Experimental Assessment of VoIP individual and trunking traffics

Presenter: Chia-Lung Wu (吳佳龍)
Directed by Prof. Po-Ning Chen
Department of Communication Engineering
National Chiao-Tung University

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Outline

- Overview
- Experimental setup
- Experiment results
- Observations and discussions
- Conclusions and future work

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Overview

- During the past several years, there has been a significant increase in interest in the use of packetized audio over wide-area, packet-switched networks.
- The interest is induced by the possible availability of low-cost, toll-quality audio that can be supported by today's Internet.
- There are three issues in today's Internet telephony,
  - Integration of packet voice and telecommunication
  - Codec should be adapted to get enough voice quality on unreliable transmission
  - Supporting acceptable QoS guarantee on the best-effort networks

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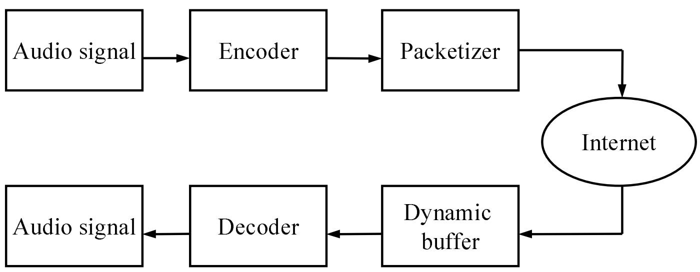
Overview

- The characteristic of the current Internet is that a packet often experiences *delay*, *delay jitter* and *packet loss*.
- VoIP data flow



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Overview

- Delay
  - Propagation delay
  - Codec processing delay
    - Compression delay at source
    - Receiver decompression and buffer delay at the receiver
- Jitter
  - The variation in delay is often called delay jitter (or jitter).
  - The packets arriving with shorter delay have to wait at the receiver's buffer.
  - Jitter is a natural result of buffering in packet-switched networks. Whenever packets are buffered, the info. about their inter-packet timing is partially lost.

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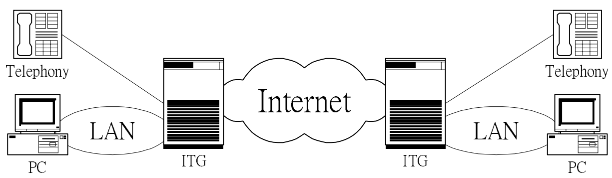
Overview

- In this thesis we investigate the performance of two difference *software gateway(SG) architectures*.
- Our objective is to measure and understand the packet loss and delay characteristics of the Internet as they affect its ability to carry interactive voice traffic.



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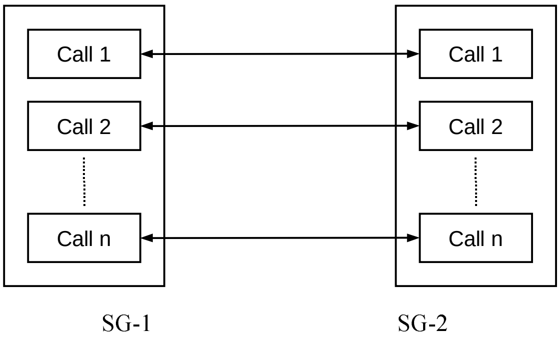
Experimental Setup

Simulation model

- Multiple-connection software gateway (MC-SG)architecture.



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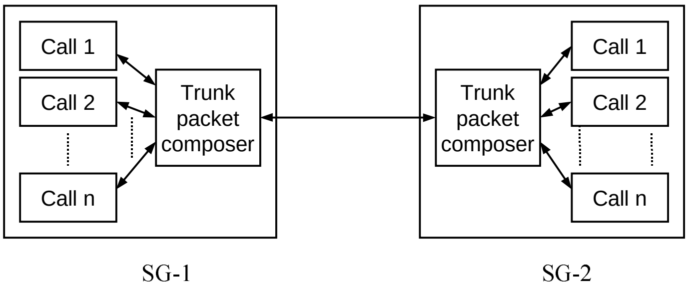
Experimental Setup

- Trunking software gateway(T-SG) architecture.



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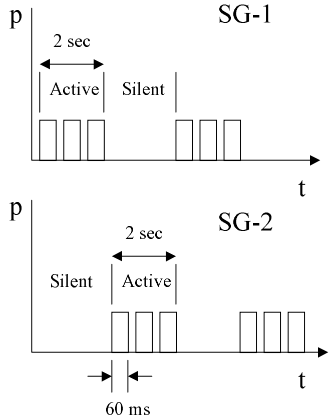
Experimental Setup

Choice of parameters

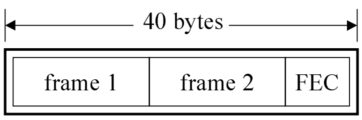
- **Talk-spurt period:**
  - Voice source is with silence detection capability.
  - Talk-spurt period is 2 seconds.
    - Brady, 1965, in measurements of speech conversations, found that the average active interval in speech is about 1.3 seconds.
  - Each party takes turns to talk.



- **Interpacket delay:**
  - Microsoft Netmeeting, 2 frames/packet.
  - Each frame is about 30 ms in length, such as G.723.1.
  - Thus, Our voice source will create a voice packet in every 60 ms.

Experimental Setup

- **Packet size:**
  - Packet content consists of two frames and media-specific FEC.
  - The packet is about of the size 40 bytes. Thus, together with the RTP/UDP/IP headers, we let experimental packet size to be 80 bytes.
- **Number of sources:**
  - The gateways are usually used by small corporate
  - Assume that the number of active calls are 10



Experimental Setup

Simulation environment

- We examine two specific connections in details.
- The connection between TNCE and Queen's University
  - UltraSPARC-II workstation running SunOS 5.7 at Queen's University, Canada.
  - Alpha PC running Red Hat Linux release 6.0 at TNCE, Hsinchu, R.O.C.(Hinet)
- The connection between NCTU and NTUT
  - UltraSPARC-II workstation running SunOS 5.5.1 at NCTU, Hsin Chu.
  - UltraSPARC-II workstation running SunOS 5.7 at NTUT, Taipei.

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Experimental Setup

Route between TNCE and Queen's University

From Queen's University to TNCE	From TNCE to Queen's University
1 jeffery.gw.queensu.ca	1 210.59.188.2
2 Border.gw.QueensU.CA	2 202.39.252.122
3 kin-rt2-queens-if.onet.on.ca	3 h30.s222.ts.hinet.net
4 exterior-kin-rt2-if.onet.on.ca	4 168.95.202.102
5 border-exterior-gige-if.onet.on.ca	5 168.95.2.3
6 103.ATM2-0.XR1.TOR2.ALTER.NET	6 210.65.161.89
7 295.ATM3-0.TR1.TOR2.ALTER.NET	7 Serial3-1-0.GW2.PAO1.ALTER.NET
8 137.at-5-1-0.TR1.SAC1.ALTER.NET	8 119.ATM3-0.XR1.PAO1.ALTER.NET
9 197.ATM6-0.XR1.PAO1.ALTER.NET	9 189.at-1-0-0.TR1.SAC1.ALTER.NET
10 189.ATM10-0-0.GW2.PAO1.ALTER.NET	10 127.ATM7-0.TR1.TOR2.ALTER.NET
11 chunghwa-gw.customer.alter.net	11 199.ATM6-0.XR1.TOR2.ALTER.NET
12 210.65.161.90	12 195.ATM7-0.GW1.TOR2.ALTER.NET
13 168.95.2.6	13 exterior-border-gige-if.onet.on.ca
14 HsinChu-R01.BR.HiNet.Net	14 kin-rt2-exterior-if.onet.on.ca
15 h19.s222.ts.hinet.net	15 queens-kin-rt2-if.onet.on.ca
16 202.39.252.121	16 FLE-BB-RSw.gw.QueensU.CA

From Queen's University to TNCE — From TNCE to Queen's University

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Experimental Setup

Route between NCTU and NTUT

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	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
From NCTU to NTUT	From NTUT to NCTU

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Experiment results

- **Packet loss:**
  - If there are packets corresponding to some sequence numbers missing, they are certainly lost.
  - At a receiver, when the received buffer is overflow, the following packets will be discarded, and is treated as lost.
- **Jitter:**
  - *D(k)* is the arrival time of *k*th packet.
  - Jitter is defined as:

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

where n is the number of consecutive loss packets.

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Experiment results

- Long term behaviors
- Effective performance due to limited buffer, as well as with FEC capability

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

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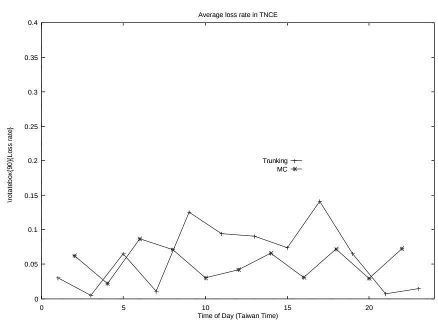
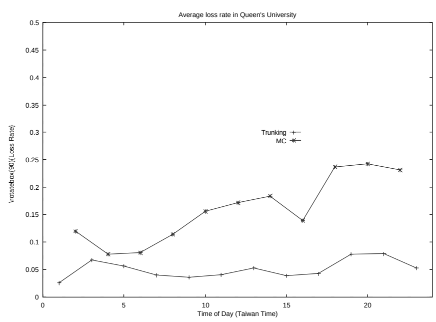
Experiment results

- Packet loss averaged over 9 day period as a function of hour(a) From Queen's University to TNCE; (b) From TNCE to Queen's University



(a)



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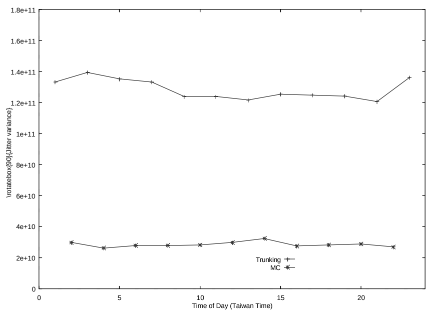
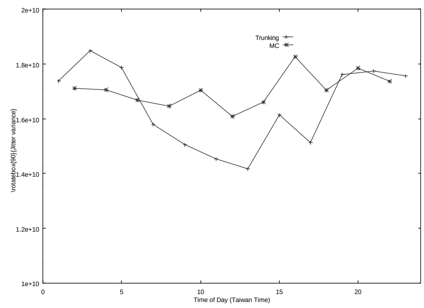
Experiment results

- Variance of jitters averaged over 9 day period as a function of hour;(a) From Queen's University to TNCE; (b) From TNCE to Queen's University



(a)



(b)

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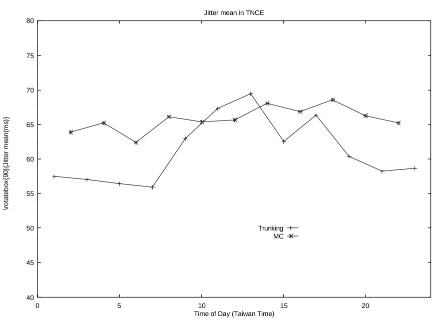
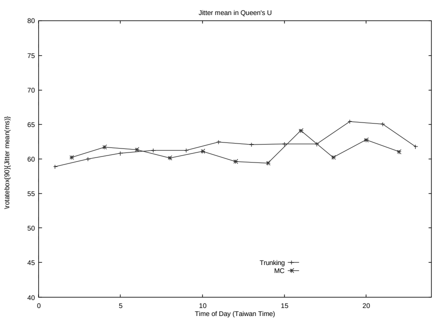
Experiment results

- Mean of jitters averaged over 9 day period as a function of hour;(a) From Queen's University to TNCE; (b) From TNCE to Queen's University



(a)



(b)

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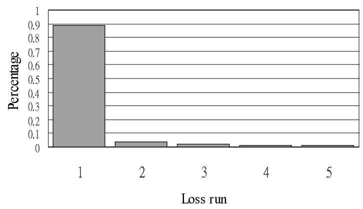
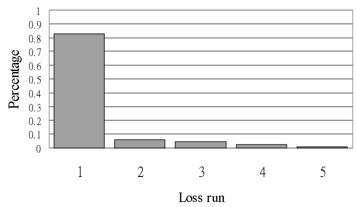
Experiment results

- 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)

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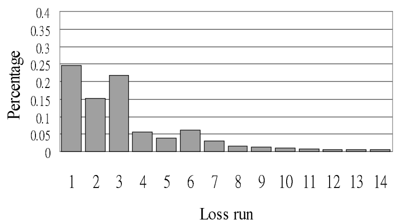
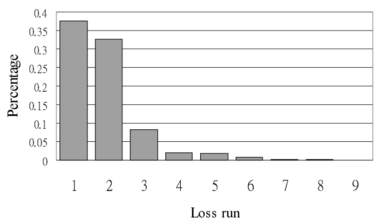
Experiment results

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



(a)



(b)

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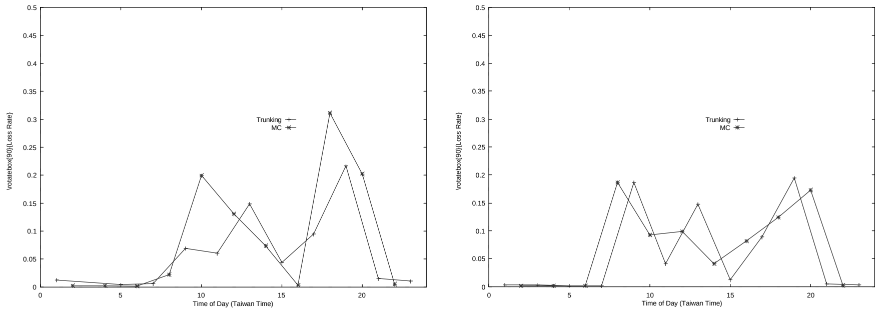
Experiment results

- Packet loss averaged over a day period as a function of hour;(a) From NTUT to NCTU; (b) From NCTU to NTUT



(a) (b)

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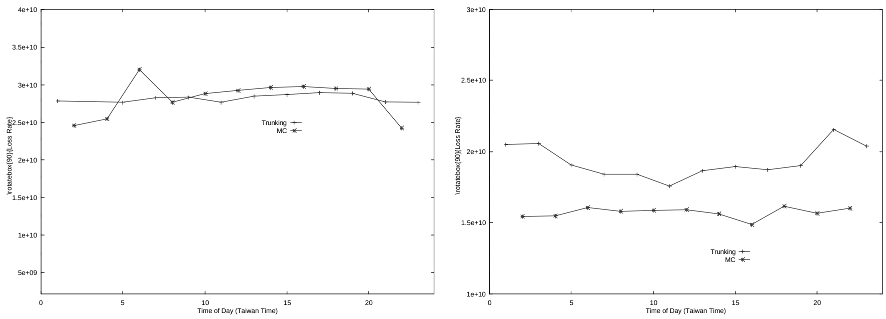
Experiment results

- Variance of jitters over a day period as a function of hour;(a) From NTUT to NCTU; (b) From NCTU to NTUT



(a) (b)

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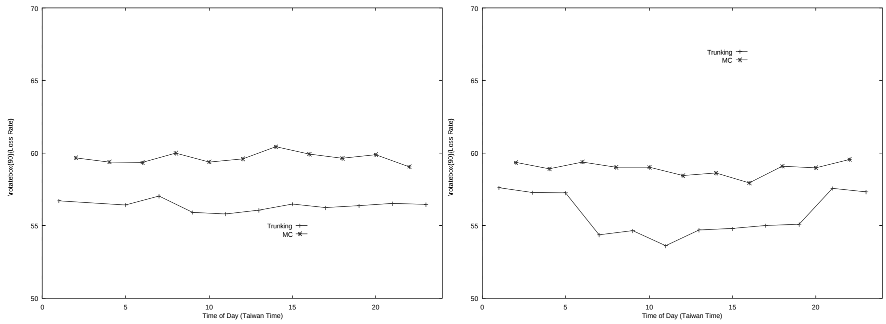
Experiment results

- Mean of jitters over a day period as a function of hour;(a) From NTUT to NCTU; (b) From NCTU to NTUT



(a) (b)

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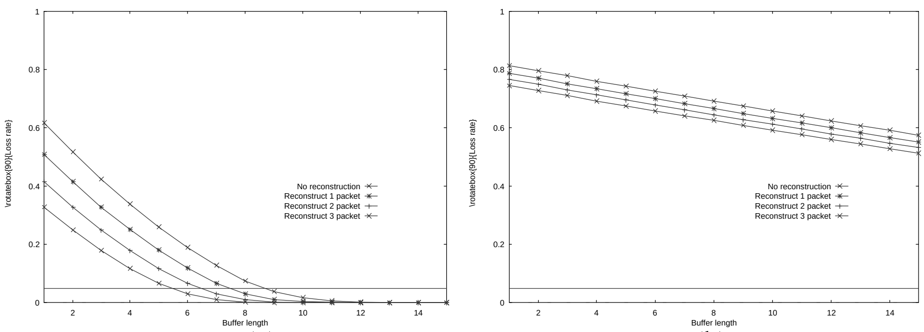
Experiment results

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



(a) (b)

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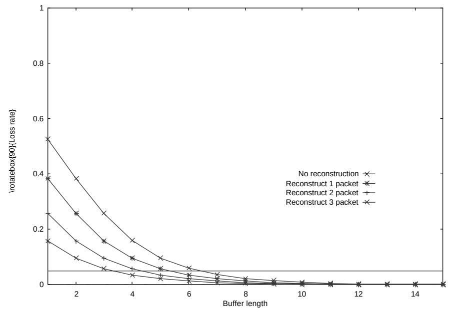
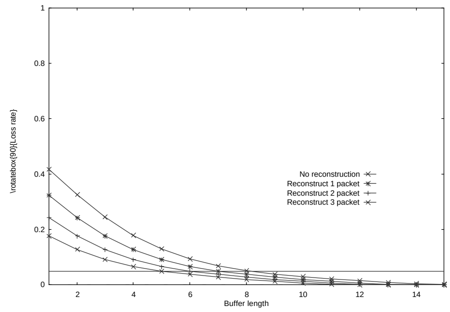
Experiment results

- The effective loss rate with respect to 0, 1, 2 and 3 packet restoration capability as a function of 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)

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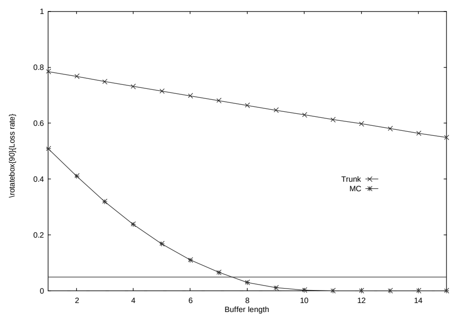
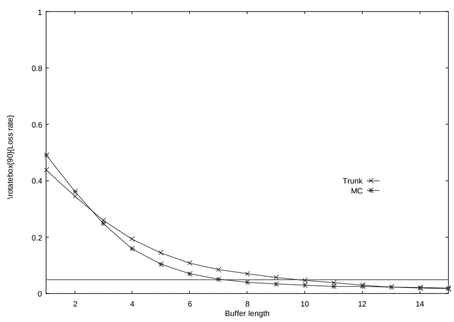
Experiment results

- 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.



(a)



(b)

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Observations and discussions

- The measurement results are dependent on the routing path. Neither T-SG nor MC-SG is conclusively superior.
- The international connection is more serious in its asymmetry. Surprisingly, it seems that the hop count does not have much influence on the packet loss and variance of jitters.
- Since no conclusive result can be made on the superiority of either MC-SG or T-SG, we suggest that the operator of a SG should monitor the statistics of traffic flow periodically, and make necessary switch when needed, in order to have a better performance.

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Observations and discussions

- A trunking packet may be large than 576 bytes. In that case, a trunking packet may be divided into several fragments (by routers); thereby, larger jitter variance is introduced due to packet re-assembling.
- At the receiver end, packet discarded due to buffer overflow is much worse than packet loss due to networks.
- 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.

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Conclusions

- 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 loss.
- 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.
- In addition, we observe that jitters do have essential impact on audio.

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