Early Estimation of Voice over IP Quality

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The big picture

- Motivation:
  - IP telephony is becoming more widely used
  - Users are sensitive to poor quality
  - Poor quality calls are a waste of
    - end users time
    - valuable network resources
  - System reacts faster than users

- Goal:
  - Early prediction of VoIP quality

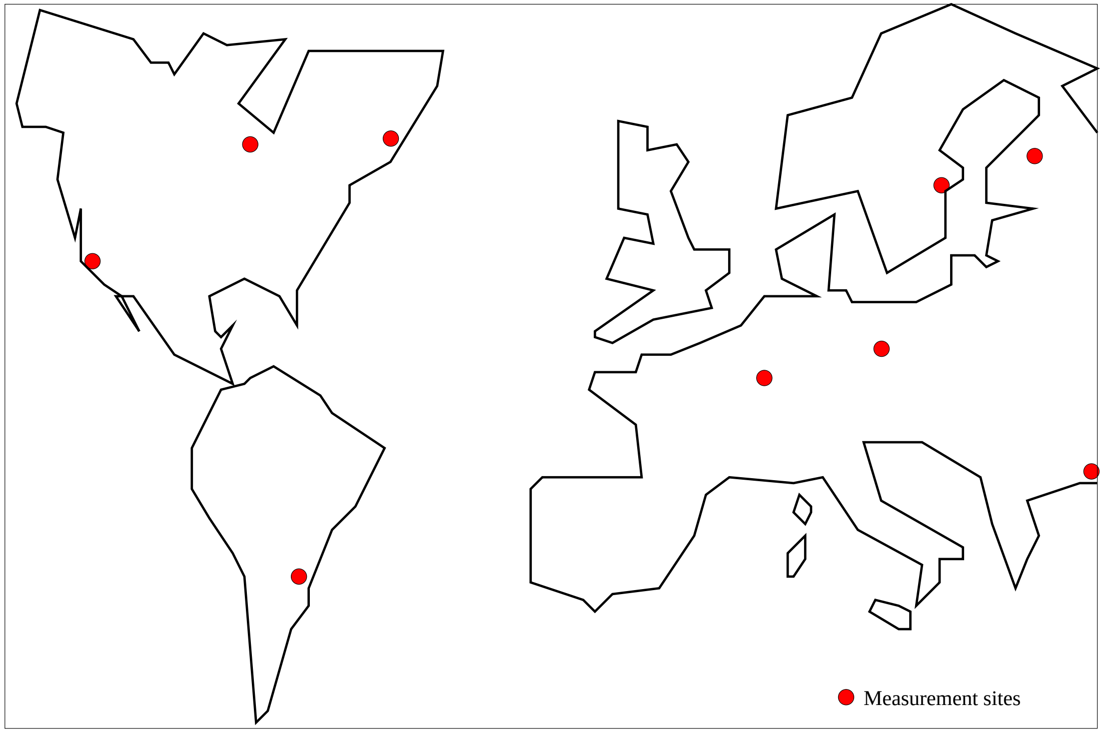
Outline of the talk

- Motivation and goal
- Assumptions and methodology
- Measurement description
- Loss measurement for a single call
- Correlation analysis over the sample set
- Conclusions and future work

Assumptions and methodology

- Paths and end systems remain unchanged
- Losses only in the fixed network infrastructure
- Methodology:
  - Probe the network state
    - Send prerecorded conversation between test sites
  - Measure the network state
    - Packet loss is the quality indicator

Measurement infrastructure



- Nine academic sites in a full mesh topology
- Large differences in time zones, hops and distances

Measurement data

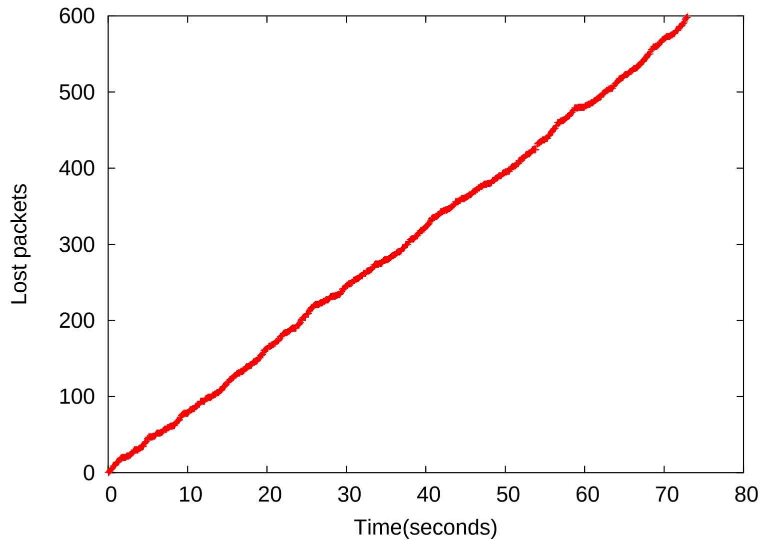
Test "signal"	
Call duration	70 seconds
Payload size	160 bytes
Packetisation time (ms)	20ms
Data rate	64kbits/sec
With silence suppression	2043 packets
Without silence suppression	3653 packets
Coding	8 bit PCM
Recorded call size	584480 bytes
Obtained data	
Number of hosts used (2003)	9
Number of traces obtained	18054
Number of data packets	32,771,021
Total data size (compressed)	411 Megabytes
Measurement duration	15 weeks

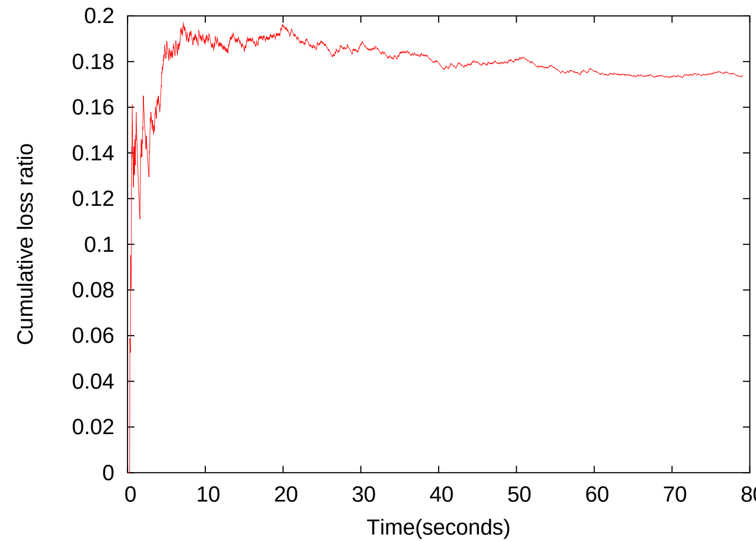
Sample set reduction:

- Calls with only 20 ms packetization
- No silence suppression
- Lossy calls

This reduction results in a subset of 564 calls, with all nine sites represented in the set

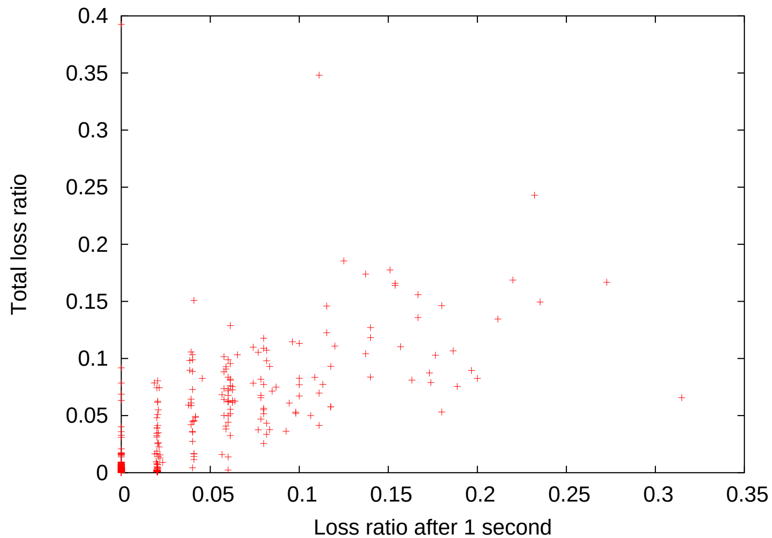
Loss behavior for a single call

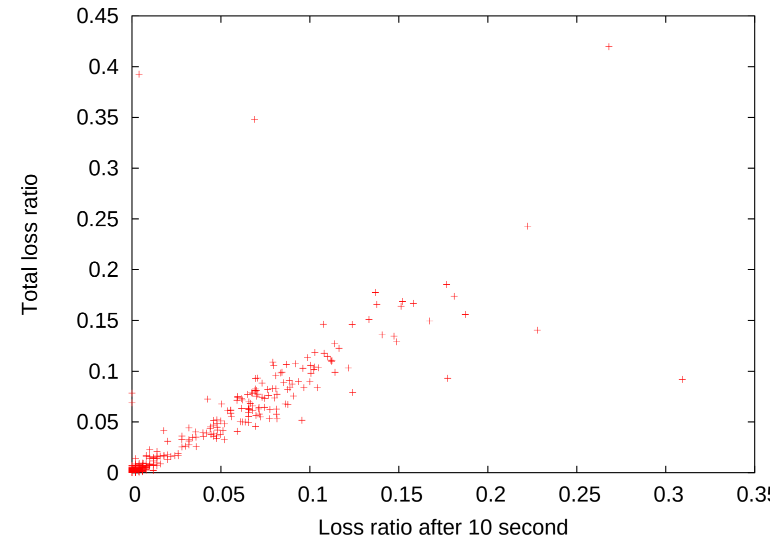




- Left plot: number of lost packets increases linearly
  - Prediction seems feasible
- Right plot: cumulative loss ratio
  - Indicates the estimation period

Loss ratio for one and ten seconds



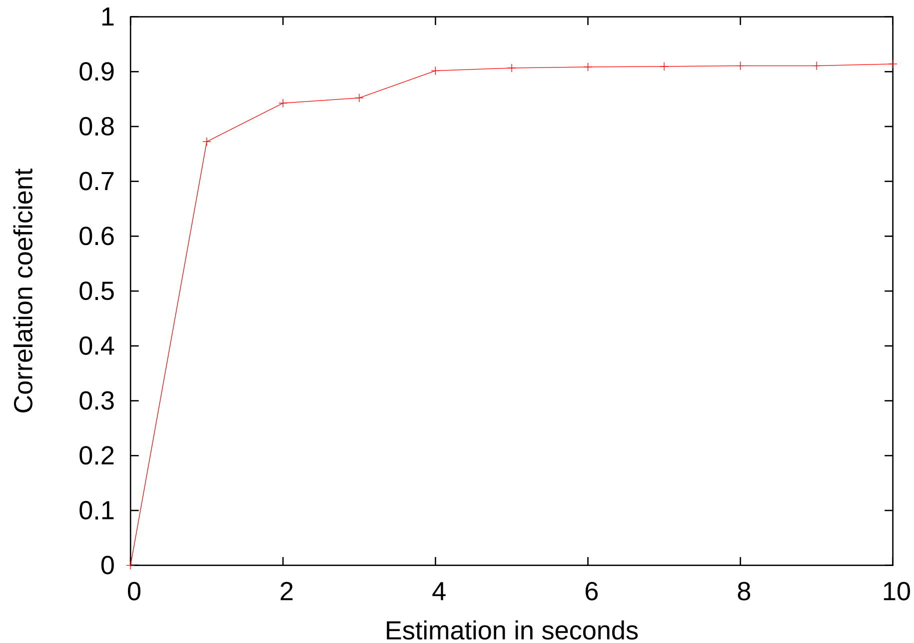


- Total loss against initial interval loss
- Each point represents a call between two hosts
- Better prediction as the points approach $y = x$

Statistical analysis

- Sample space with loss ratios:
  - for 1,2 ... 10 seconds (L_1, L_2 ... L_{10})
  - for the whole call (L_t)
- Correlation coefficient between L_1 and L_t, or L_2 and L_t, up to L_{10} and L_t
  - Statistical measure of the interdependence of two or more random variables

Loss correlation coefficient



- Correlation coefficient for all the calls
- The correlation increases as the probing interval increases
- Correlation stabilizes after four seconds

Conclusion and applications

- Conclusion:
  - Possible to predict the quality of VoIP calls
    - From our data set, measuring over the initial four seconds of a call suffices to perform an accurate prediction
- Possible applications:
  - Admission control
  - Quality assurance
  - Faster feedback from the system

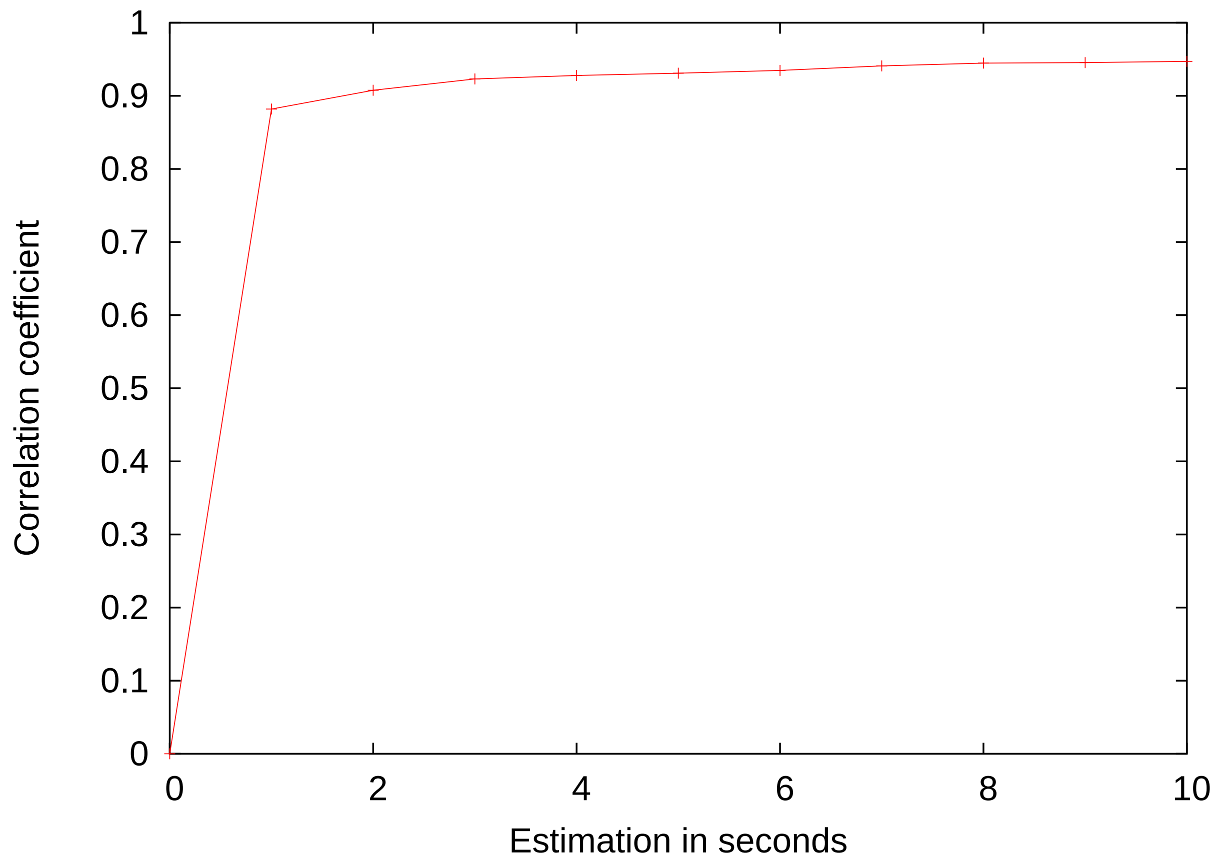
Future work

- Improvements:
  - Extend the repository with more communications environment, e.g. wireless links
  - Implement the early estimation algorithm in the real VoIP tool and perform real tests
- Perform an extensive statistical analysis to provide confidence interval to the prediction

Further references

- Ian Marsh and Fengyi Li.
  A VoIP measurement infra-structure.
  16th Nordic Teletraffic Seminar, Helsinki, Finland.
- Ian Marsh.
  Quality aspects of audio communications.
  Licentiate thesis, Royal Institute of Technology, Sweden.
- Ignacio Más Ivars.
  PBAC: Probe–Based Admission Control in IP networks.
  Licentiate thesis, Royal Institute of Technology, Sweden.

New results



- Correlation coefficient for the whole trace set