VoIP Measurements Revisited

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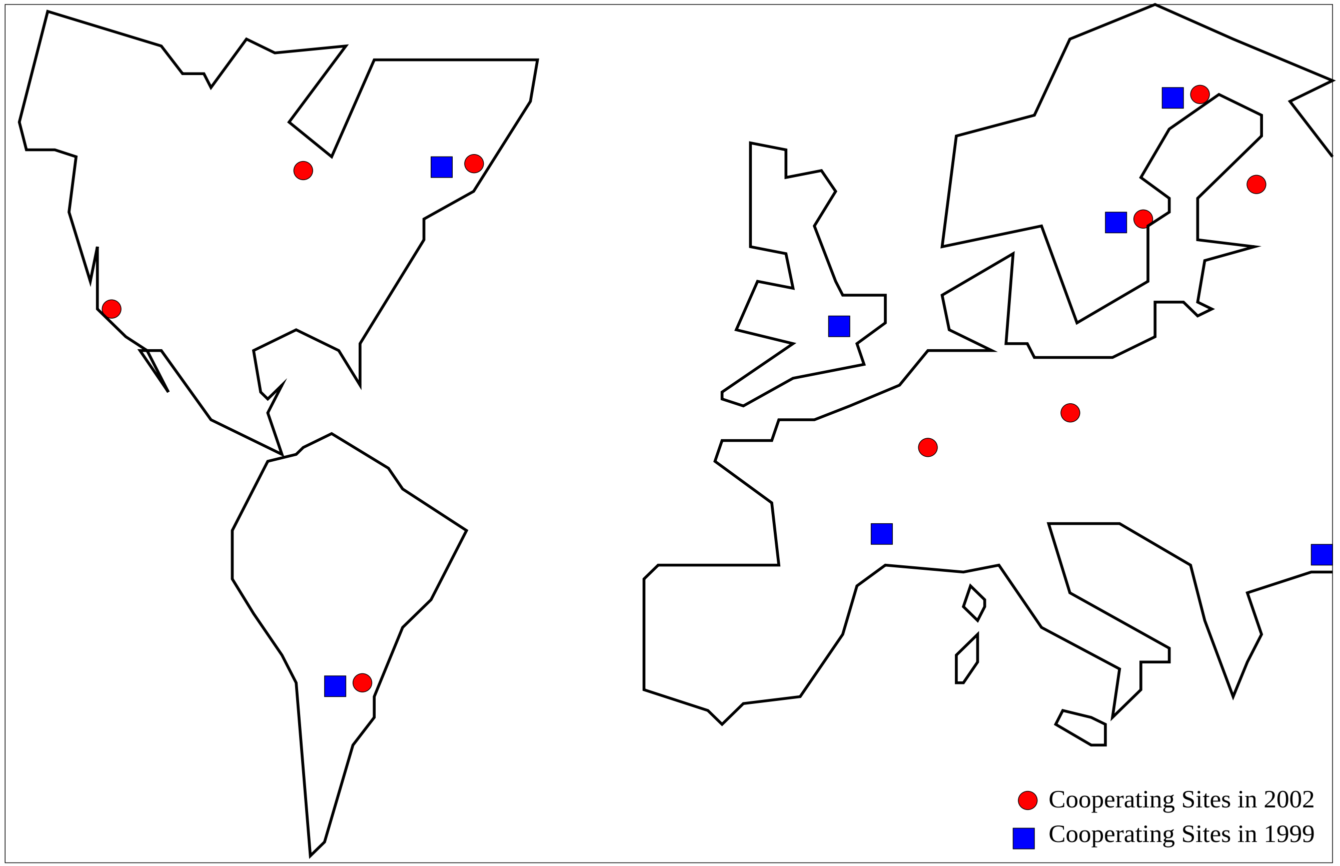
Goals

- Produce: loss, delay and jitter results for VoIP
- Compare with results obtained in 1999
- Construct a robust measurement infra-structure
- Collect a repository of VoIP trace files
- Stimulate collaboration in COST 263 group
- Use China's large labour workforce :-)

Lessons Learned from Last Measurements

- Make better use of available sites (full-mesh configuration)
- Perform simultaneous measurements. e.g. bidirectional
- Process data *both* in real-time and off-line automatically

Co-operating Test Sites



- 10 hosts (New Zealand added this morning)

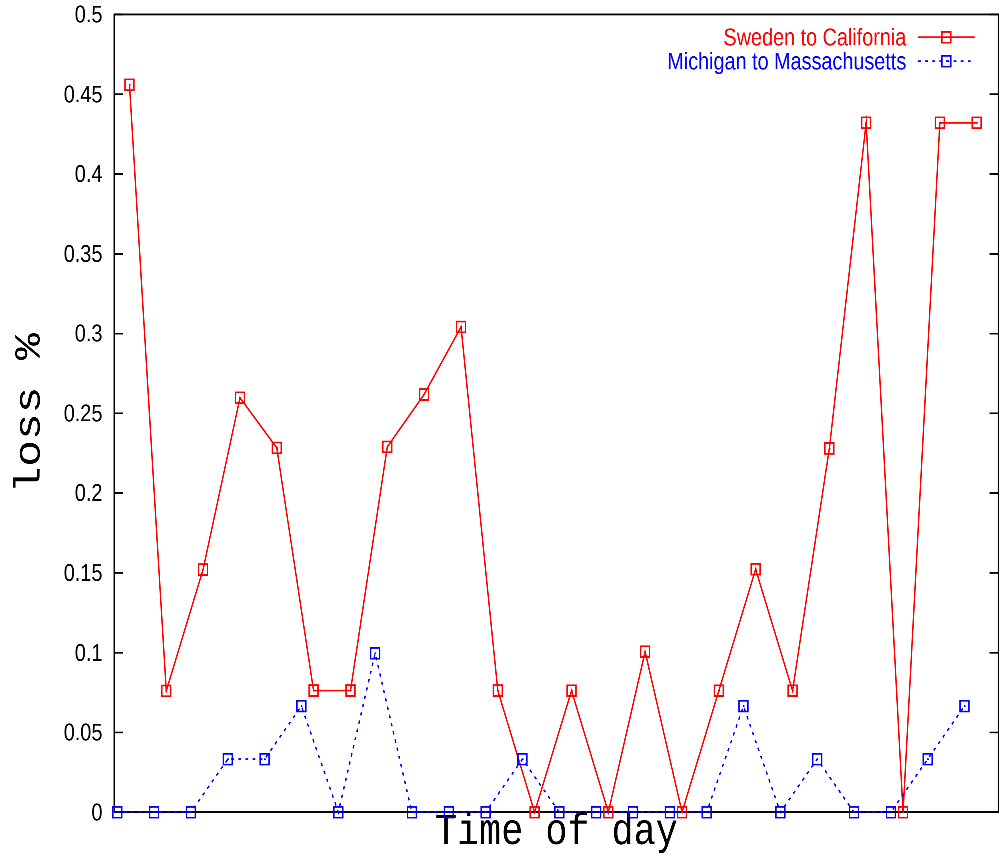
Methodology

- Send pre-recorded wav file between sites
- Currently once per hour (mainly to keep load down)
- Use script “wrappers” around Sics*o*phone
- Using scripts allows any tool to be used (even w/o source)
- Use ssh and scp to login without passwords
- Timestamp files as GMT offset (post-processing easier)

Used and Obtained Data

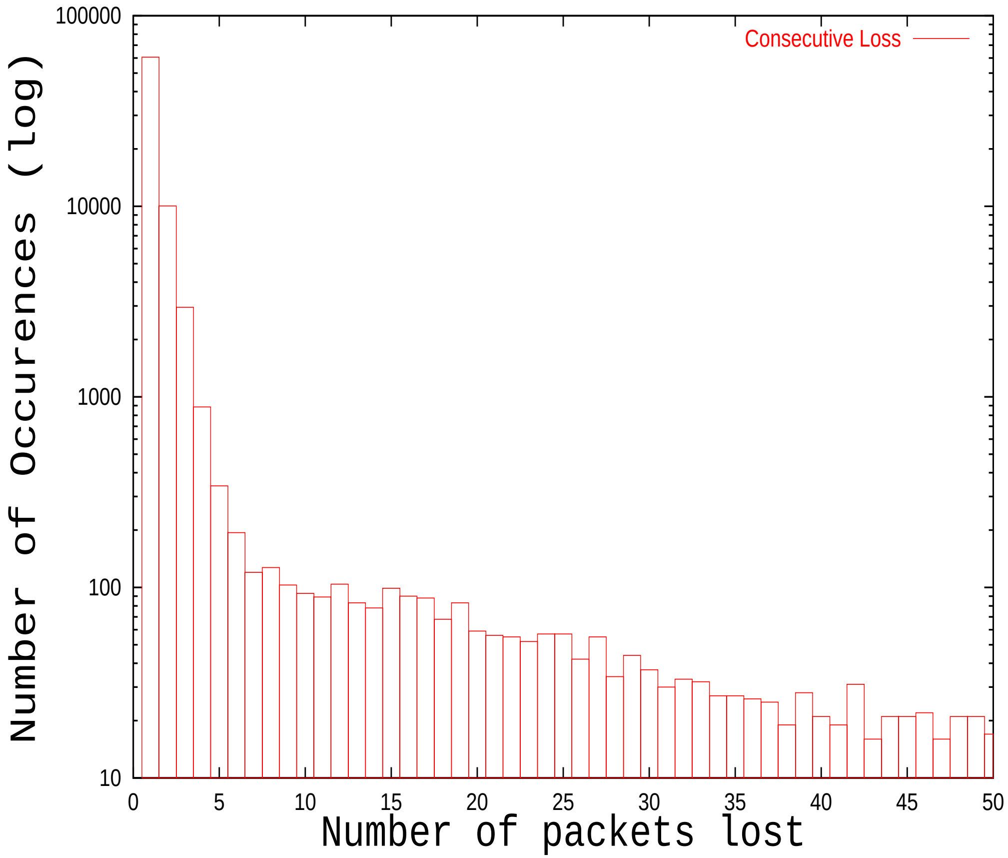
Used Data	**Details**
Call duration	70 secs
Silence Sup.	2043 packets
No Silence Sup.	3653 packets
Coding	8 bit PCM
File Size	584480 bytes
Obtained Data	
Total Trace Files	22436
Bidirectional Tests	3742
Differing Packet Size Test	2491
Suppression Tests	1251
Measurement Duration	12 weeks

Loss



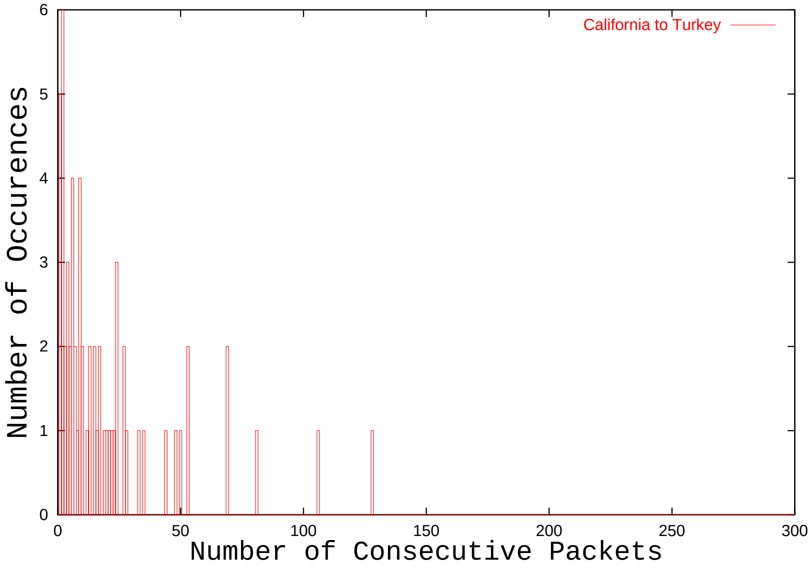
- X-axis receiver time for each plot

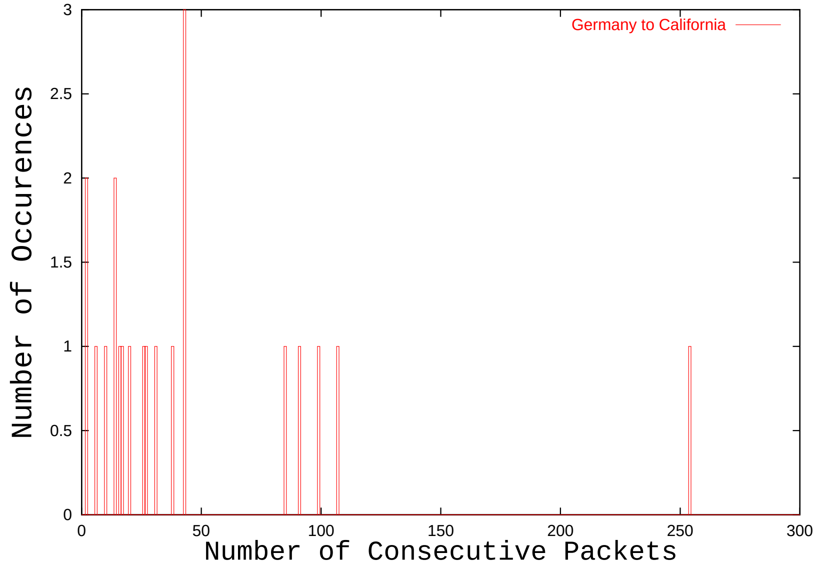
Consecutive Loss



- Log scale on Y-axis

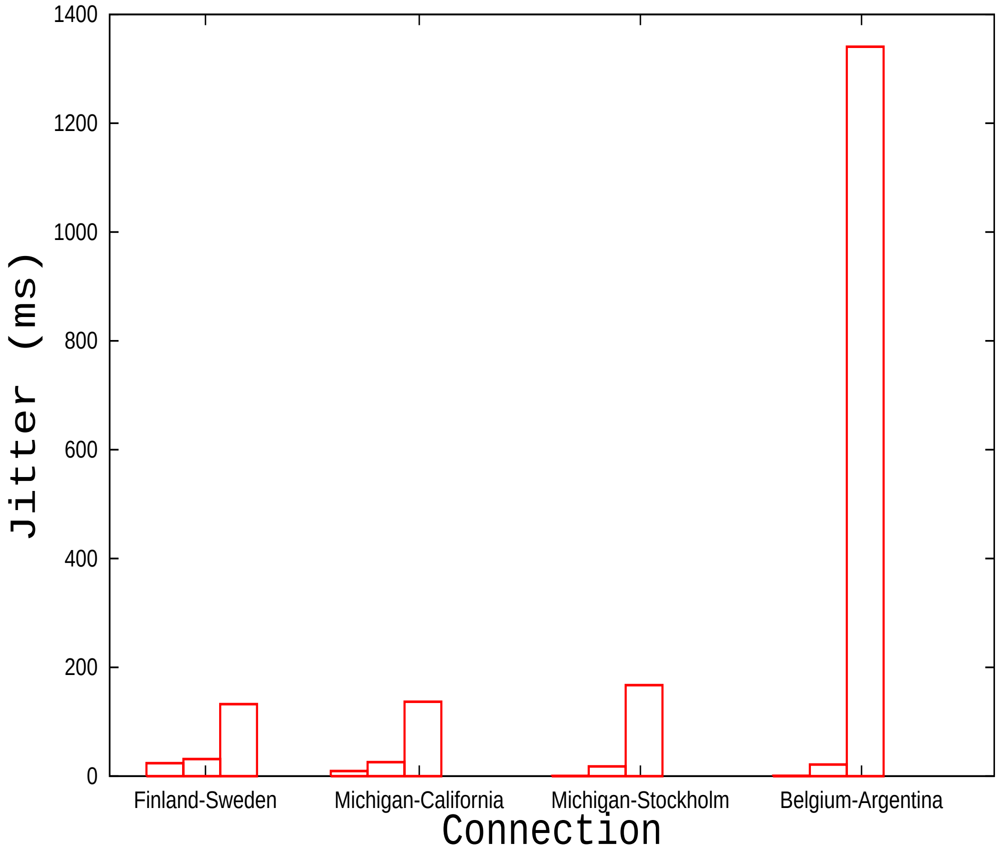
Packet Sequences (non-loss)





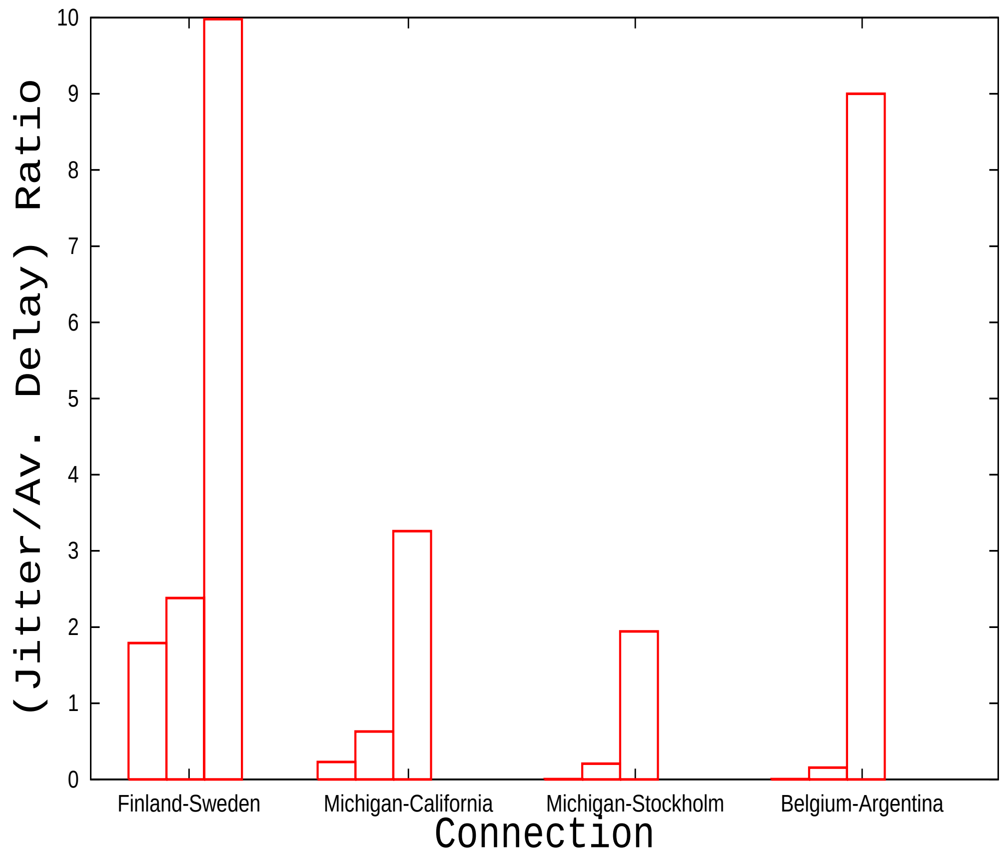
- Looked at traces w/out suppression (long sequences possible)
- Successive packets over talkspurts is not interesting (FEC)

Jitter



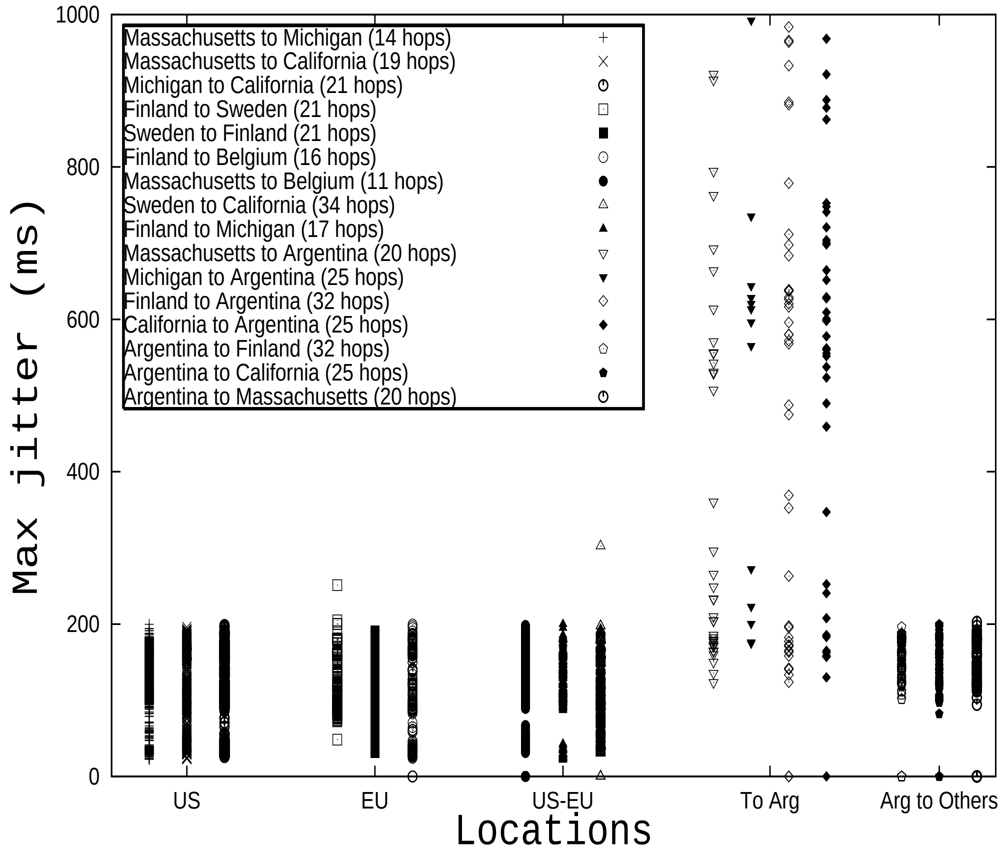
- Most sites experience low jitter

Jitter to Delay Ratio



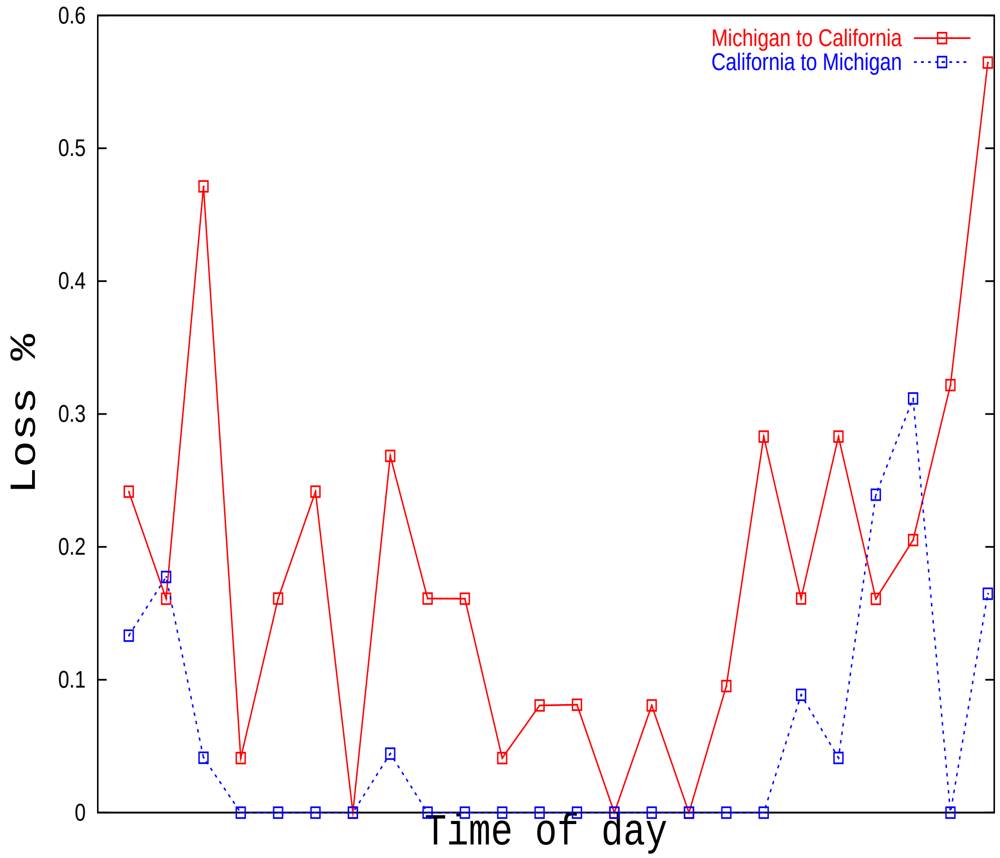
- This time look at the jitter / average delay

Maximum Jitter by Region



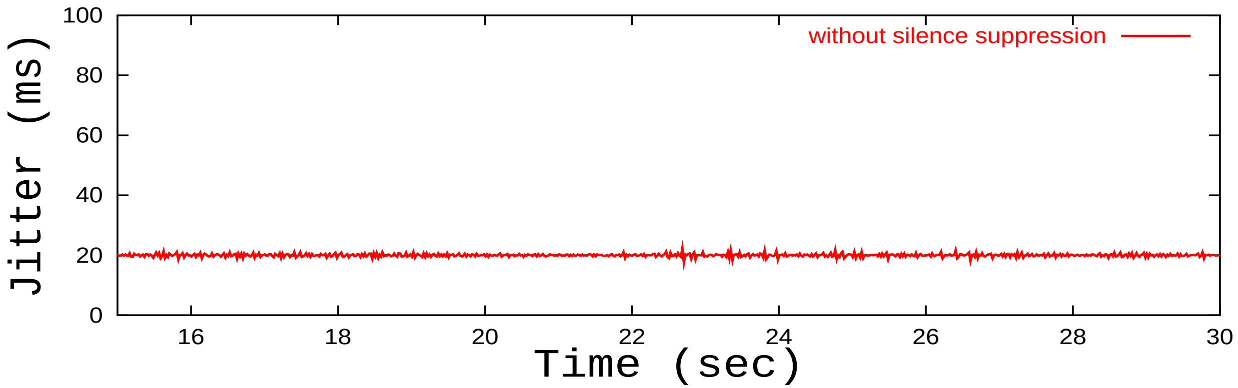
- Emphasises the poor quality *to* South America

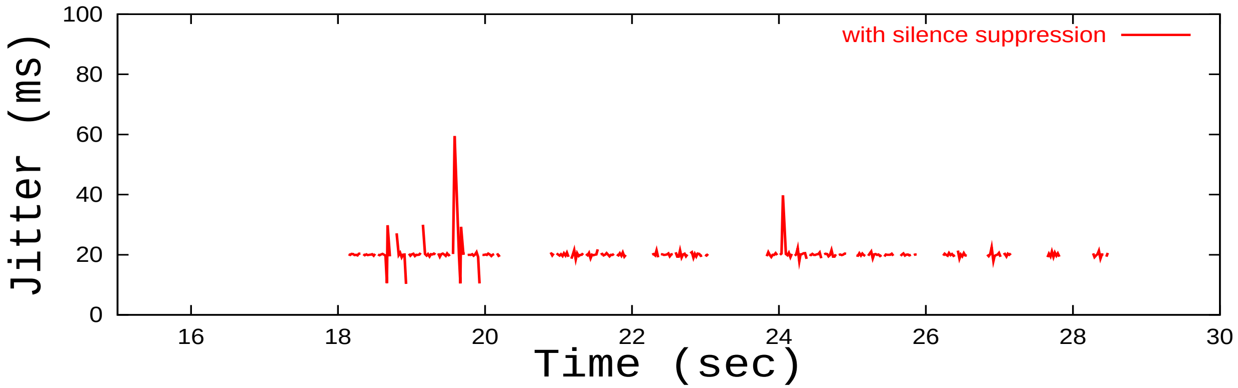
Asymmetry



- There is a difference - but in absolute terms is low

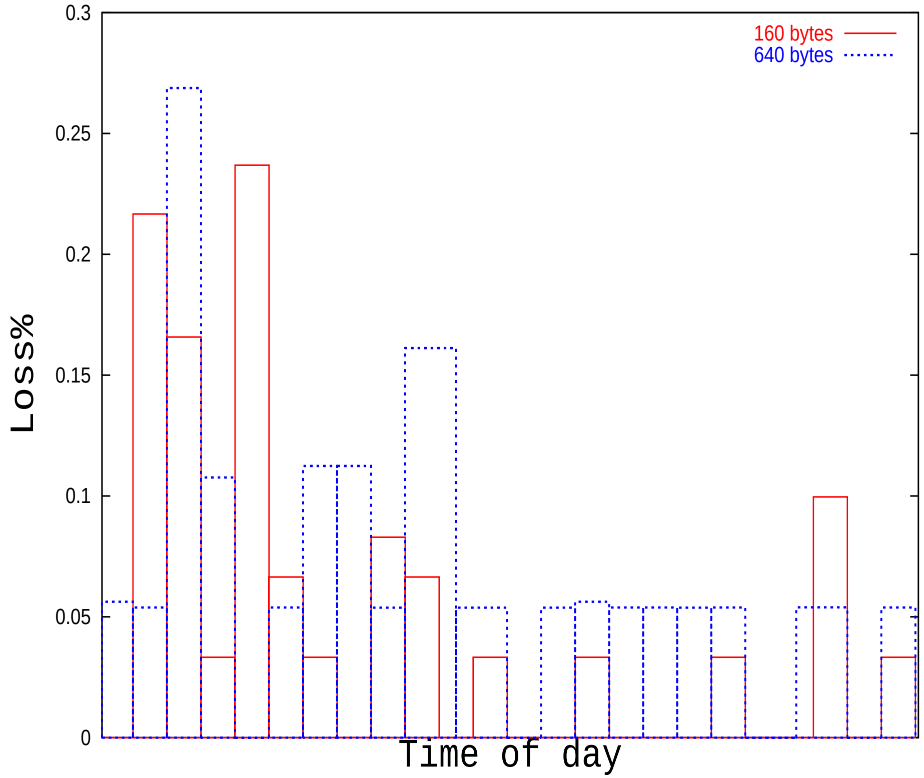
Silence Suppression





- Taken at the same time

Packet Size



- Size does not affect loss (we tried 160, 640 and 1280 bytes)

1999 and 2002 VoIP Quality Differences

Quality	**1999**	**2002**	**% Diff. (+/-)**
Jitter	45.1ms	22.6ms (± 13.7)	-50.0%
Loss	1.2%	0.5% (± 2.6)	-58.3%
Delay	115 ms	84.95ms (± 44.85)	-26.1%

- Loss, jitter and delay improved
- Attribute it to upgrading links
- 2002 we had 1000's of traces whilst 1999 fairly few in comparison

Related Work

- Bolot (and Hardman RAT) looked consecutive loss 1-5 at 8am and 1-10 at 4pm
- Maxemchuk investigated delay and loss, depends on length of connection and time of day - we had lower loss rates
- Dong Lin results similar to Bolot (even 4 years later)
- She concludes packet size is not significant (we found the same)
- As far as we know no-one looked at silence suppression

Suggestions

- Plan carefully what you want to measure
- Be prepared for outages, configuration changes
- Have contact with your remote colleagues because...
- Router configurations, reboots, reverse traceroute, sprobe etc. *will* be needed ...

Conclusions

- VoIP quality has *improved* over the past 3 years
- VoIP still feasible (G.114) but *not* on a global scale
- Vast majority less than 5% loss and under 50ms jitter
- Gathered over 22,000 sessions (and still going)
- Plenty of practical difficulties, firewalls, ssh versions...
- Trace files, Paper, Java, Perl scripts available from: http://www.sics.se/~ianm/COST263/cost263.html