EXPERIMENTAL ASSESSMENT OF VOIP INDIVIDUAL AND TRUNKING TRAFFICS

By

Chia-Long Wu

SUBMITTED IN PARTIAL FULFILLMENT OF THE
REQUIREMENTS FOR THE DEGREE OF
MASTER OF SCIENCE
AT
NATIONAL CHIAO TUNG UNIVERSITY
1001, DA-HSUEH ROAD, HSIN CHU, TAIWAN 30050, R.O.C.
May 29, 2000

NATIONAL CHIAO TUNG UNIVERSITY
DEPARTMENT OF
COMMUNICATION ENGINEERING

The undersigned hereby certify that they have read and recommend to the Faculty of Graduate Studies for acceptance a thesis entitled "**Experimental Assessment of VoIP Individual and Trunking Traffics**" by **Chia-Long Wu** in partial fulfillment of the requirements for the degree of **MASTER OF SCIENCE**.

Dated: May 29, 2000

Supervisor: ______________________
Po-Ning Chen

Readers: ______________________

____

____

Table of Contents

List of Tables

List of Figures

Abstract

We have developed a UDP-based measurement tool to simulate the VoIP traffic behavior of different types of software gateways. Two types of software gateway architectures are considered: multiple-connection, where each VoIP call induces an individual connected session over the Internet, and trunking, where multiple VoIP calls share a common Internet session. Our measurements are based upon the two-way multiple-users traffic behavior over the best-effort datagram services of the Internet. From long-term and short-term trends of our measurement, we point out some implementation considerations of pure software-based VoIP gateways. It is out hope that our measurements and conclusions can be a useful reference for the VoIP service providers over the Internet.

Acknowledgements

I would like to thank Dr. Po-Ning Chen for his encouragement and guidance. This work would not have been possible without his advice and commitment. I would also like to thank all the people who have helped me, especially the members of Network Technology Laboratory at Department of Communication Engineering, National Chiao Tung University.

Chapter 1

Overview

During the past several years, there has been a significant increase in the use of packetized audio over wide-area, packet-switched networks. This is induced by the availability of low-cost, toll-quality audio that can be possibly supported by today's Internet. At present, the Internet is indeed being used to carry voice conversations in certain specific applications.

There are three technology issues in today's Internet telephony. First, as for the integration of packet voice and telecommunication systems, Internet telephony must support telecommunication system signaling in order to have seamless support to voice calls. Secondly, codec should be adapted to get enough voice quality on unreliable transmission. To employ different codecs for different applications over Internet is seemingly a trend. Finally, due to the best-effort characteristic of the current Internet delivery service, the quality of service (QoS) has not been guaranteed. As such, how to make Internet to support acceptable QoS guarantee is still an important research topic.

The advantage of transmitting packet voice over Internet is perhaps the low price. Due to the limited bandwidth of the Internet, voice data often have to be compressed before its transmission. To furthermore save the bandwidth, voice packets are only sent at the time the speaker is talking. Since the Internet bandwidth is shared among all users, some sort of unreliability should be encountered during packet transmission. Hence, one may experience that the voice quality is acceptable at one time, but is terrible at another time. In conjunction with a hop-by-hop resource reservation protocol such as RSVP, end-to-end capacity can be set aside for real-time traffic. However, it is not yet popular over Internet. The current dominant protocol for transmitting multimedia in packet-switched networks is RTP/RTCP. RTP, the Real-Time Protocol, is a generic mechanism for carrying data with real-time properties, e.g., audio and video. The headers of RTP provide the sequence number and time stamp information necessary to re-assemble a real-time stream from packets, but RTP still can not provide any QoS guarantee.

In this thesis, we present a measurement of traffic, which simulated two types of software gateway(SG) architectures. The two different gateway architecture generate different properties of traffics based on RTP/UDP/IP protocol stack. We will examines their implications for VoIP deployment.

1.1 Characteristics of VoIP data stream

Figure 1.1 shows the VoIP data flow. To transport audio over a packet-switched network, audio samples had to be compressed first by encoder, followed by insertion of sequence number and time stamp into the compressed packet data. The packet data then transports through the Internet, and is received by the decoder enforced with a dynamic buffer. An underlying characteristic of the current Internet is that a packet often experiences *delay*, *delay jitter* and *packet loss*.

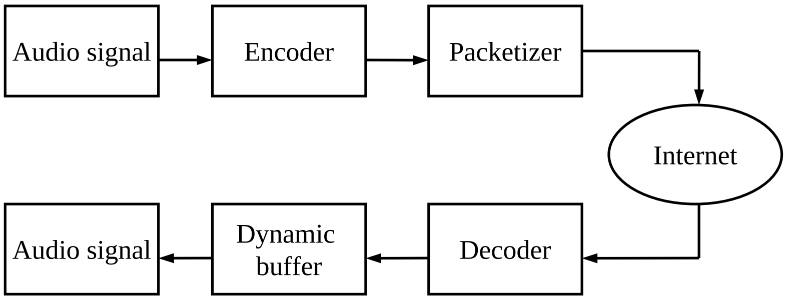


Figure 1.1: VoIP data flow.

The delay consists of codec processing delay and network propagation delay. The network propagation delay can be divided into two parts: a fixed component and a variable component. The fixed component includes propagation delay and transmission delay due to physical layer processing, which is often uncontrollable by the upper layer application. The variable component is caused by processing and buffering at the intermediate network nodes. When the intermediate nodes experience congestion where buffer overflow is occurred, packets may be lost.

The codec delays include a packet compression delay at the source and a receiver decompression and buffer delay. The receiver buffer delay may be flexibly adjusted to compensate for the variable component of the network delay. The speech signal is reconstructed at the receiver by periodically playing out the samples. If a packet fails to arrive before the last sample in the previous packet is played out, a glitch occurs in the speech waveform.

There is a tradeoff between the receiver delay and the fraction of the packets that are lost. If a packet is not available when it is scheduled to be disassembled, either because it was actually lost in the network or because it experienced a long delay, the

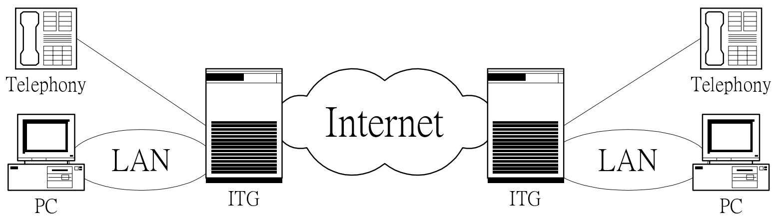


Figure 1.2: Internet telephony architecture.

receiver discards it. Effectively, packets that arrive after a deadline are considered lost.

When a stream of packets traverse the network, each packet may experience different delay; this variation in delay is often called *delay jitter*. Real-time application such as VoIP, however, have to faithfully recreate the original voice stream at the receiver by playing back the data after a fixed delay offset from the original departure time. The packets arriving with shorter delay may have to wait in the receiver's buffer in order for the packets with longer delay to arrive before their playout time. Jitter is a natural result of buffering in packet-switched networks. Whenever packets are buffered, the information about their inter-packet timing is partially lost.

1.2 Internet telephony gateway

As shown in Figure 1.2, we assume in this thesis that two Internet telephony gateways are involved in the realtime voice calls. For the corporate network side (cf. Figure 1.2), PCs and telephones are used interchangeable to receive voice calls. The sender's gateway collects the traffic that is designated for a remote corporate, and (may choose to) uses the Internet to transport that voice traffic to reduce the phone bill. At the distant location, the voice packets are placed by the recipient's gateway, and may be played out by either local PC or telephone. Then the end-to-end communication is formed with voice exchanged in the form of IP packets between the two gateways.

The gateways have two interfaces that link LAN and PBX, and are able to handle many simultaneous calls. Users access multimedia computers or use standard telephones to place and receive a phone call over the Internet via the gateways. When using standard telephones, voice should been digitized by gateway or other equipment. In this thesis, we are assume that the gateways is implemented by pure software. We investigate the performance of two difference software gateway architectures: multiple-connection, where each VoIP call induces an individual connected session over the Internet, and trunking, where multiple VoIP calls share a common Internet session.

1.3 Objectives

Many studies of the network behaviors have appeared in the literature in recent years. For example, Maxemchuk and Lo [3] have compared the voice quality among intrastate, cross country, and international Internet connections. Sanghi and Jain [1] design the experiments to capture roundtrip transit delays over Internet. Andren, Hiding and Veitch [2] presented an investigative tool for the direct study of end-to-end Internet traffic, and found a number of consistent properties. M. Yajnik, S. Moon, J. Kurose, D. Towsley [5] presented analysis of 128 hours of end-to-end unicast and multicast packet loss measurement. Previous research in this area has shown that Internet delay are often in the hundreds of milliseconds, and are usually correlated with packet loss. However, these studies were limited in scope, usually measuring just "unidirectional" traffic behavior, which does not match the VoIP traffic characteristic.

Our measurements are based upon the current best-effort datagram service of the Internet, and uses the UDP protocol. The experiment were made two-way, multiple-users traffic behavior over 200 hours. They can show long-term trends, as well as short-term trends, of the Internet. They also provide numbers that can be used as a reference by VoIP gateway manufacturers.

In order to evaluate the impact of Internet delay on real-time applications, knowledge of unidirectional delay, delay jitter, and packet loss rates are necessary. Accurately determining one-way packet delay from a client host to a server host in the Internet is difficult due to the need of synchronizing the client and server clocks. Thus, we choose to measure the jitter and packet loss in our experiment. Our objective is to measure and understand the jitter and packet loss characteristics of the Internet as they affect its ability to carry interactive voice traffic. These metrics must be available in order to determine the most effective codecs, transmission redundancy rates, and receiver buffer size.

1.4 Organization of the thesis

The rest of the thesis is organized as follows. In chapter 2, we describe the data collection process, i.e., how the measurements of packet delay jitter and loss are obtained. In chapter 3, we present the result of our experiment. In chapter 4, we summarize the observatory characteristics of the measured jitter and packet losses. Chapter 5 concludes the thesis.

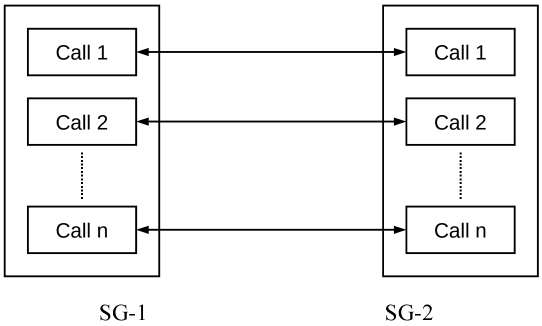
Chapter 2

Experimental setup

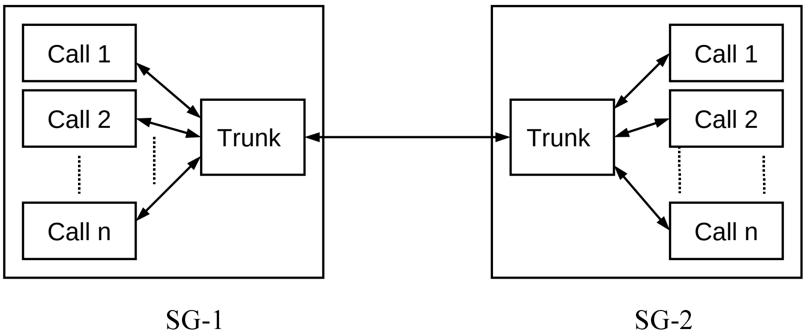
2.1 Simulation model

Our Internet telephony model, such as a corporate network in section 1.2, PC's and telephones in corporate offices are used to place voice call to a distance receiver. A software gateway collects the traffic, and uses the packet-switched network to transport the traffic. At the distant location, the voice packets are received and handled by another software gateway, and delivered to the recipient. We consider two kinds of software gateway (SG) architecture: multiple-connection, where each VoIP call induces an individual connected session over the Internet, and trunking, where multiple VoIP calls share a common Internet session. It is our hope that our experiment results can suggest solutions of practical implementation of software gateway.

Figure 2.1(a) is the model of multiple-connection software gateway (MC-SG) architecture. There are multiple connections between SG-1 and SG-2. When SG-1 gets the voice packet from a call, it will assemble them into RTP packet for each individual call, and transmit it through an individual connection. SG-2 also executes identical procedure, when it delivers voice packets to SG-1. In our experiment, each two-way call is simulated by a process running in the software gateway, which consists of a



(a)



(b)

Figure 2.1: (a) Multiple-connection software gateway (MC-SG) architecture; (b) Trunking software gateway (T-SG) architecture.

sender and a receiver process.

Different from MC-SG, as shown in Figure 2.1(b), before sending out the voice packets, a T-SG will periodically collect the voice packets, and aggregate these packets into a big trunking packet in order to save the overhead due to IP header. There are several criterions that can be used in T-SG. For example, a T-SG can send trunking packets whenever the fixed-size trunking buffer is full. Or a T-SG can send the trunking packet regularly in a fixed period, even if there is still room in the trunking

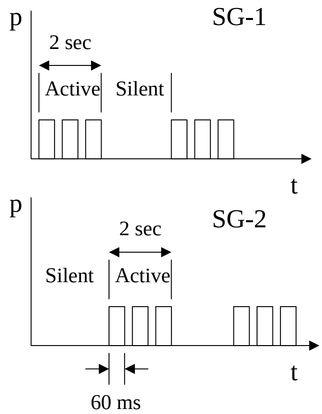


Figure 2.2: Voice source with silence detector.

buffer. In our experiment, we choose to let the T-SG send a trunking packet in a regular time in order to fulfill the real time demand.

2.1.1 Choice of parameters

There are four source parameters in our system, which are *talk-spurt period*, *inter-packet duration*, *packet size* and *number of source*. As expected, different parameters will result in different behaviors of traffic.

The statistic of the talk-spurt period has been studied by many researchers. Through thorough measurements of speech conversations, Brady [7] found that the average active interval for a one party in a speech conversation is about 1.3 seconds. In our simulations, we choose a longer talk-spurt period, 2 seconds, for programming convenience. In other words, each part in a conversation will take turns to speak for a 2-second duration, as shown in Figure 2.2.

The second parameter is the inter-packet duration. From the current standards in VoIP codecs, such as such as G.723.1, G.729, G.729A, the voice is divided into 30-msec

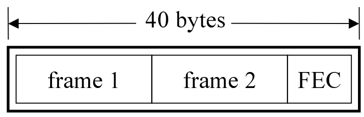


Figure 2.3: Content of a voice packet.

frames. Also, by measuring a Netmeeting conversation, we found that each packet should consist of two 30-msec frames of voice data. Based on these observations, our voice source will create a voice packet in every 60 msec.

The third parameter is the packet size. As mentioned in the previous paragraph, two 30ms frames will be included in a packet. From the current VoIP standards, two 30ms frames are approximately equivalent to 40 bytes of data, as shown in Figure 2.3. Thus, together with the RTP/UDP/IP headers, we let the experimental packet size to be 80 bytes.

Finally, we note that the software gateways are usually used by small corporate for its cheaper price. Hence, the number of users should be small. We therefore assume that the number of active calls are 10.

2.2 Measurement tool

We have developed a UDP-based measurement package. We use multiple-connection mode to simulate the traffic behavior of the MC-SG, and use trunking mode to simulate the T-SG. The tool consists of a client and a server that are run on two different machines. The client and server program consist of a certain number of processes running in parallel. In multiple-connection mode, numbers of transmitters and receivers are running, where each couple of a transmitter and a receiver simulate a two-way

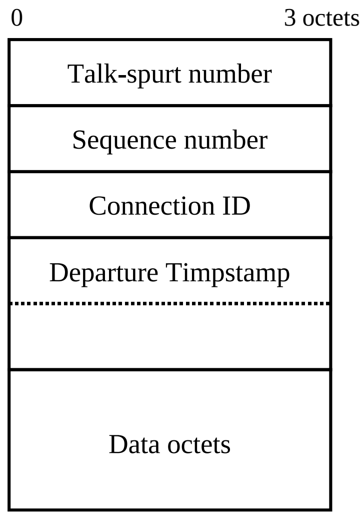


Figure 2.4: Packet format.

conversation. In trunking mode, there are also packet transmitters and receivers in T-SG, where each couple of a transmitter and receiver represents a conversation. Besides, there are a trunk sender and a trunk receiver in T-SG.

The transmitter process uses the host's operating system timer to schedule transmission of a stream of UDP packets with regular inter-departure times. The format of the packet exchanged by gateways is shown in Figure 2.4. The format permits recording of talk-spurt numbers, sequence numbers, connection identifier and departure time stamps. Note that these information can in fact be provided by RTP/RTCP protocol stack. The talk-spurt number is a sequence number of the talk-spurt. The sequence number of packet is used for recording packet loss. A recipient's gateway can use connection identifiers to know which receiver the packets belong to. By recording departure time stamps, we can analyze the delay jitters of traffic.

	route address
1	140.113.13.254
2	CB-gw.NCTU.edu.tw
3	GE-Gw-ATM.NCTU.edu.tw
4	Nctu-Gw1-HcRC.TANet.edu.tw
5	TANet-MOE.edu.tw
6	ags1.ntut.edu.tw

Table 2.1: Route from NCTU to NTUT (May 26 AM 9:00, 2000).

2.3 Simulation environment

In this thesis, we examine two specific connections in details. The first connection is between an Alpha PC located at TNCE, Inc., in Hsin Chu, Taiwan, R.O.C., and an UltraSPARC-II workstation located at Queen's University, Canada. Another connection is between an UltraSPARC-II workstation at National Chiao Tung University, Hsin Chu, and an UltraSPARC-II workstation at National Taipei University of Technology, Taipei. The Alpha PC is running Red Hat Linux release 6.0. Both the UltraSPARC-II workstations at Queen's and at NTUT are running SunOS Release 5.6. The UltraSPARC-II workstation at NCTU is running SunOS release 5.5.1.

By means of a Unix command–ping, we can know the paths that the voice packets are routed. Tables 2.3 and 2.4 present the route path between TNCE, Inc. and Queen's University. From the two tables, we can know the paths from TNCE to Queen's University and from Queen's University to TNCE are not entirely identical. But the backbone from HiNet of ChungHwa Telcom to UUNET of MCI is the same. Tables 2.1 and 2.2 are the route path between NCTU and NTUT. The backbone is TANet and the route path is the same.

	route address
1	140.124.40.254
2	140.124.254.254
3	mgs.ntut.edu.tw
4	TANet-NCTU.edu.tw
5	HcRC-Gw1-NCTU.TANet.edu.tw
6	ATM-Gw-GE.NCTU.edu.tw

Table 2.2: Route from NTUT to NCTU (May 26 AM 9:00, 2000).

	route address
1	210.59.188.2
2	202.39.252.122
3	h30.s222.ts.hinet.net
4	168.95.202.102
5	168.95.2.3
6	210.65.161.89
7	Serial3-1-0.GW2.PAO1.ALTER.NET
8	119.ATM3-0.XR1.PAO1.ALTER.NET
9	189.at-1-0-0.TR1.SAC1.ALTER.NET
10	127.ATM7-0.TR1.TOR2.ALTER.NET
11	199.ATM6-0.XR1.TOR2.ALTER.NET
12	195.ATM7-0.GW1.TOR2.ALTER.NET
13	exterior-border-gige-if.onet.on.ca
14	kin-rt2-exterior-if.onet.on.ca
15	queens-kin-rt2-if.onet.on.ca
16	FLE-BB-RSw.gw.QueensU.CA

Table 2.3: Route from TNCE to Queen's University (May 7 AM 8:00, 2000).

	route address
1	jeffery.gw.queensu.ca
2	Border.gw.QueensU.CA
3	kin-rt2-queens-if.onet.on.ca
4	exterior-kin-rt2-if.onet.on.ca
5	border-exterior-gige-if.onet.on.ca
6	103.ATM2-0.XR1.TOR2.ALTER.NET
7	295.ATM3-0.TR1.TOR2.ALTER.NET
8	137.at-5-1-0.TR1.SAC1.ALTER.NET
9	197.ATM6-0.XR1.PAO1.ALTER.NET
10	189.ATM10-0-0.GW2.PAO1.ALTER.NET
11	chunghwa-gw.customer.alter.net
12	210.65.161.90
13	168.95.2.6
14	Hsin Chu-R01.BR.HiNet.Net
15	h19.s222.ts.hinet.net
16	202.39.252.121

Table 2.4: Route from Queen's University to TNCE (May 7 AM 8:00, 2000)

Chapter 3

Experimental Results

3.1 Definitions of packet loss and jitter

Our study of end-to-end behavior in the Internet is based on 9 days of traces. These traces are gathered by sending out packet probes along end-to-end connections at periodic intervals and by recording the sequence numbers of the probe packet that arrives successfully at the receiver. There are two situations that could cause the packets loss. If there are packets corresponding to some sequence numbers missing, they are certainly lost. In a receiver, when the received buffer is overflow, the following packets will be discard, and is treated as lost.

The recorded data of our experiments consists of $D(k)$ and $L(k)$. $D(k)$ is the arrival time of kth packet. $L(k) = 1$ if kth packet is lost, otherwise $L(k) = 0$. Jitter is defined as:

$$J(k) = J(k\frac{D(k) - D(k-n)}{n}$$

and

$$J(j) = \text{not-applicable}, \ \text{for } j = k - n + 1, \cdots, k - 1,$$

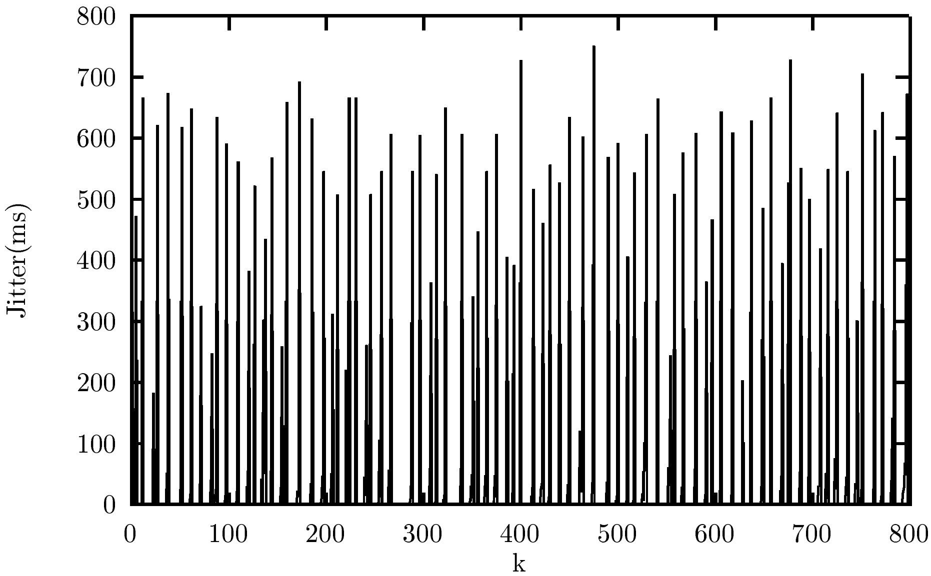


Figure 3.1: Jitter illustration of MC-SG connection starting at 14:00pm of April 24, 2000, Taiwan time, at TNCE Inc.

where n the number of consecutive lost packets (i.e, packets $k-n+1, \ldots, k-1$ are lost), and a not-applicable value in $J(\cdot)$ means that the quantity will not be included in our final statistics.

Figure 3.1 illustrates an example of measured jitters as a function of k in the range $0 \leq k \leq 800$.

We summarize our overall measurements in Table 3.1.

3.2 Long term behaviors

In this section, we observe the long-term behavior of jitters and losses. The night-day collected data is averaged for a duration of a day, as shown in Figure 3.2. We can

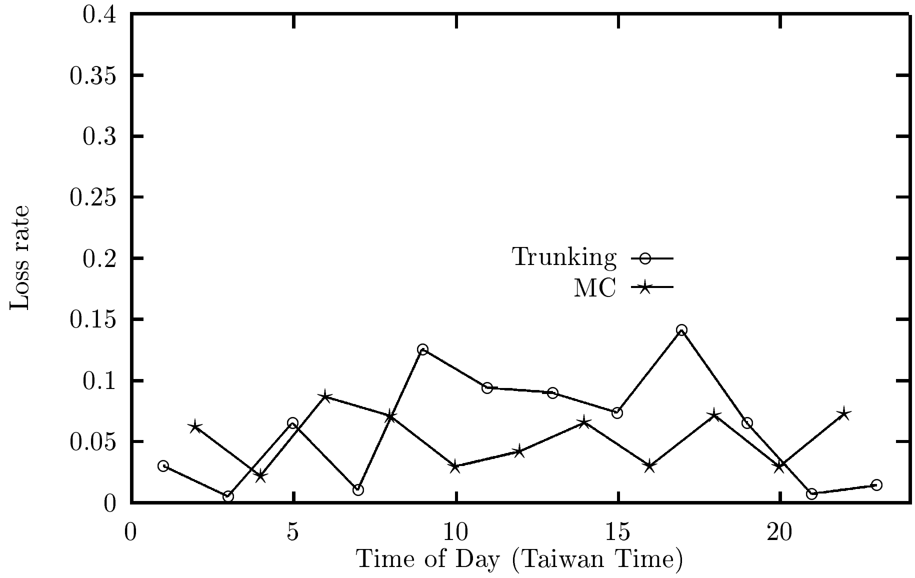
Connection between TNCE and Queen's U	
T-SG	116 hours
MC-SG	111 hours
Connection between NCTU and NTUT	
T-SG	12 hours
MC-SG	11 hours

Table 3.1: Summary of overall measurement time in this study.

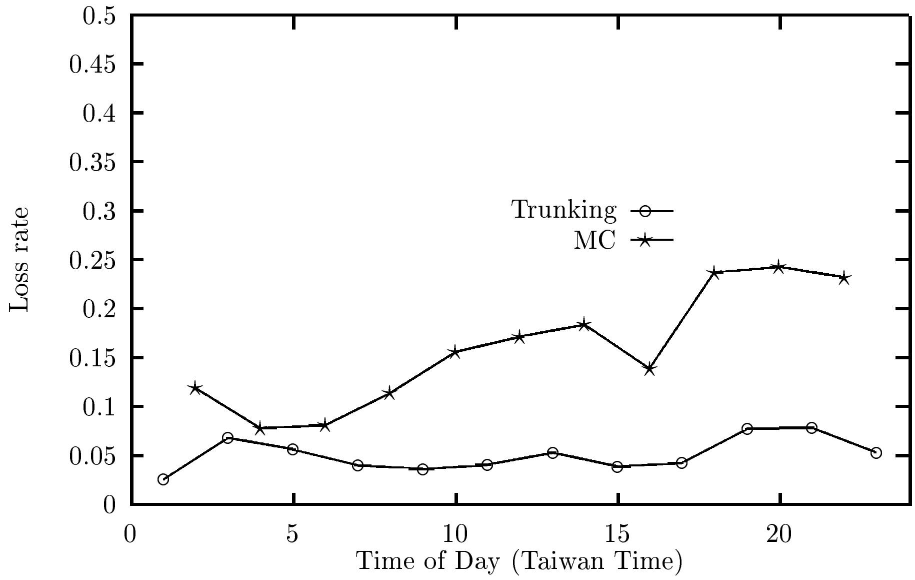
see from Figure 3.2 that the software gateway has different performance in different path. Figure 3.2(a) shows that the average packet loss rate from Queen's University to TNCE is better for MC-SG in most of the time. However, in the direction from TNCE to Queen's University, the average packet loss rate for MC-SG performs worse than T-SG. We conclude that the long term behavior in packet loss is path-dependent, and none of the SG architectures is apparently superior.

Figure 3.3 presents the average variance (averaged for the time of day over 9-day period) of the jitters in the two route path. These variances indicate the degree of jitter change. From Figure 3.3(a), we observe that the variance of jitters for T-SG is much larger than that of MC-SG. However, in Figure 3.3(b), we observe that the difference of jitter variance between the two architectures is not so large; yet, the jitter variance of T-SG is smaller than that of MC-SG in most of the time.

From Figure 3.2 and Figure 3.3, we conclude that because of lower loss rate and gentle change of jitters, using MC-SG architecture is better than T-SG architecture in the path from Queen's University to TNCE. However, in the path from TNCE to Queen's University, T-SG is a better choice.

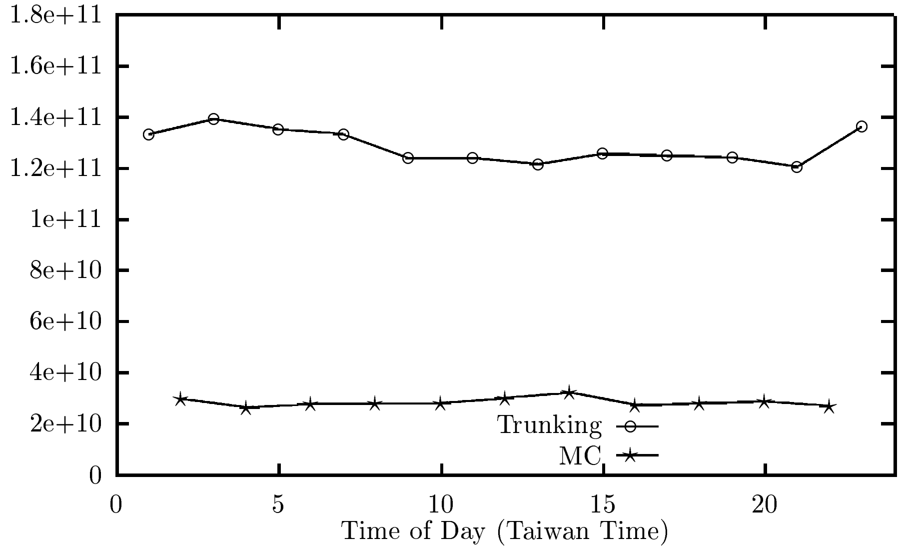


(a)

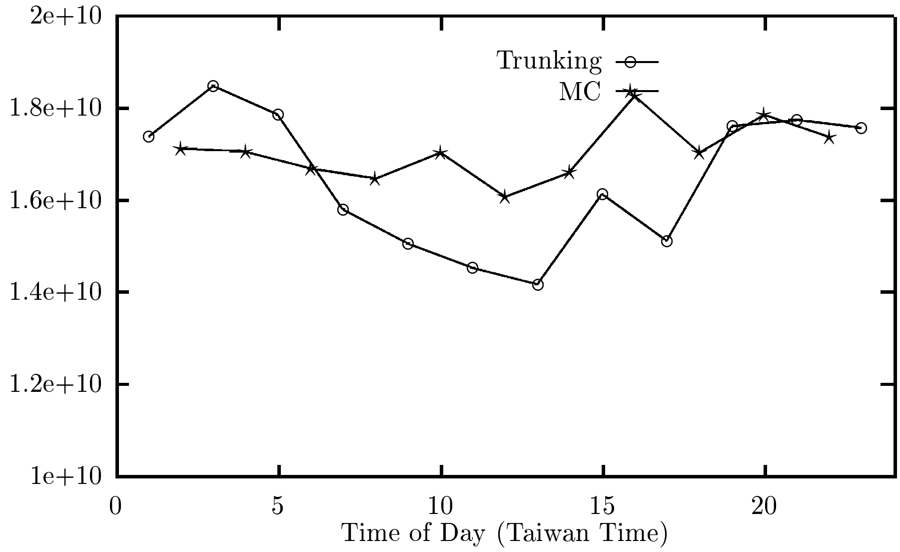


(b)

Figure 3.2: Packet loss averaged over 9 day period as a function of the hour. (a) From Queen's University to TNCE. (b) From TNCE to Queen's University. Here, infinite receiver buffer is assumed; and hence, the loss rate is solely contributed by those non-arrived packets.



(a)

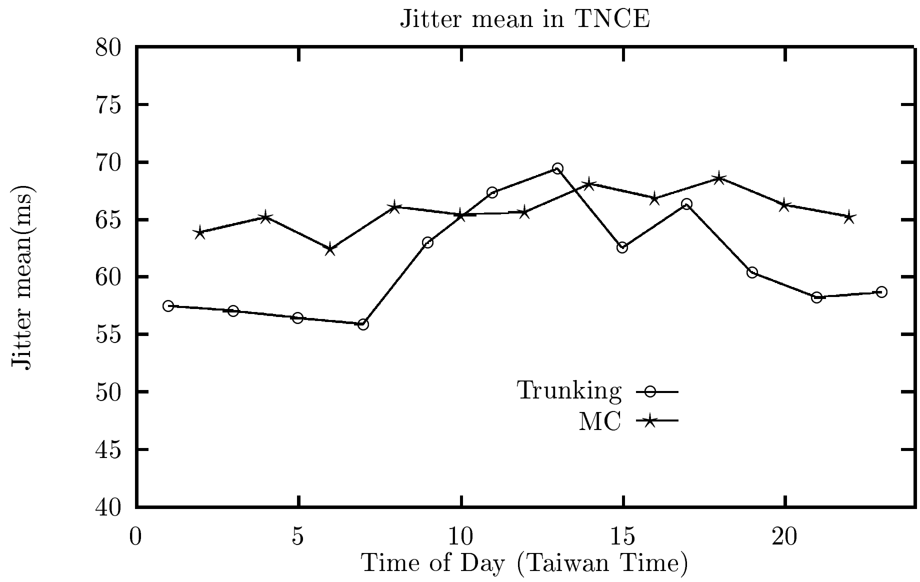


(b)

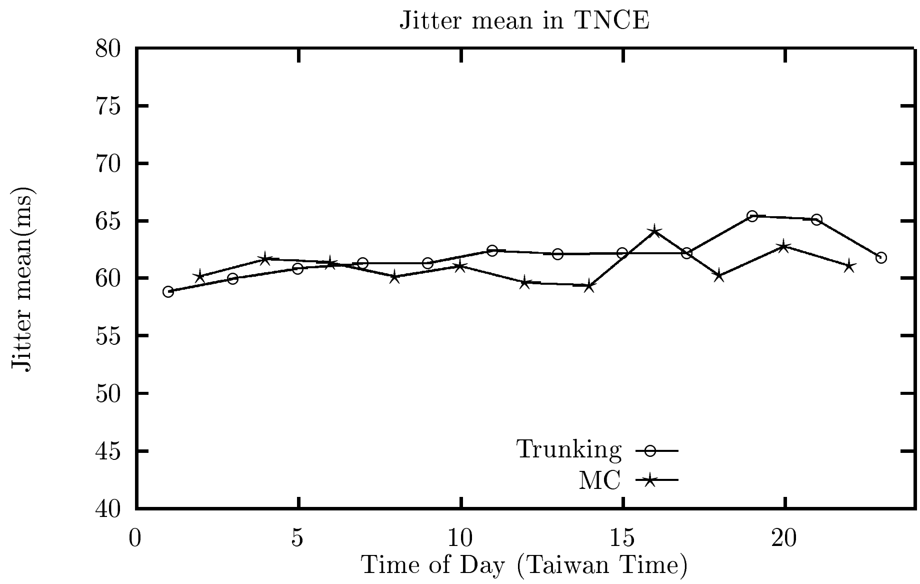
Figure 3.3: Variance of jitters averaged over 9 day period as a function of the hour. (a) From Queen's University to TNCE. (b) From TNCE to Queen's University.

The means of the jitters is shown in Figure 3.4. We can find the means are oscillating between 55 and 65 ms. It is surprised to observe that the long-term jitter means could fall below the inter-departure time, i.e., 60ms. It could be a consequence of the *buffering* of the intermediate network nodes. In other words, when routers in networks receive packets, they may queue these packets, and transmit them all together later according to their scheduling strategy.

In Figure 3.4(a), long-term jitter mean of MC-SG is never below 60 ms, but T-SG does have jitter mean smaller than 60ms for a certain period of day. This means that smaller packet is more unlikely to be queued during the network. In the path from TNCE to Queen's University, jitter mean of MC-SG is lower than T-SG, but the difference is limited.

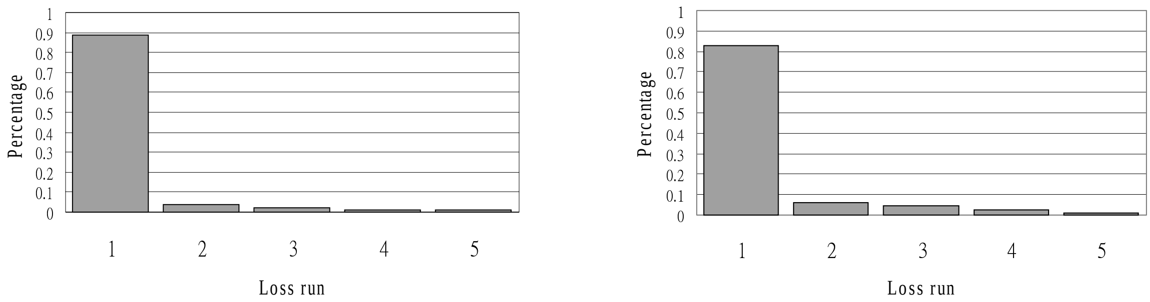


(a)



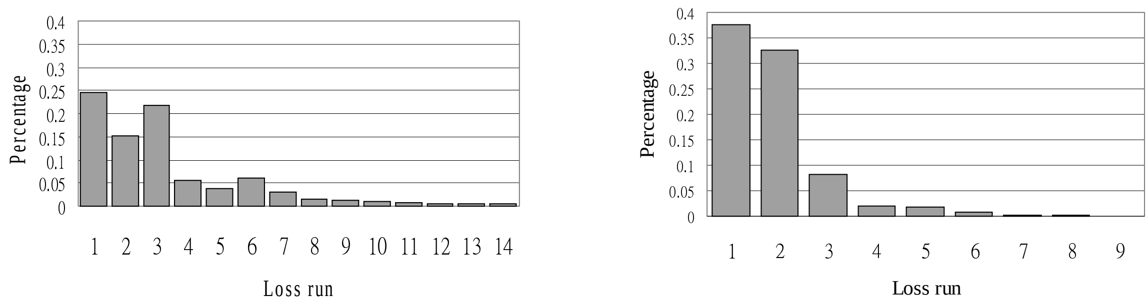
(b)

Figure 3.4: Mean of jitters averaged over 9 day period as a function of the hour. (a) From Queen's University to TNCE. (b) From TNCE to Queen's University.



(a) (b)

Figure 3.5: The loss run distribution of the path from Queen's University to TNCE. (a) Multiple-connection software gateway architecture; (b) Trunking software gateway architecture.



(a) (b)

Figure 3.6: The loss run distribution of the path from TNCE to Queen's University. (a) Multiple-connection software gateway architecture. (b) Trunking software gateway architecture.

We also establish the statistics for *loss run*, which is defined as the number of consecutive packet loss. Figure 3.5 presents the consecutive loss distribution in the path from Queen's University to TNCE. We can see that the loss run is mostly one or two long. Specifically, for both MC-SG and T-SG, over 80% of the loss run is no greater than one. The result coincides with J.Andren's [2] finding [2].

The loss run distribution of path from TNCE to Queen's University is shown in Figure 3.6. We are surprised to know that the loss run distribution of MC-SG, as shown on Figure 3.6(a), spread its weight to those loss runs longer than two. In Figure 3.6(b), most of the loss run under T-SG are still confined within one or two.

From Figures 3.7–3.9, we show that loss rate, jitter mean and jitter variance between NCTU and NTUT, which are measurement results of shorter route that between TNCE and Queen's University. We note from Figure 3.7 that the loss rate of T-SG and MC-SG is about the same. However, we found that the long-term average jitter and jitter variance of T-SG are smaller than those of MC-SG. We also found that the difference in jitter variance between international route (between TNCE and Queen's university) and domestic route (between NCTU and NTUT) is prohibitively small.

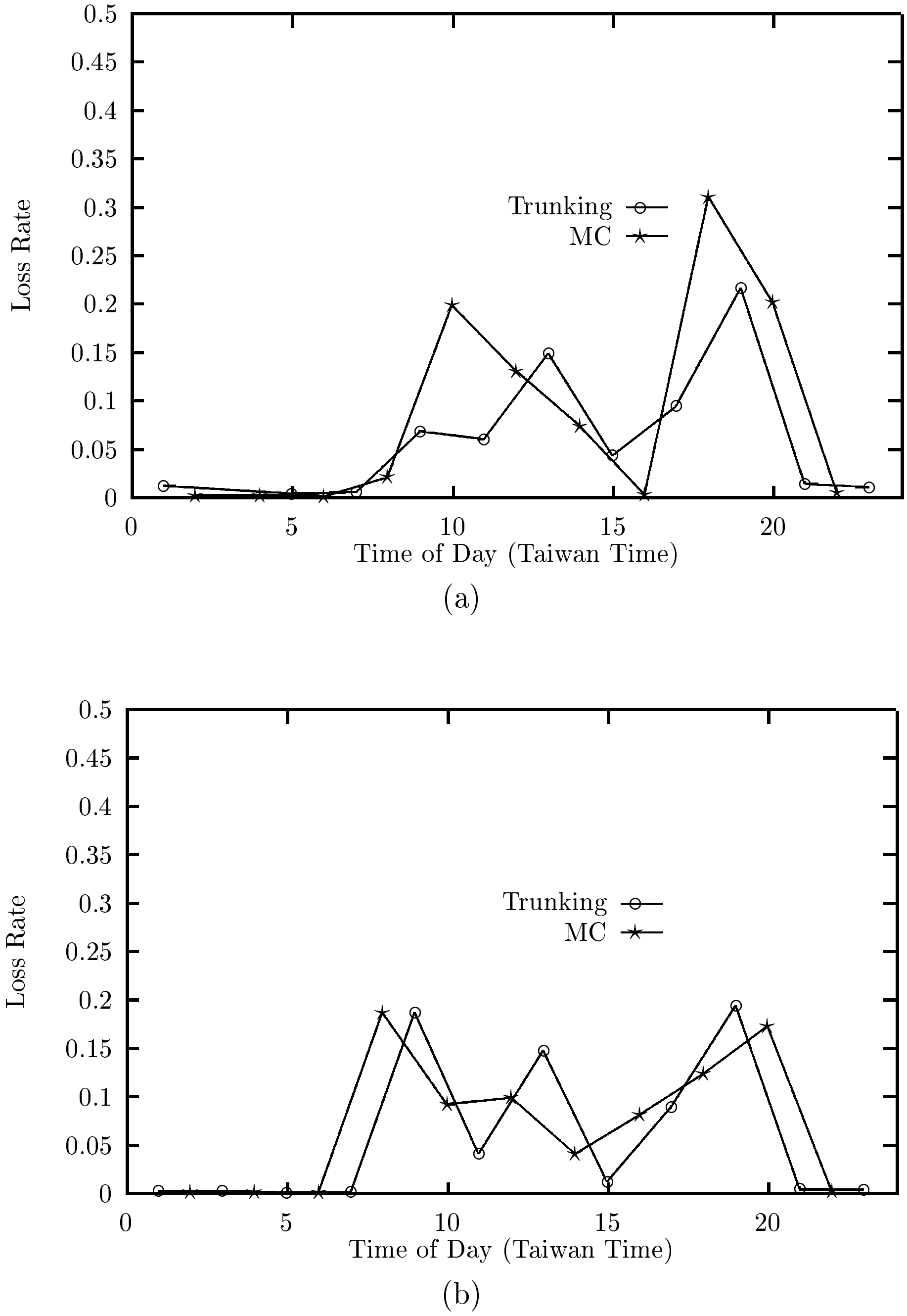
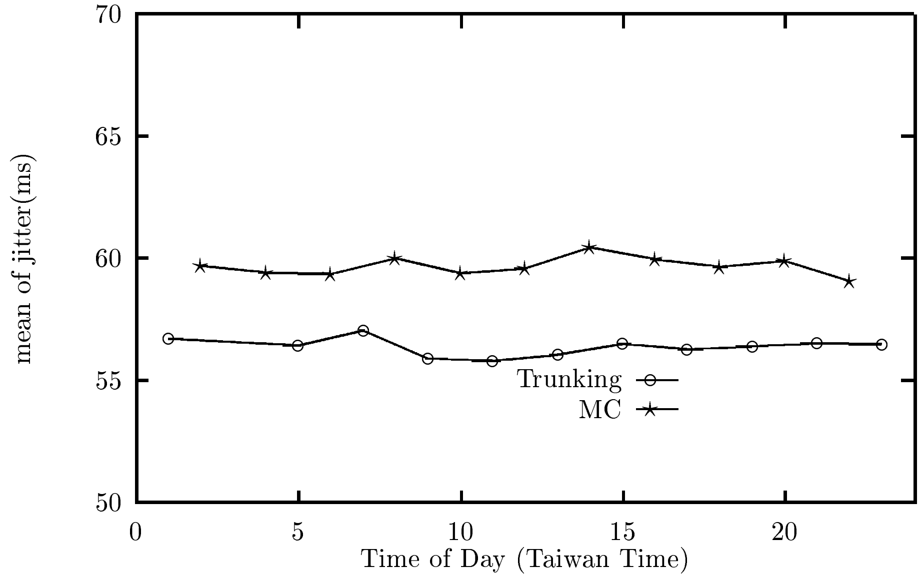
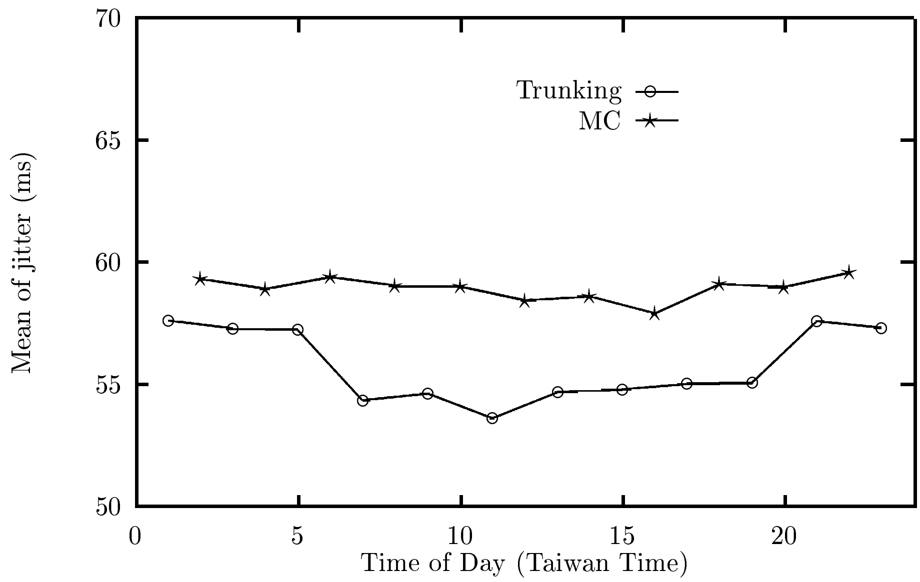


Figure 3.7: Packet loss averaged over a day period as a function of the hour. (a) From NTUT to NCTU. (b) From NCTU to NTUT.

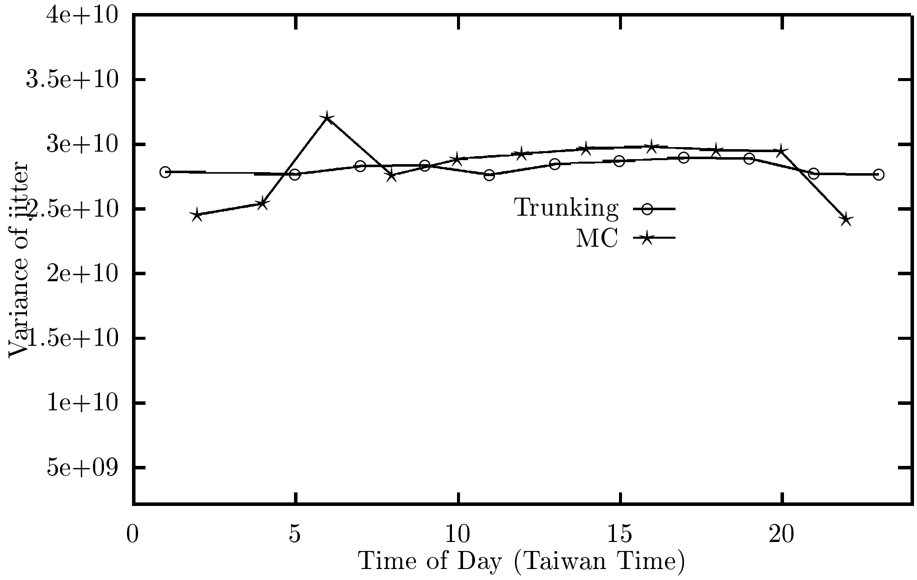


(a)

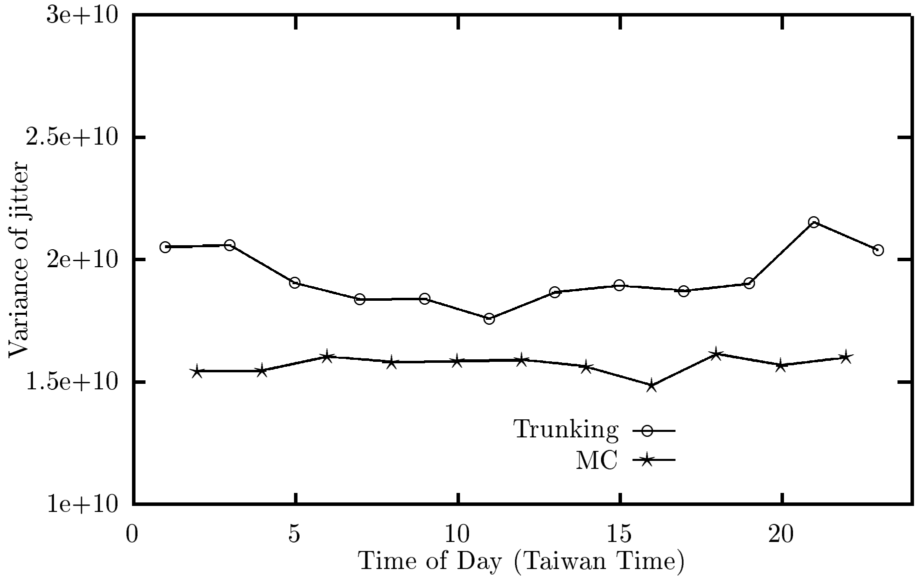


(b)

Figure 3.8: Mean of jitters over a day period as a function of the hour. (a) From NTUT to NCTU. (b) From NCTU to NTUT.



(a)



(b)

Figure 3.9: Variance of jitters over a day period as a function of the hour. (a) From NTUT to NCTU. (b) From NCTU to NTUT.

3.3 Effective performance due to limited buffer and with FEC capability

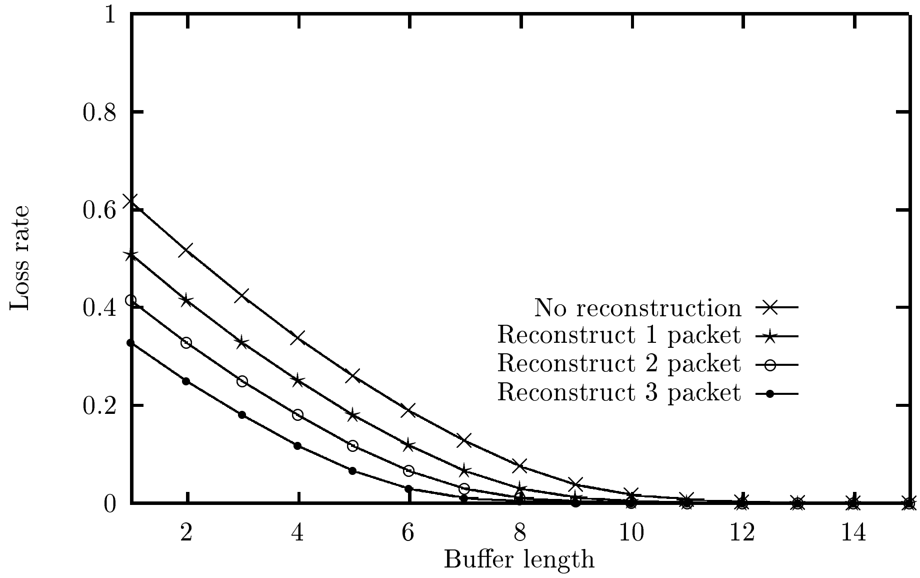
In the previous session, the packet loss is mainly due to network congestion, for which an infinite buffer is assumed at both ends. Nevertheless, if the buffer at receiver side has limited length, then the packet may be discarded at the end terminal when the receiver buffer is full. Although what concerns us is the software IP phone, still one needs to set aside the buffer limit due to real-time consideration. By the way, if some FEC is applied for the packet transmissions such that consecutive one packet loss can be compensated, then the effective packet loss rate will be smaller.

Using the same data used in the previous section, Figures 3.10 and 3.11 show the effective packet loss rate at the receiver due to limited buffer and with different packet-restored capability.

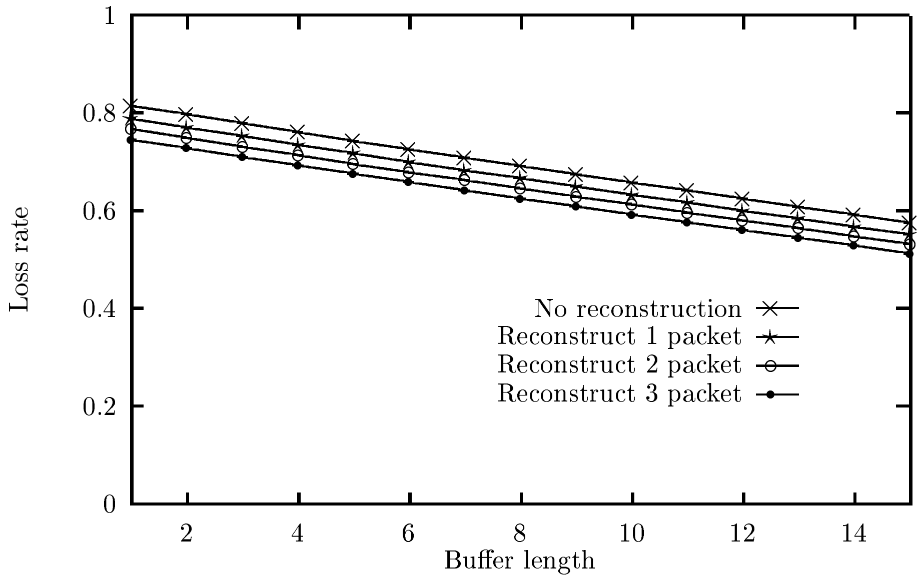
Figure 3.10 depicts the effective packet loss rate at TNCE. By comparing Figures 3.10(a) and 3.10(b), we found that MC-SG improves its packet loss rate faster than T-SG architecture for which the loss rate decreases only linearly. In addition, we observe from Figure 3.10 that a large gap is induced from 0 restoration to 1 restoration (meaning that the FEC is capable of reconstruct one consecutive loss packet).

Figure 3.11 depicts the effective packet loss rate at Queen's university. When the buffer length increases, the effective loss rates of both MC-SG and T-SG decrease exponentially. Both Figures 3.10 and 3.11 suggest that restoring only single lost packet may be the most cost-effective choice, since capability with restoring of more than one loss packet may induce more overhead in packet content.

We also compare the effective loss rate of the two SG architectures at TNCE and at Queen's University. In Figure 3.12(a), the effective loss rate of T-SG not only decreases slower, but also is always smaller than MC-SG. However, in Figure 3.12(b), we found that the effective loss rate of T-SG falls below that of MC-SG when buffer length is short, but gradually becomes larger when buffer length grows beyond 3.



(a)



(b)

Figure 3.10: The effective loss rate with respect to 0, 1, 2 and 3 packet restoration capability as a function of the buffer length at TNCE. (a) Multiple-connection architecture at 1:00am, May 1, 2000. (b) Trunking architecture at 2:00am, May 1, 2000.

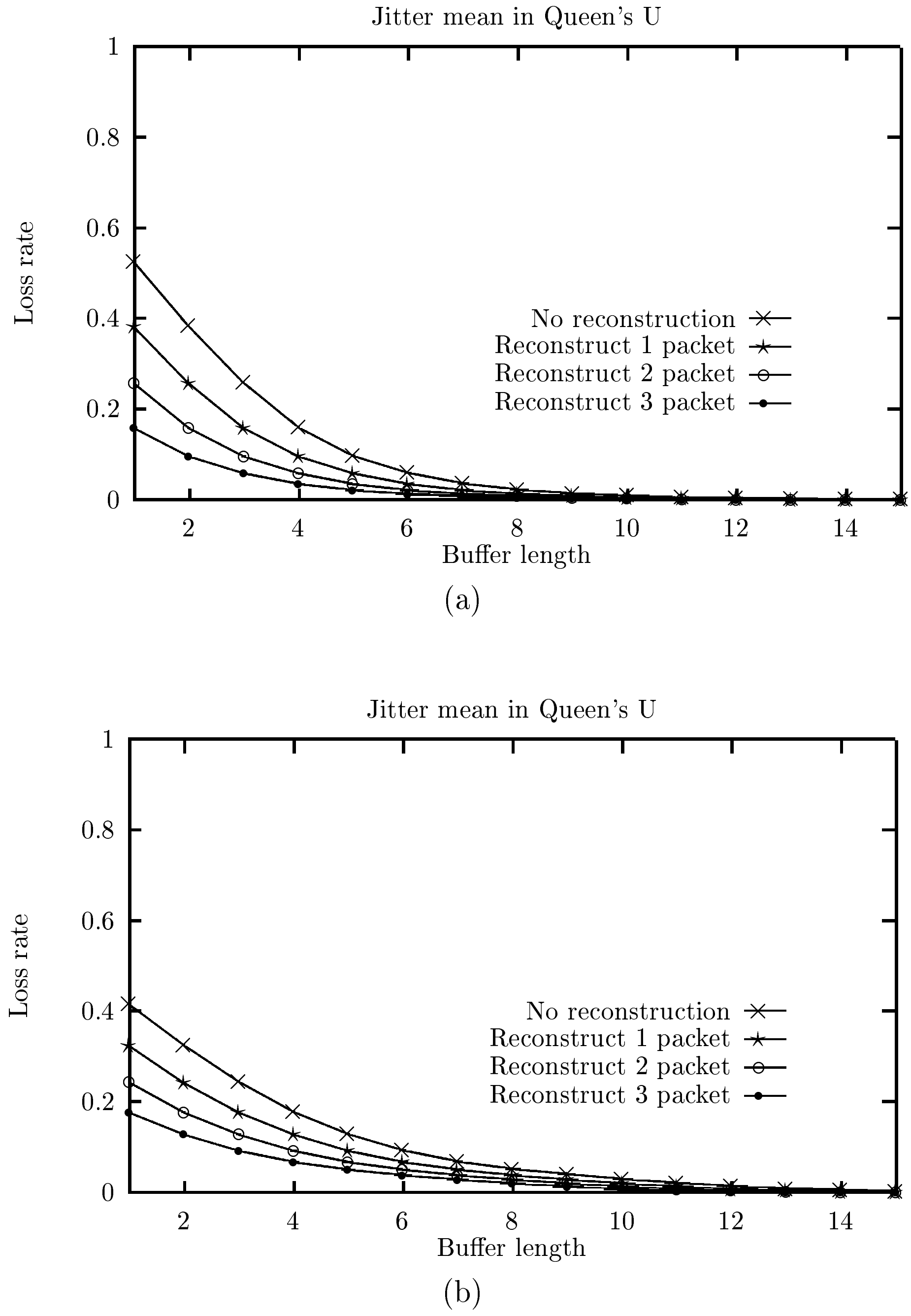
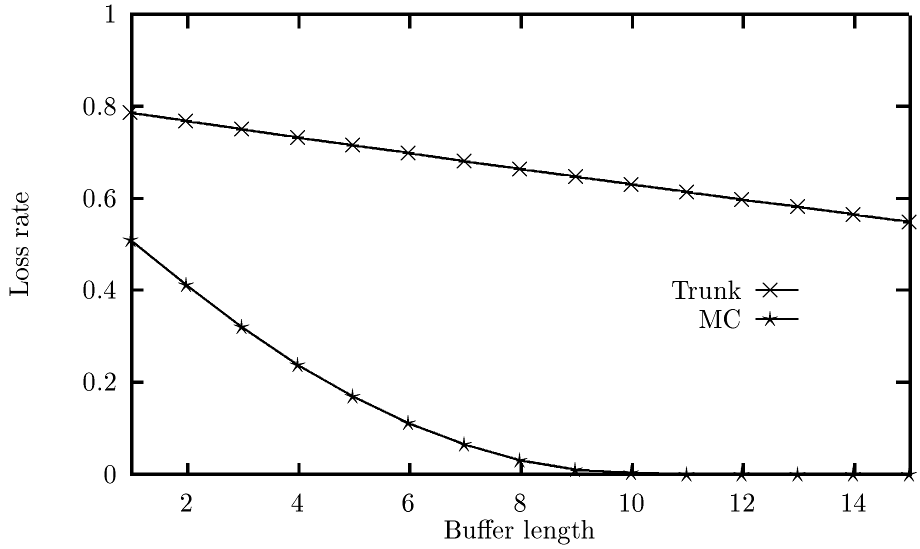
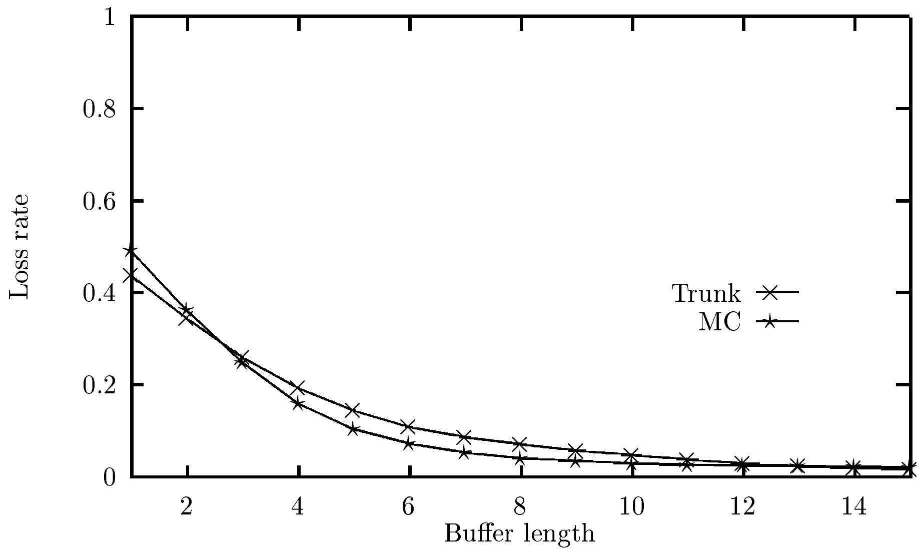


Figure 3.11: The effective loss rate with respect to 0, 1, 2 and 3 packet restoration capability as a function of the buffer length at Queen's university. (a) Multiple-connection architecture at 14:00am, April 28, 2000. (b) Trunking architecture at 15:00am, April 28, 2000.



(a)



(b)

Figure 3.12: Effective loss rate of T-SG and MC-SG. (a) Multiple-connection architecture at 16:00am and trunking architecture at 17:00am, May 1, 2000 at TNCE. (b) Multiple-connection architecture at 16:00am and trunking architecture at 17:00am, May 7, 2000 at Queen's University.

Chapter 4

Observations and Discussions

In this chapter, we present some preliminary observations based on the results of our experiments.

4.1 Suggestions for SG implementation

1. From the discussions in the earlier chapters, the measurement results are dependent on the routing path. Neither T-SG nor MC-SG is conclusively superior.

2. The international connection is more serious in its asymmetry, i.e., the degree of the smoothness (in jitter) of the VoIP traffic flow is quite different for both directions. As expected, the asymmetry is less serious for domestic connection. Surprisingly, it seems that the hope count does not have much influence on the packet loss and variance of jitters.

3. Since no conclusive result can be made on the superiority of either MC-SG or T-SG, we suggests that the operator of a SG should monitor the statistics of traffic flow characteristic periodically, and make necessarily switch when needed, in order to have a better performance.

4. In this experiment, a trunking packet may be larger than the largest router unit, i.e., 576 bytes. In that case, a trucking packet may be divided into several fragments; thereby, larger jitter variance is introduced due to packet re-assembling. Hence, we suggest to limit the size of the trunking packet to 576 bytes so that it will not be segmented during any intermediate network nodes.

4.2 Suggestions for codec implementation

1. At the receiver end, packet discard due to buffer overflow is much worse than packet loss due to networks. However, a larger buffer may induce longer buffer delay which is unexpected for real-time VoIP applications. Therefore, a good compensation is to involve the forward-packet-loss-reconstruction technique so that a proper compromise between buffer size and buffer delay can be reached.

Chapter 5

Conclusions and future work

5.1 Concluding remarks

In this thesis, we have presented results from a series of experiments in which end-to-end behavior of two different paths was measured by recording the arrived time and losses. The probe packets were sent every 60 ms to emulate the real VoIP traffic pattern. Only an initial analysis of these measurements is presented here.

A specific characteristic of our experiment is that what we do here is a two-way measurement, instead of conventionally one-way measurement. Through the experiments, we confirm the serious asymmetry problems for two-way real-time applications over networks. This is specially harmful to a conversation-type application where a bad quality only in one path can destroy the whole conversation. In addition, we observe that jitters do have essential impact on audio.

5.2 Future work

We have presented some initial thoughts on the observed behaviors. The data from these experiments may be used to carry out other forms of analysis. This work is on-going, and a more complete analysis could be done in the future.

Bibliography

[1] D. Sanghi, A. K. Agrawala and B. Jain, " Experimental assessment of end-to-end behavior on Internet," *Proc. IEEE Infocom '93*, San Fransisco, CA, pp. 867-874, March 1993.

[2] J. Andren, M. Hilding and D. Veitch, " Understanding end-to-end internet traffic dynamics," *The Bridge to Global Integration. IEEE*, pp. 1118-1122, vol. 2, 1998.

[3] N. F. Maxemchuk and S. Lo, " Measurement and interpretation of voice traffic on the Internet," *ICC '97* Montreal, Towards the Knowledge Millennium, 1997.

[4] J. C. Bolot, " End-to-end packet delay and loss behavior in the Internet," *SigComm'93*, pp. 289-298, 1993.

[5] M. Yajnik, S. Moon, J. Kurose and D. Towsley, " Measurement and modelling of the temporal dependence," *Conference Record of the International Conference on Communications(ICC)*, Montreal, Canada, June 1997.

[6] T. J. Kostas, M. S. Borella, I. Sidhu, G. H. Schuster, J. Grabiec and J. Mahler, " Real-time voice over packet-switched networks," *SigComm'93*, pp. 289-298, 1993.

[7] P. T. Brady, " A Technique for investigating On-Off patterns of speech," *BSTJ*, vol. XLIV, no. 1, Jan. 1965, pp. 1-22.