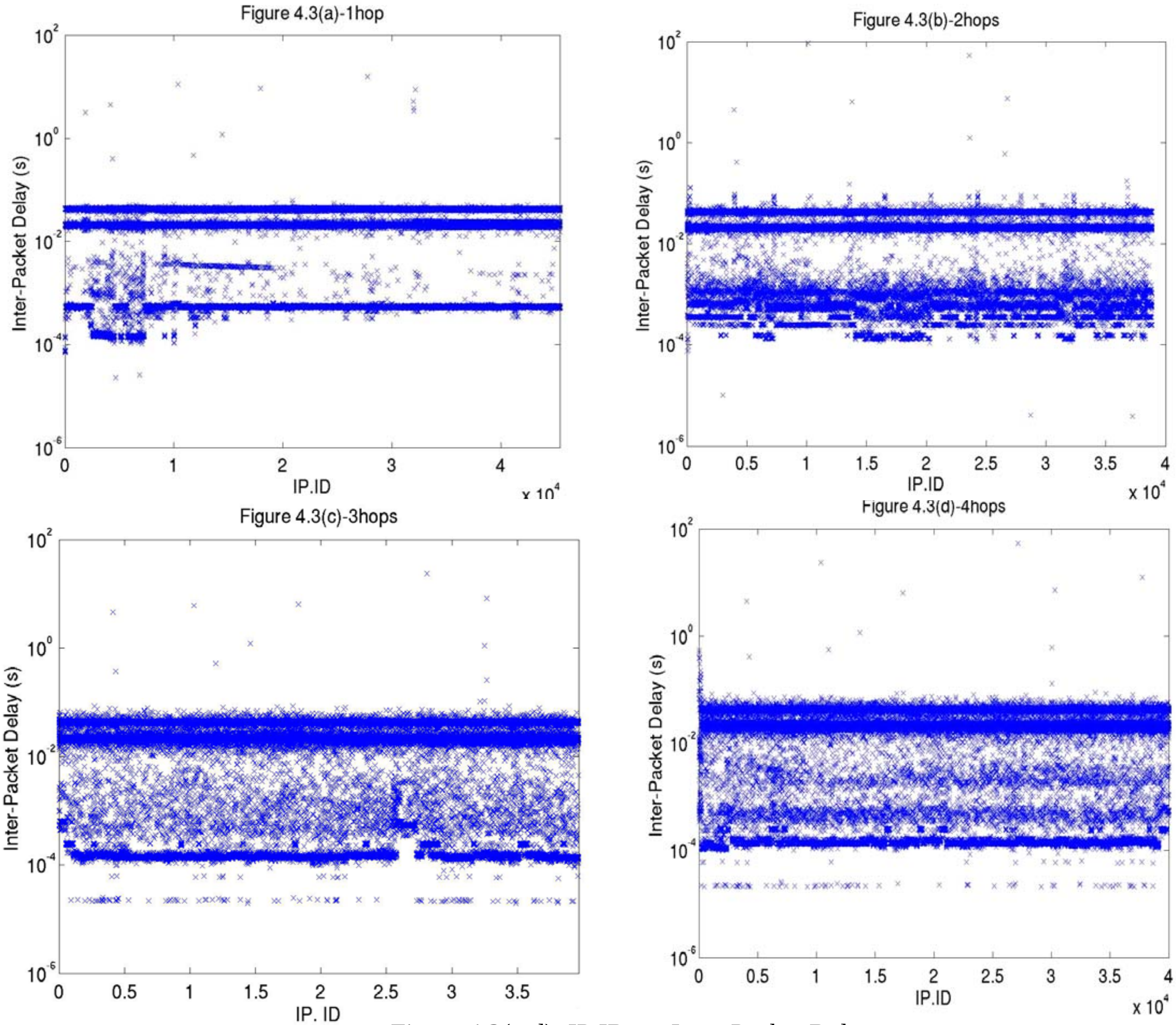


Figure 4.3(a-d): IP.ID vs Inter-Packet Delay

4. EXPERIMENT RESULT

In this section, we evaluate the performance of audio transmission through a multi-hop path in a wireless mesh network. We also explain the cause for certain patterns and results from examining the voice communication process.

4.1 Packet Loss

In an audio transmission, the quality of reception is maintained only if the percentage of missing packets is at least less than 5%. After various trials and experiment runs of a 3-song long audio piece over a multi-hop network with the 1-hop, 2-hop, 3-hop and 4-hop paths being considered, a table of percentage of missing packets for each hop was drawn.

1-hop	**2-hop**	**3-hop**	**4-hop**
0.0022%	0.4510%	0.5047%	0.9286%

Table 1: Missing Packets

Construing from Table 1, we can observe that the loss of packets increases with the number of hops. The 1-hop experiment resulted in only 1 packet loss. This is because; with each hop, there is increased channel access delay and due to the magnitude of packets being transmitted per second, voice packets from the same stream compete for channel access. Thus, the probability of packet loss increases with every hop. It was also observed that the quality of audio decreased with increased loss of packets. From the graphs 4.1(b-d)-Packet Loss vs. 0, we can see that the packet loss is bursty. I.e. there is a high loss rate at particular stretches of time during the transmission process. Also, the bursty nature is more profound in longer paths. This can be attributed to interference occurring at those particular instances.

4.2 Latency

The latency or end-to-end delay increases with longer paths.

#Hops	Standard Deviation (Latency ms)	Average Latency (ms)
1	0.5670	0.1203
2	0.8564	0.7217
3	1.536	0.9817
4	3.0507	2.3616

Table 2: Latency

From Table 2, it can be noted that the average latencies are in the order of milli-seconds. It can be observed from the graphs 4.2(a-d)-IP.ID vs. Latency that, although the average latency is positive and majority of the data points are on the positive y-axis there are a few data points with negative end-to-end delay values. This discrepancy can be attributed to the imperfection in time synchronization between the two laptops. Vertical gaps can be noticed in timestamp vs. latency graphs. This is due to the increased time difference between the transfer of voice packets at the end of one song and start of the next. Also, sudden spikes in latency values can be observed on the graphs. This is due to interferences occurring at that specific instances and re-transmissions of data packets.

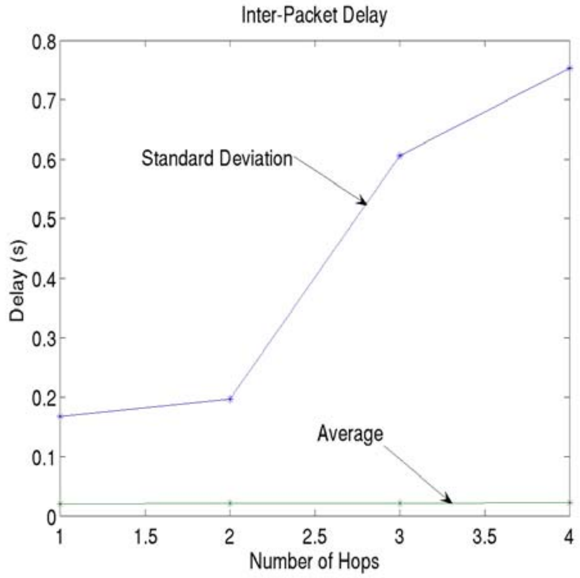
4.3 Inter- Packet Delay

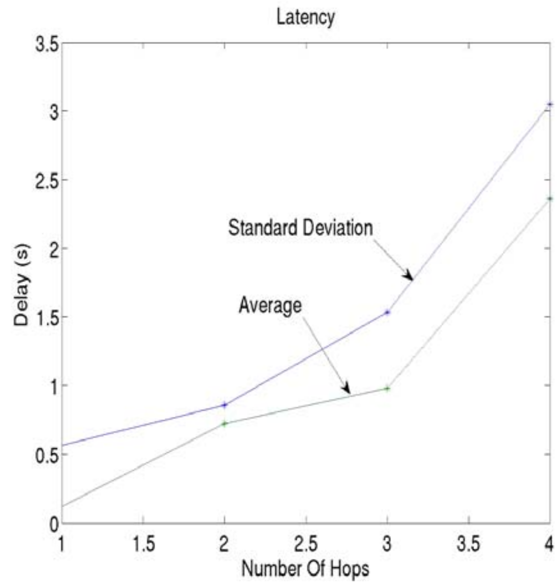
The average and standard deviation of the inter-packet delays at the receiver's end are tabulated in Table 3.

#Hops	Standard Deviation (Inter Packet Delay)	Average Inter-Packet Delay (sec)
1	0.1672	0.02044
2	0.1973	0.02076
3	0.6059	0.02137
4	0.7539	0.02195

Table 3: Inter-Packet Delay

From the graphs 4.3(a-d)-IP.ID vs. Inter-Packet Delay, it can be observed that there are three distinct lines of concentration of inter-packet delay. It can also be observed that this variation decreases with the increase in number of hops. This is due to the manner in which the packets are encoded and sent by the RAT application. With each hop, the propagation delay, queuing delay and channel access delay become more prominent hence decreasing the variation observed in the graphs. Also observed on the graphs are data points with unusually high or low inter-packet delay. This is due to re-transmission of packets from the source side (hence the high inter-packet delay) and interference. The





numbers of re-transmissions are observed to increase with increased path lengths. This can be attributed to the loss of packets due to channel access as discussed with packet loss, hence requiring re-transmissions. Also, it can be seen that unlike the ip.id (Packet ID number) vs. latency graphs, these graphs are plotted with a log scale on the y-axis. This is because the delay values are low and extremely proximal. The log scale provides better clarity for graphical analysis.

4.4 Other Observations

Although the experiments are conducted in a quiet environment at night, we can not prevent random interferences that impact the result. Some random interference such as a microwave operating in the vicinity were far too profound on our results and the data obtained when such interferences occurred needed to be discarded. Also, trial experiments performed during the day resulted in far higher packet loss and delay than performed at night. This is due to the presence of other wireless networks in the building and increased number of users on the network. In our performance analysis, time synchronization between the mesh nodes is needed. The Network Time Protocol (NTP) was applied to eliminate the clock skew [4]. However, our result showed that the time stamp between the two clients was not absolutely precise and it was reflected in our delay computations. But, the application of NTP did improve our results and the graphs for latency etc were more stable and dominantly positive as compared to the results obtained before synchronizing the clocks on the two nodes. Also, it was observed that there was an increased delay between the end of one song and start of next. This was due to the time taken by RAT to start communicating once a continuous audio stream ended. We also performed the same experiment on a wired network to observe the difference. It was observed that the losses and delays were far lower on a wired than wireless network.

5. CONCLUSION

From the study of voice transmission over a wireless network we can gather that the quality of transmission decreases with the increase in the number of hops or path length. We can also see that interference and traffic play a very important role in minimizing the packet loss and delay. Since interference is bound to be present in practical situations and long distance voice conversations are an important aspect of audio communication, this provides for future research in finding means to find solutions for this problem. Further trial experiments showed that the voice performance on a wired network was far better than on the wireless. With the growing deployment of wireless technology, there arises the need to reduce this disparity. Hence, this study outlines the current performance of voice transmission over a wireless network and provides insight for further research areas.

6. ACKNOWLEDGMENT

This work is supported by the Distributed Mentors Program sponsored by the Committee on the Status of Women in Computing Research. (CRA-W)

7. REFERENCES

[1] Y. Sun, I. Sheriff, E.M. Belding-Royer, K.C. Almeroth, An experimental study of multimedia traffic performance in mesh networks, in: WiTMeMo'05: Workshop on Wireless Traffic Measurements and Modeling, Seattle, WA, 2005.

[2] K. Ramachandran, K. Almeroth, and E. Belding-Royer. A Novel Framework for the Management of Large-scale Wireless Network Testbeds. In Proceedings of the 1st Workshop on Wireless Networks Measurements (WinMee), Trentino, Italy, April 2005

[3] V. Hardman, M.A. Sasse, I. Kouvelas, Successful multiparty audio communication over the Internet, Communications of the ACM, v.41, n.5, p.74-80, May 1998

[4] D.L. Mills. Internet Time Synchronization: The Network Time Protocol. In Global States and Time in Distributed Systems. IEEE Computer Society Press, 1994