Impact of Link Failures on VoIP Performance

International Workshop on Network and Operating System Support for Digital Audio and Video (NOSSDAV)

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Sprint ATL

May 2002

Introduction

- Tier-1 ISPs interested in providing Voice-Over-IP (VoIP)
- Need to provide quality
  - Voice quality and availability
- Possible causes of degradation
  - Congestion (*what is this?)*
  - Link failures (*what is this*?)
  - Routing instabilities (*what is this*?)
- Goal of this work is to study the frequency of these events (at Sprint) and assess their impact on VoIP performance

Introduction

- Use passive monitoring for congestion
  - Assess loss plus delay
  - Can't get routing info
- Use active measurement
  - on two well-connected locations
  - Across one IS-IS boundary
- Find Sprint IP backbone ready for toll-quality VoIP
  - Congestion effect is negligible
- Link failures impact availability
  - Cause routing instability for 10s of minutes

Outline

- Introduction (done)
- Related Work (next)
- Measurements
- Voice Call Rating
- Results
- Conclusion

Related Work

- Lots of work on delay and loss characteristics
  - Mostly focus on delay
- But delay and loss alone not sufficient for perceptual quality (PQ)
- Work that develops E-model (Cole et al.) to map network characteristics for voice to PQ
- Work using E-model that finds some backbones have toll-quality today
  - Do not investigate network or routing problems

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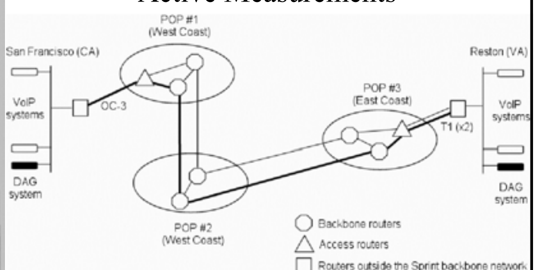
Measurement

- Passive
  - Via Sprint infrastructure
- Active
  - Induce own data

Passive Measurements

- Sprint has a passive measurement architecture
  - traces on more than 30 links in POPs
  - Includes 44 byte IP packet and timestamp via GPS reference signal
- Use traces from OC-12 (622 Mbps)
  - Jul 24th, 2001; Sep 5th, 2001; Nov 8th, 2001
  - Compute delays across backbone
- But
  - Can't get loss since → leave out non-monitored links
  - Can't control traffic source

Active Measurements



- Free BSD with 200 byte UDP traffic at 50 packets/second (G.711 compatible), Nov 27th, 2001 for 2.5 days

- have more data but it all looks similar

Verify no loss at last hops

DAGs provide GPS timestamps

Routing Data

- Capture IS-IS routing at POP #2
- Link-state
  - links assigned a weight
  - router broadcasts link weights to other routers
    - In Link State PDU (LSP)
    - Periodically and when topology change
  - When have path information from all, use SPF to construct route (called decision process)
- For some conditions (reboot), decision process can take minutes
  - Router sets paths infinite so not used for route

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Voice Call Rating – The E-model

- Combine loss and delay into single rating
- Use to compute Mean Opinion Score (MOS)
  - ITU recommendation

R-value range	MOS	Speech transmission quality
100 – 90	4.50-4.34	best
90 – 80	4.34-4.03	high
80 – 70	4.03-3.60	medium
70 – 60	3.60-3.10	low
60 – 0	3.10-1.00	very poor

- Below 60 unacceptable
- Above 70 is toll quality
- Above 90 is excellent

The E-Model

$R = R_0 - I_s - I_d - I_e + A$

- R_0 is effects of noise
- I_s is impairments in signal (quantization)
- I_d is impairment from mount-to-ear delay
- I_e is impairment from distortion (loss)
- A is advantage factor (tolerance)
  - Different for different systems
  - Example: wireless is a "10"
  - Since not agreed upon, drop further
- *(Ok, but how does it map to transport layer?)*

The E-model at the Transport Layer

- Since R_0 (background and circuit noise) and I_s (quantization) are impairments on signal, not underlying IP network
  - Use defaults [4] for voice

$$R = 94.2 - I_d - I_e$$

The E-model at the Transport Layer

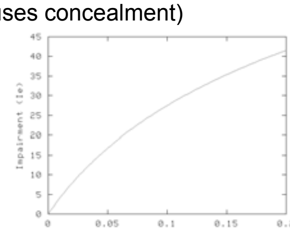
- I_d includes expression encompassing entire telephone system
- Simplify
  - All delays collapse into one: mouth-to-ear
  - Use defaults [4] for all save for IP network delay

$$I_d = 0.024d + 0.11(d-177.3)H(d-177.3)$$

- d is mouth-to-ear delay
  - Encoding (packetization)
  - Network (transmission, propagation and queuing)
  - Playout (buffering)
- H is "heavyside" function
  - H(*x*) = 0 if *x* < 0
  - H(*x*) = 1 if *x* > 0

The E-model at the Transport Layer

- No analytic model for I_e (impairment)
  - Must use subjective measurements
  - Appendix includes samples for different encodings
- Focus on G.711 (uses concealment)



- Effects of loss is logarithmic
  - $I_e = 30 * \ln(1 + 15 * e)$
  - (*e* is loss probability)

The E-model at the Transport Layer

- Summary R-factor:

$$R = 94.2 - 0.11(d-177.3)H(d-177.3) - 0.024d - 30 * \ln(1 + 15 * e)$$

(Linear with delay, logarithmic with loss)

Call Generation

- Emulate arrival of short business calls
- Poisson distribution, mean 60 seconds
- Durations from exponential distribution, mean of 3.5 minutes [17]
- Simulate talkspurts (*what* and *why*?) from exponential distribution of 1.5seconds [15]
- Fixed buffer size of 75 msec
  - Not adaptive as represents worst case
- Can then get mount-to-ear delay + loss

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  - Delay
  - Failures
  - Voice Quality
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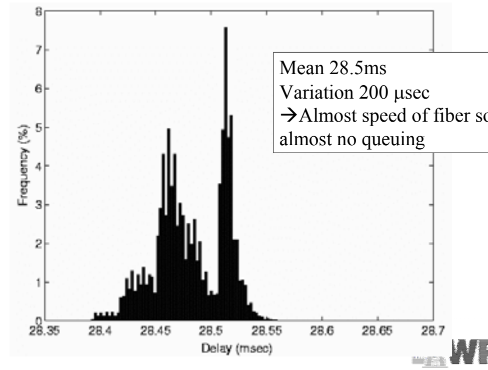
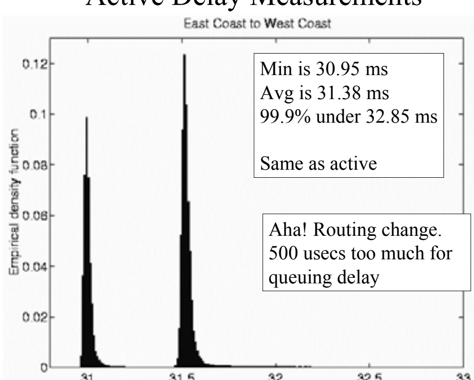
Passive Delay Measurements



Active Delay Measurements



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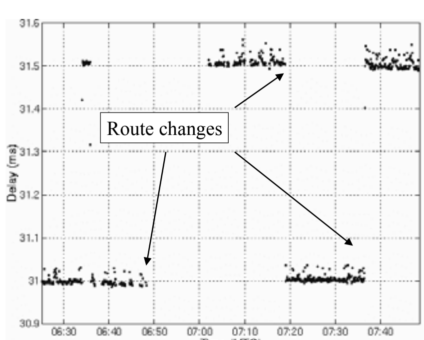
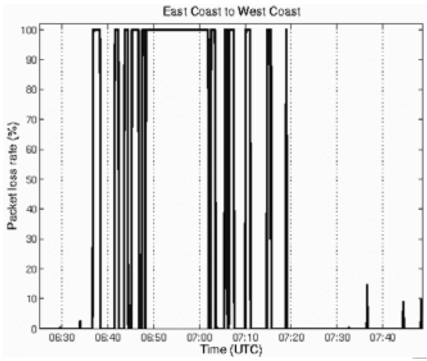
Impact of Failures on Data Traffic

- During weeks of study, only 1 failure
  - But distributed traffic for about 50 minutes
  - Periods of 100% loss

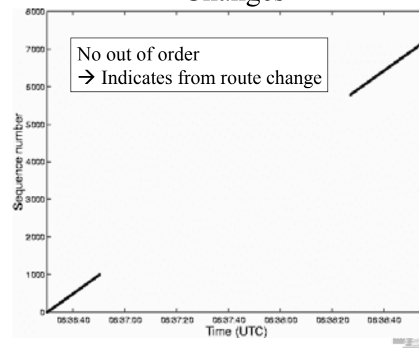
Delay from Route Changes



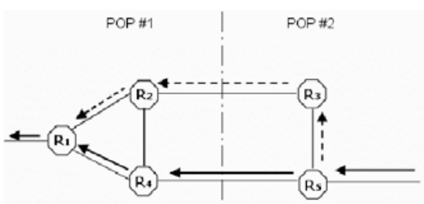
Loss from Route Changes



Packet Sequence Numbers during Route Changes



Routers involved in Failure



Solid is primary
Dashed is backup
R4 has problems

Router Messages

Time	IS-IS LSPs	Impact on traffic
06:34	R_1, R_2, R_5: link to R_4 is down	Re-routed through R_3 in 100ms
06:35	R_1, R_2, R_5: adjacency with R_4 recovered	Re-routed through R_4
from 06:59 to 07:06	R_1: link to R_4 "flaps" 7 times	100% loss periods. Re-routed through R_3
from 07:00 to 07:17	R_2: link to R_4 "flaps" 5 times	100% loss periods. Re-routed through R_3
from 07:04 to 07:17	R_5: link to R_4 "flaps" 4 times	100% loss periods. Re-routed through R_3
07:07	R_1: link to R_4 is down	Re-routed through R_2
07:17	R_1, R_2, R_5: link to R_4 is definitely up	Traffic restored on the original path

(Rebooted at 6:48, but does not set bit so 100% loss Until 6:59)

Summary

- 6:34 to 5:59 caused by instability in router R4
- 6:48 to 7:19 caused by R4 not setting infinite length bit
- Recommendations
  - Not from IS to IS protocol (so MPLS would not help)
  - Engineers should work on improving reliability of hardware and software

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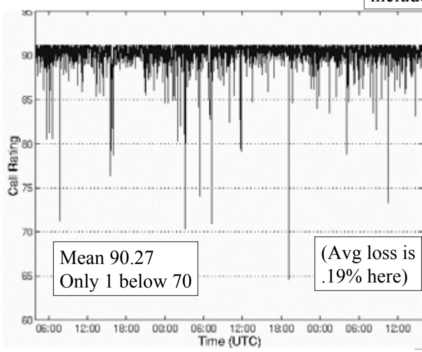
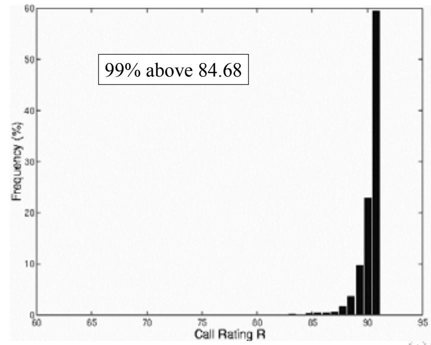
Voice Quality

(Does not include failure)



Distribution of Voice Call Ratings



Loss Burst length

- Model assumed independent losses

Loss burst length	Frequency of occurence
1	90.42%
2	8.71%
3	0.71%
4 and above	0.16%

Majority single losses
→ Packet loss concealment should help
99.84% less than 4

Conclusion

- Evaluate VoIP over backbone via passive and active measurements
- Toll quality can be delivered
  – Delay and loss typical of traditional phone systems
- Degradation mainly through link and router failures
  – Not from routing protocols but from equipment
  – More important as hops increase

Future Work

- More experiments
  – Want overall likelihood of link failure
- Compare network availability with telephone availability
  – FCC defines standards that affect 90k lines for more than 30 minutes
  – Difficult to define for IP since no "lines", customer count tough, and outage could be from non-network (ie- DNS) cause