Performance Measurement for Voice Services in Heterogeneous Wired Networks

Basel Alawieh
Department of Electrical Engineering
Concordia University
Montreal, Quebec,Canada
b_alawi@encs.concordia.ca

Rana Ahmed
Department of Computer Engineering
American University of Sharjah
Sharjah, United Arab Emirates
rahmed@aus.edu

Hussein T. Mouftah
Optical Networks Research Lab
SITE, University of Ottawa
Ottawa,Ontario,Canada
mouftah@site.uottawa.ca

ABSTRACT

Voice over IP (VoIP) service providers implementing different protocols in their core networks are required to coordinate in order to provision, support and deliver voice services to end clients. This paper presents a performance measurement experience for the transport of voice services across multiple heterogeneous wired networks. Our test results show that Multiple Protocol Label Switching (MPLS) protocol slightly outperforms IP in delivering voice services. Moreover, the type of call signaling protocol used, tunneling category adopted and the type of Virtual Private Network (VPN) implemented affect the overall quality and performance of voice services in terms of jitter and delay.

Categories and Subject Descriptors

C.4 [**PERFORMANCE OF SYSTEMS**]: Measurement techniques , Performance attributes, Reliability, availability, and serviceability

General Terms

Measurement, Performance, Design, Testbed, Reliability, Experimentation,

Keywords

Testbed, VoIP, SIP, H.323, QoS, Tunneling

1. INTRODUCTION

Recent advances in Voice over IP (VoIP) technologies has provided tremendous opportunities for service providers and enterprises, as one can use a single IP (Internet Protocol) network for both data and voice applications in a very cost-effective manner. Service providers are now leveraging VoIP technologies to introduce a variety of new services and applications to their customers.

One important issue for service providers faced during the deployment of VoIP infrastructure is the provision and maintaining high-quality voice services to the clients. Quality support for voice services [4] remains an important concern for multiple service operators besides security. The situation becomes even more challenging when VoIP technologies are used to provide voice service to remote network sites over heterogeneous networks implementing dissimilar protocols. Packet loss, delay and jitter degrade the quality of service in VoIP, especially in heterogeneous networking environment. Therefore, an experimental measure of those parameters is strongly needed in the planning process of new services and applications over such heterogeneous networks.

VoIP phones consist of encoders and decoders (also known as voice codecs). At the sender side of a VoIP session, the voice signal is sampled, digitized and encoded. Encoders encode the speech samples and compress them into a frame. At the receiver's side, the decoder receives the encoded frame and regenerates the original speech samples after passing through a playout buffer to smooth out the variations of jitter.

There are several attributes that affect the performance of codecs. A representative list of standard voice codecs [3] and their corresponding bit rates is shown in Table 1.

Table 1. CODEC TYPES USED IN VOIP

CODEC	BIT RATE (Kbps)
G711	64
G723.1	53,63
G726	16, 24, 32, 40
G728	16
G729	8

The voice service quality of a VoIP session is determined by three major factors: delay, jitter, and packet losses. Delay is introduced at both the end hosts (sender and receiver) and the underlying network. In particular, the delay introduced by the end hosts includes codec delay and playout delay. The codec delay is incurred by the encoding and packetization process, and is usually fixed for a given codec. The delay introduced by the underlying network consists of the transmission, propagation, and queueing delay in the network.

The quality of a VoIP session is typically measured by the ITU-T E Model [2]. In this model, a subjective quality score, called Mean Opinion Score (MOS), is defined as the perceived

TRIDENTCOM 2008, 17th – 20th Mar 2008, Innsbruck, Austria.

DOI 10.4108/tridentcom.2008.3222

VoIP quality. The MOS indices allow five different levels of customer satisfaction for the voice quality, as shown in Table 2.

In this paper, we propose a testbed experience to measure performance parameters for voice services over heterogeneous networks implementing dissimilar protocols. We used the metro-based testbed environment at Optical Networks Research Lab (ONRL), University of Ottawa. We present the procedures needed to install and configure a Virtual Private Network (VPN) voice connection. We present the results of experiments conducted in the testbed environment in order to study the delay, jitter, and packet loss rate. First, we consider the transport of voice services across layer 2 using Stacked Virtual LAN Services (S-VLANs). Secondly, the transport of voice services using IP and MPLS are considered. We then introduce different domain networks and study the performance of the delivery of voice services across them. Two different VPN protocols have been adopted to ensure secure communication services, namely MPLS VPNs and IPSec. We used H.323 and SIP call signaling protocols. We used tunneling techniques, based on [1], to provide the connectivity across heterogeneous networks.

Our tests show that MPLS slightly outperforms IP in delivering the voice services. Moreover, the experiments show that the type of call signaling protocol used, tunneling category adopted and the type of VPN implemented affect the performance of voice services delivery in terms of jitter and delay. We conclude our implementations by reporting Mean opinion score (MOS), defined by ITU-T [2], for each network voice services transported. The paper is organized as follows. Section II presents a technical review on the delivery requirements of VoIP services. In Section III, we describe the experiences followed in carrying out the experiments in the optical testbed environment, including testbed infrastructure used and also provide analysis for the results obtained. Finally, section IV concludes the paper.

TABLE 2. MEAN OPINION SCORE (MOS)

MOS Index	User Satisfaction
5	Very satisfied
4	Satisfied
3	A few users dissatisfied
1	Everyone dissatisfied

2. VOIP REQUIREMENTS

Service providers are required to maintain the MOS indices of voice quality defined by ITU-T [2]. The transport of voice over an IP network imposes some constraints on the level of performance expected from the data network due to the real-time nature of the voice transport. Some of those major performance factors that affect MOS index are: transmission delay, variation of transmission delay (Jitter), and packet loss ratio. The following requirements must be met by the service providers' network transporting voice services [3].

- For a very good audio quality (close to the MOS index 4), network round trip delay must be less than 150ms; jitter must be less than 20ms; and packet loss must be less than 1%.
- For an acceptable audio quality (slightly superior to the MOS indice of 3), the network round trip delay must be less than 400ms; jitter must be less than 50ms; and packet loss must be less than 3%.
- A VoIP communication cannot go through more than two voice compressions/ decompressions (with G729A or 723.1)[3].

Trade-offs can be made between bandwidth/quality and the cost. For example, a client's objective can be a very high level of audio quality on the LAN (equivalent to Time Division Multiplexing (TDM)), and a high audio quality across the WAN. As the client (or client's enterprise) does not control (or own) the WAN, the voice transport cost reduction across the WAN needs to be considered. In other words, bandwidth optimization aspects are more important to consider than voice quality in WAN configuration. One way to optimize the bandwidth usage is to use compression algorithms (i.e., to use codec G723.1 or G729A). On the other hand, if there is no bandwidth restriction (e.g. in a LAN or MAN configuration), G711 is strongly recommended to be used as it has better voice quality. In the case of G729/723.1, the user satisfaction cannot be better than MOS score of 4.

The two most popular signaling protocols used in VoIP are Session Initiation Protocol (SIP)[5] and H.323. SIP is an Internet Engineering Task Force (IETF) standard protocol for establishing, manipulating, and tearing down an interactive user session that involves multimedia elements such as audio, video, instant messaging, or other real-time data communications [3]. Even though H.323 was the first protocol, VoIP analysts predict that SIP will play a major role in the coming years. SIP is considered easier to implement and use as compared to H.323.

3. EVALUATION OF VOICE PARAMETERS

3.1 Overview

We compare the performance characteristics of various Quality of Service (QoS) implementations and their impact on the end-to-end data/voice connection. In each case, an end-to-end virtual connection is established using a tunnel or a series of concatenated tunnels spanning different heterogeneous provider domains, each implementing a different tunneling technique. The concatenated tunnels constitute a virtual connection that interconnects remote customer VLAN sites. We used a testbed environment [7] at Optical Networks Research Lab (ONRL), University of Ottawa, to implement and test the setup. Either H.323 or SIP has been used for performance comparison in each testbed scenario implementation. For our tests, the Real-time traffic tool [6] generator was used to generate RTP traffic flows and measure throughput. This tool is capable of measuring a number of other QoS parameters, including bandwidth, delay jitter, and packet loss. 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. The packet loss values are represented in the percentage form of the total RTP packets being transmitted.

3.2 Description of Testbed Hardware

The testbed [7] used to carry out our experiments consisted of equipment from Nortel Networks, Juniper and Navtel. The testbed setup is shown in Figure 1. The Nortel Optera Metro 8000s (OM-8000s) MPLS switches have been equipped with Gigabit Ethernet (GigE) ports, and have been used in the Layer 2 MPLS domain. The Passport 8600s (PP-8600s) router/switches were equipped with GigE ports and were used in the IP and Stacked VLAN domains. The Juniper M-10 and M-160 were equipped with GigE ports and OC-12 ATM over SONET interfaces, and were used as Provider Edge (PE) routers in the Layer 3 MPLS based network and also as IP based domains. Two Intel-based Pentium IV machines equipped with a GigE port were placed in each of the customer's remote VLAN-sites, and were used as traffic sources/sinks in our tests.

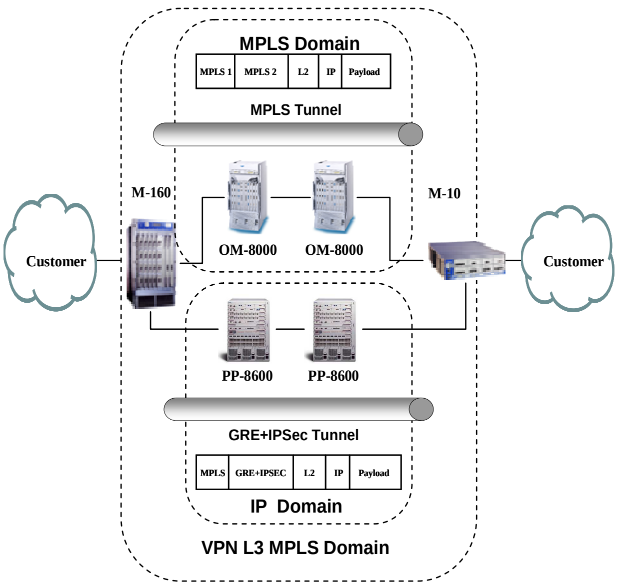

Figure 1 Testbed setup

3.3 Experimental Setup

The purpose of the experiments was to transport voice traffic over networks running IP or MPLS or combination of both. Data traffic was injected with voice traffic for quality of service verification. We also considered transporting voice services over IP-based network with IPsec. In this setup, the two Junipers M160 and M10 were configured as IP-based network with IPsec tunnel carrying voice traffic. Transporting VPN voice services over MPLS Layer-2 and Layer 3 based network was another defined scenario for experimentation. In this setup, MPLS Layer 2 and Layer 3 VPNs are configured in the two Juniper routers, M160 and M10. Carrying VPN voice services using the S-VLAN technology is also considered and the two Passports 8600s are configured to perform that without the two Juniper routers. The VLANs' voice traffics are carried from the customer sites to the provider's side represented by the two passports 8600. At provider's side, the traffics are aggregated or combined/separated with stack label and delivered as a unique VLAN to the other PP-8600. Upon receiving the unique traffic, the PP-8600 pops up the stacked label and redistributes the voice VLANs to their intended destination. Finally, we considered the transport of voice services across heterogeneous network, and two heterogeneous scenarios are considered. In scenario 1, the provider's core transport infrastructure supports MPLS. In scenario 2, we used GRE with IPsec in the case where no MPLS support exists in provider's infrastructure. MPLS Layer three VPN is configured across the two Juniper routers. A point-to-point MPLS layer two VPN is provisioned across the two OM- 8000s. In this way, the voice packets are encapsulated over two MPLS headers and then extracted at the edge router. For scenario 2, in order to transport voice packets over IP-Network, an IP/GRE with IPsec tunnel is first established between the Juniper edge routers. The voice packets are first encapsulated over MPLS and then encapsulated over IPsec/GRE packets. At the egress edge router, the IPsec/GRE headers are extracted, the voice services are restored back, and forwarded to the customer network.

3.4 Experimental Results

Figures 2 to 11 show the performance of the voice flows in all scenarios. Securing the voice services with IPSec and MPLS VPNs increased the QoS measured values for both SIP and H.323 since IPsec adds extra overhead. In our experimentation, H.323 showed better signaling capabilities for voice services. It may be concluded from the results that the call signaling protocols, including H.323 and SIP, produce different jitters. In this context, H.323 offers significantly lower jitter values for RTP packets as compared to SIP. However, both SIP and H.323 are not significantly apart with respect to the delays and the packet losses. H.323 entities use a reliable transport for signaling where most SIP entities use an unreliable transport for signaling. Although delivering voice services across S-VLAN based network is the best fit in terms of delay, jitter and packet loss, S-VLANs backbone suffers from scalability issues. An S-VLAN enabled router can support up to 4096x4096 voice calls. Tests results also showed that an MPLS service is not the solution for VoIP. MPLS service is in fact essentially comparable to Internet service; both provide good base connectivity, but they themselves cannot deliver the quality and availability required for business-quality voice communication. For the last two test scenarios, protocol heterogeneity in networks deteriorates the voice service performance. It was noted that, in both heterogeneous cases, the packet loss can reach up to 50%. This is due to the GRE tunnel with MPLS overhead added to the voice packet. The MOS for the different scenarios is shown in Table 3.

4. CONCLUSION

This paper presents a testbed experience to measure performance of voice services across multiple heterogeneous networks. We describe the architecture for the testbed. The test results indicate that Multiple protocol Label Switch (MPLS) protocol slightly outperforms IP in delivering the voice services. Moreover, the experiments have shown that the call signaling protocol used, and the security requirements affect the QoS parameters, and hence the overall quality of voice services (i.e., the MOS score) in a VoIP session.

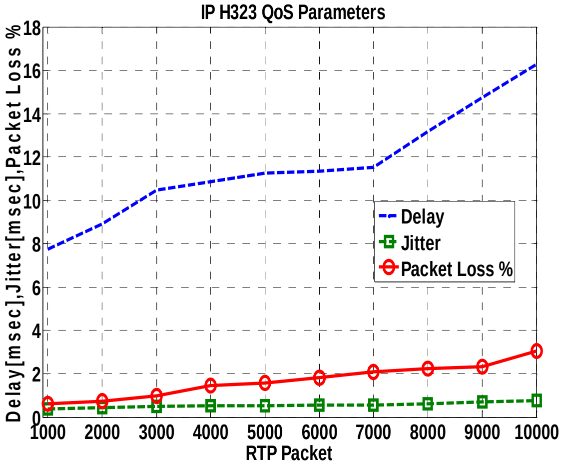

Figure 1: Voice over IP Domain -H323

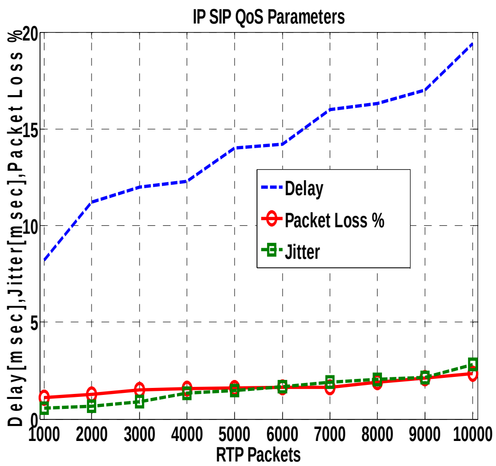

Figure 2: Voice over IP Domain -SIP

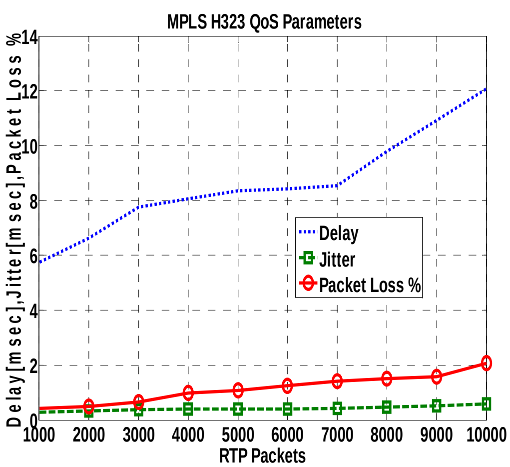

Figure 3: Voice over IP over MPLS Domain –SIP

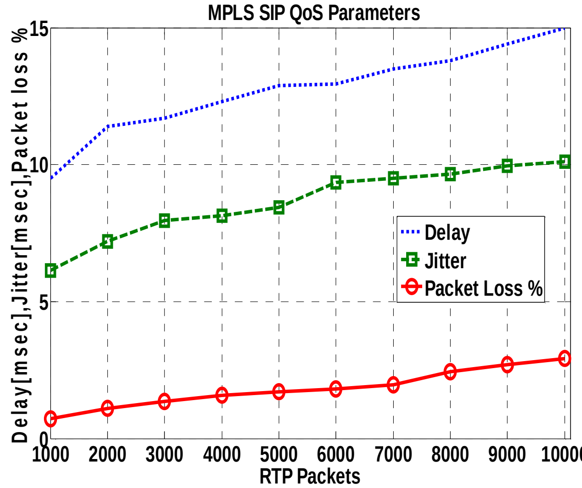

Figure 4: Voice over IP over MPLS Domain –SIP

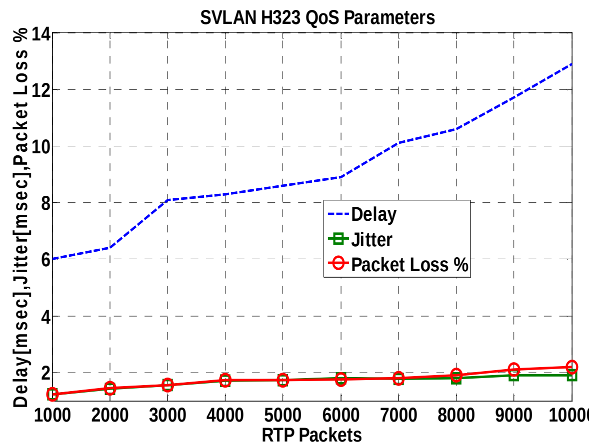

Figure 5: Voice over Stacked VLAN Domain –H323

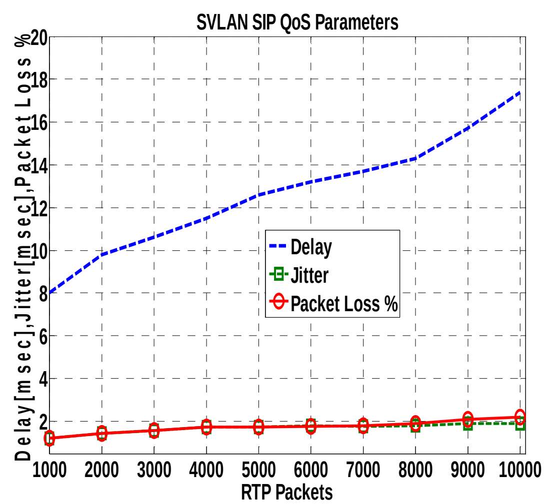

Figure 6: Voice over Stacked VLAN Domain –SIP

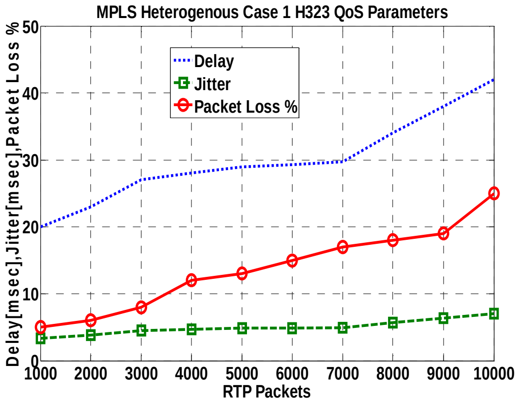

Figure 7: Heterogeneous Network Scenario 1 –H323

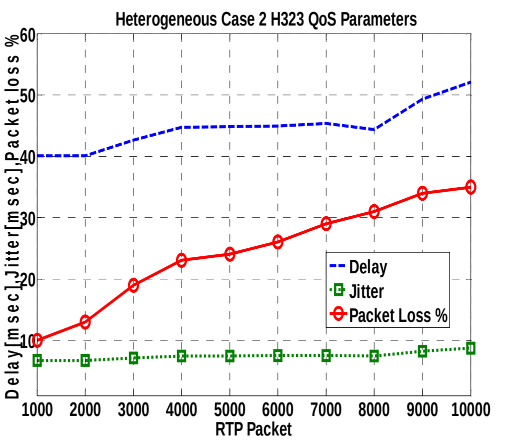

Figure 8: Heterogeneous Network Scenario 2 –H323

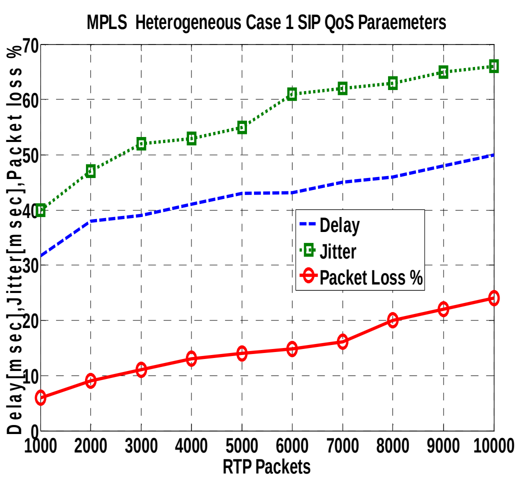

Figure 9: Heterogeneous Network Scenario 1 –SIP

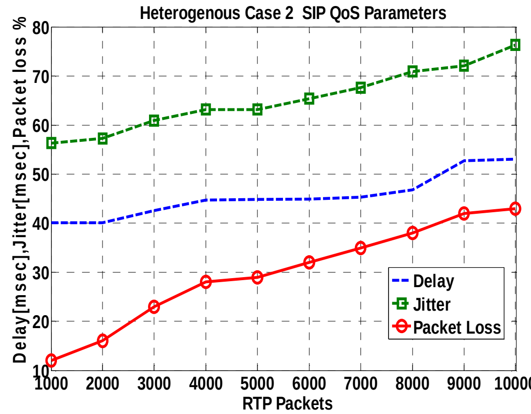

Figure 10: Heterogeneous Network Scenario 2 –SIP

TABLE 3. MEAN OPINION SCORE FOR SEVERAL TESTBED SCENARIOS

Network	MOS Index
IP	4
MPLS	4
SVLAN	5
Heterogeneous Case 1	3
Heterogeneous Case 2	2

5. REFERENCES

[1] T. Saad, B. Alawieh, S.Gulder , H.Mouftah “Tunneling techniques for end-to-end VPNs: generic deployment in an optical testbed environment,” IEEE Communication Magazine, vol 44, issue 5, pp. 124- 132, May 2006..

[2] International Telecommunication Union : http://www.itu.int/rec/T-REC-P.800.1/en.

[3] V. Kumar, M. Korpi and S. Sengodan. IP Telephony with H.323: Architectures for Unified Networks and Integrated Services. John Wiley and Sons, New York, 2001.

[4] S. Shenker, C. Partridge and R. Guerin. Specification of Guaranteed Quality of Service. Request for Comments RFC 2212, Internet Engineering Task Force, September 1997

[5] ETF SIP Charter, http://www.ietf.org/html.charters/sip-charter.html

[6] Real time Traffic Tool Generator http://www.cs.ucr.edu/~msamidi/projects.htm

[7] http://onrl.site.uottawa.ca