Analysis and Evaluation of Skype and Google Talk

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Ref: B. Sat and B. W. Wah, "Analysis and Evaluation of the Skype and Google Talk VoIP Systems," Proc. IEEE Int'l Conf. on Multimedia and Expo, July 2006.

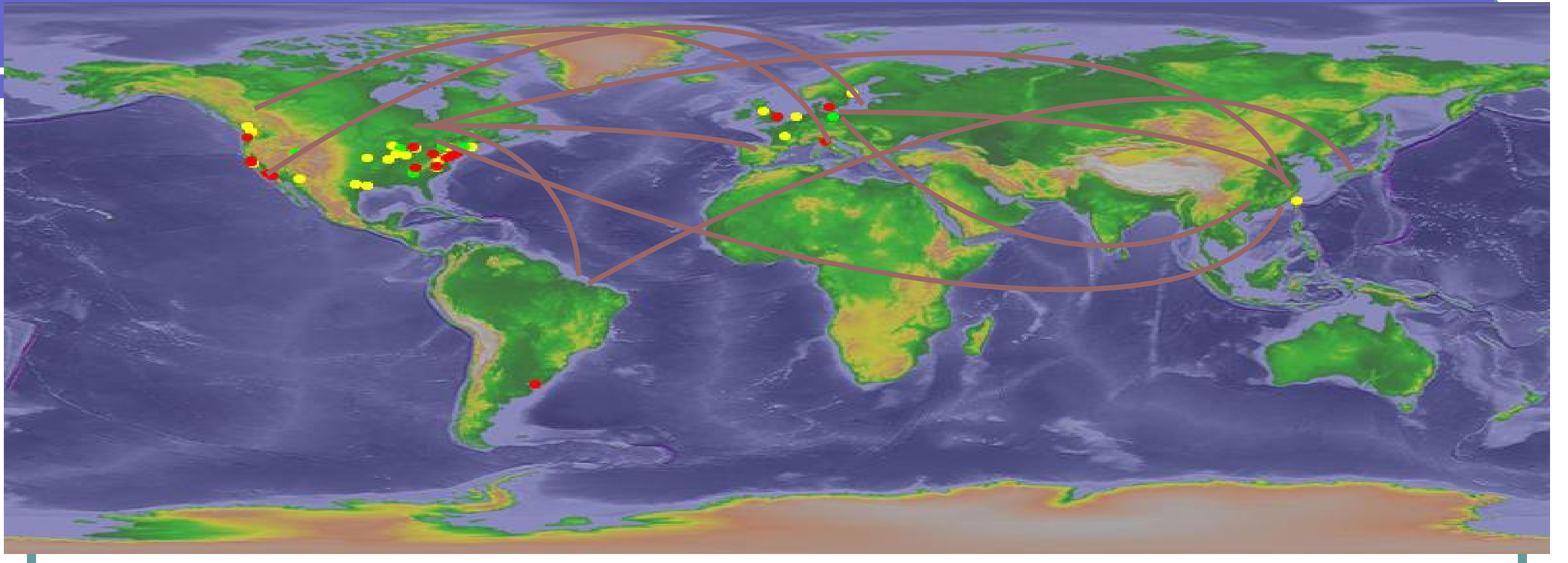
Problems Studied

- Motivations
  - Skype (1.3.0.65) & Google Talk (Beta) are widely used but proprietary
  - Understanding their limitations help design better VoIP systems
- Problem statement
  - Study the various components of Skype and Google Talk
  - Identify their limitations
  - Propose new VoIP system
    - Adapt better to a large range of network conditions

Outline

- Experimental Setup
  - Network evaluations
- Architecture of VoIP Systems
  - Speech Codecs
  - Packetization
  - Play-out Scheduling
  - Loss Concealments
- Experimental Results
- Future Work

Network Traces from PlanetLab



Collecting Internet traces from various geographical locations via PlanetLab nodes

Intra-continental paths:

- Intra-Asia, intra-Europe, intra-America

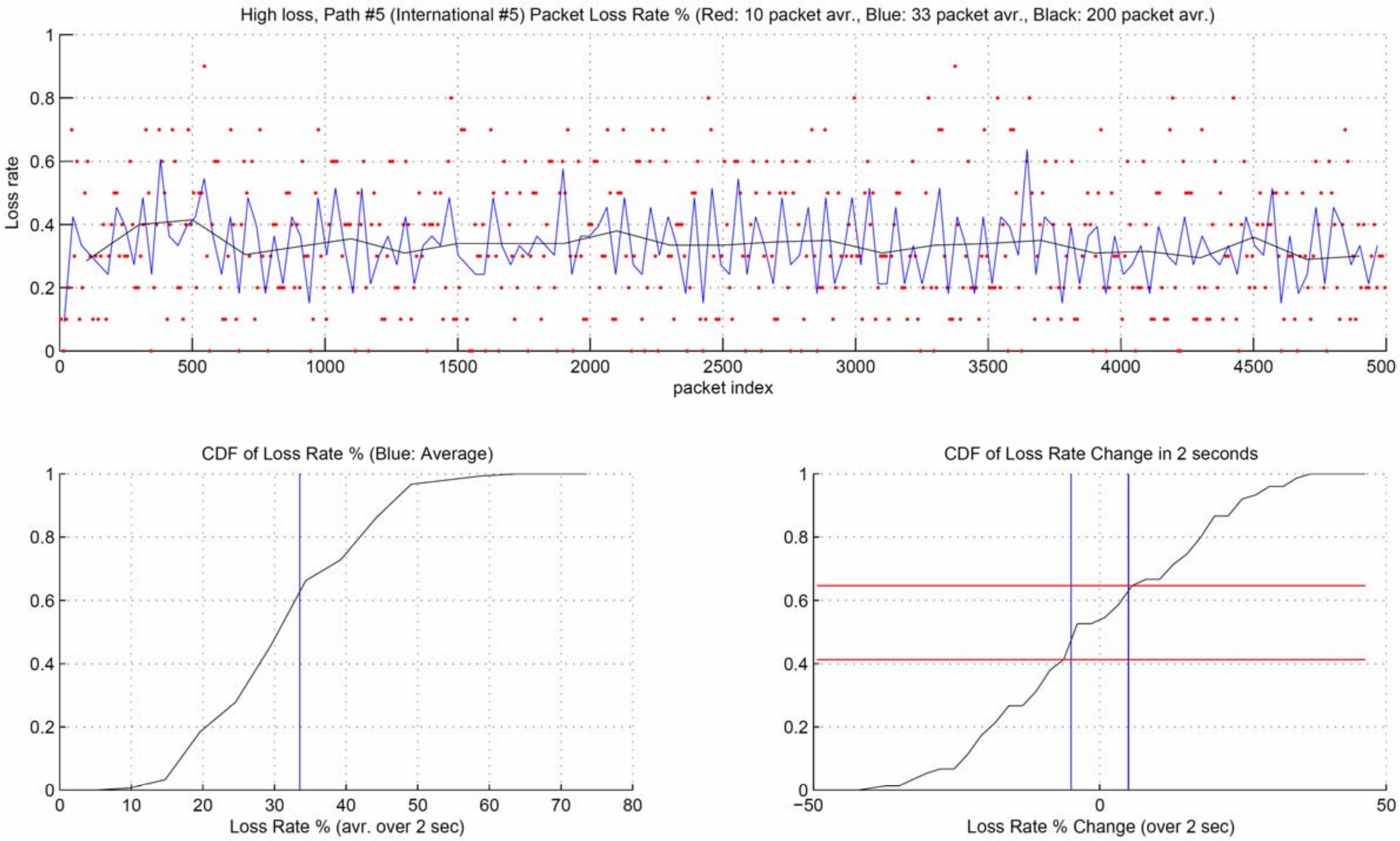
Inter-continental paths:

- North America – Europe/Asia/South America
- Asia – Europe/South America

Observations

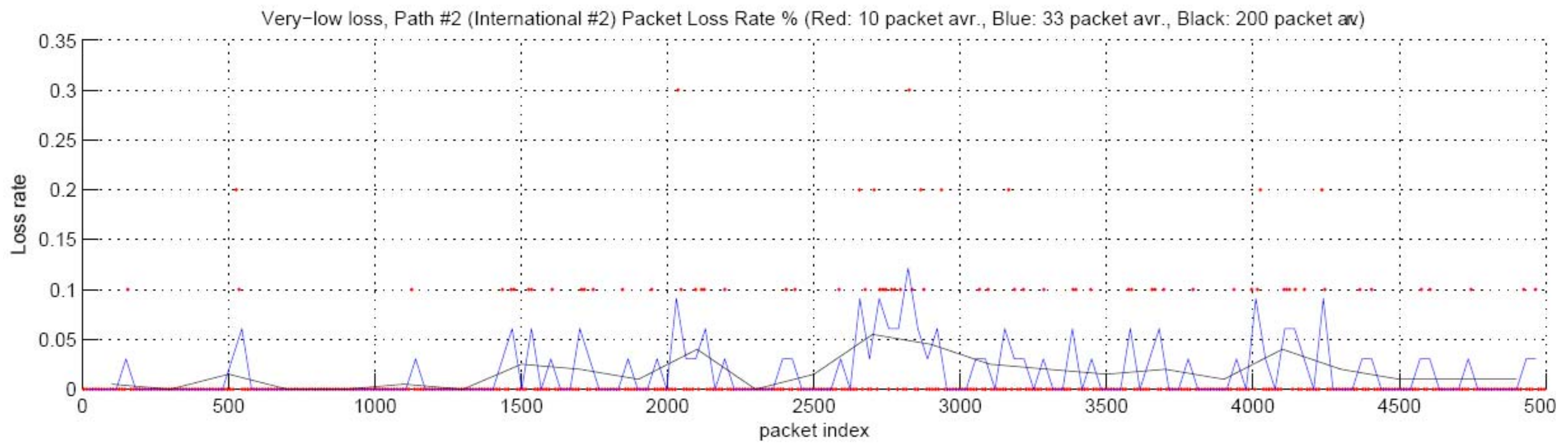
- Highly varying packet delays and jitters $\Rightarrow$ dynamic jitter control
  - < 50ms for most intra-continental paths
  - > 250 ms for some inter-continental and intra-Asian paths
- Highly varying packet loss rates $\Rightarrow$ dynamic loss concealment
  - < 1% for most intra-continental paths
  - > 33% for some inter-continental and intra-European paths
- Fast changing loss rates $\Rightarrow$ real-time estimation & frequent feedbacks

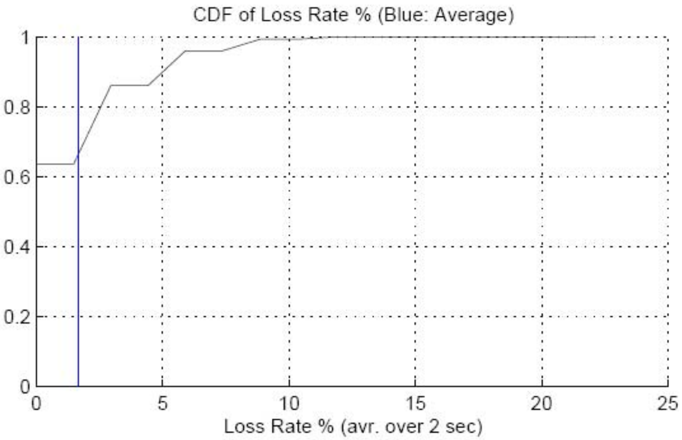
Connection with High Loss Rate



Bursty Losses: Single: 12.8%, Double: 7.8%, Triple: 0.7%, Quadruple: 12.2%

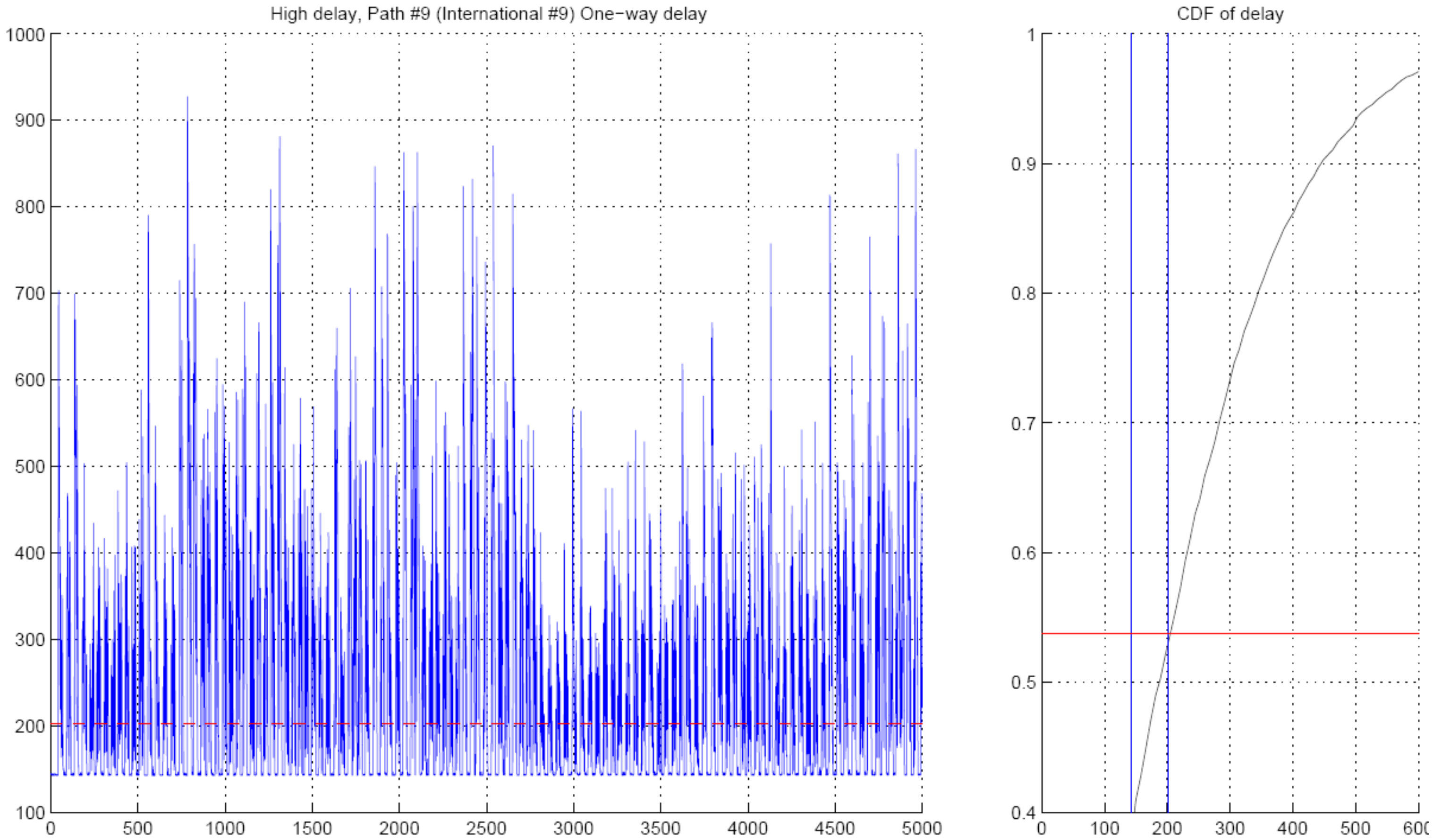
Connection with Very Low Loss Rate





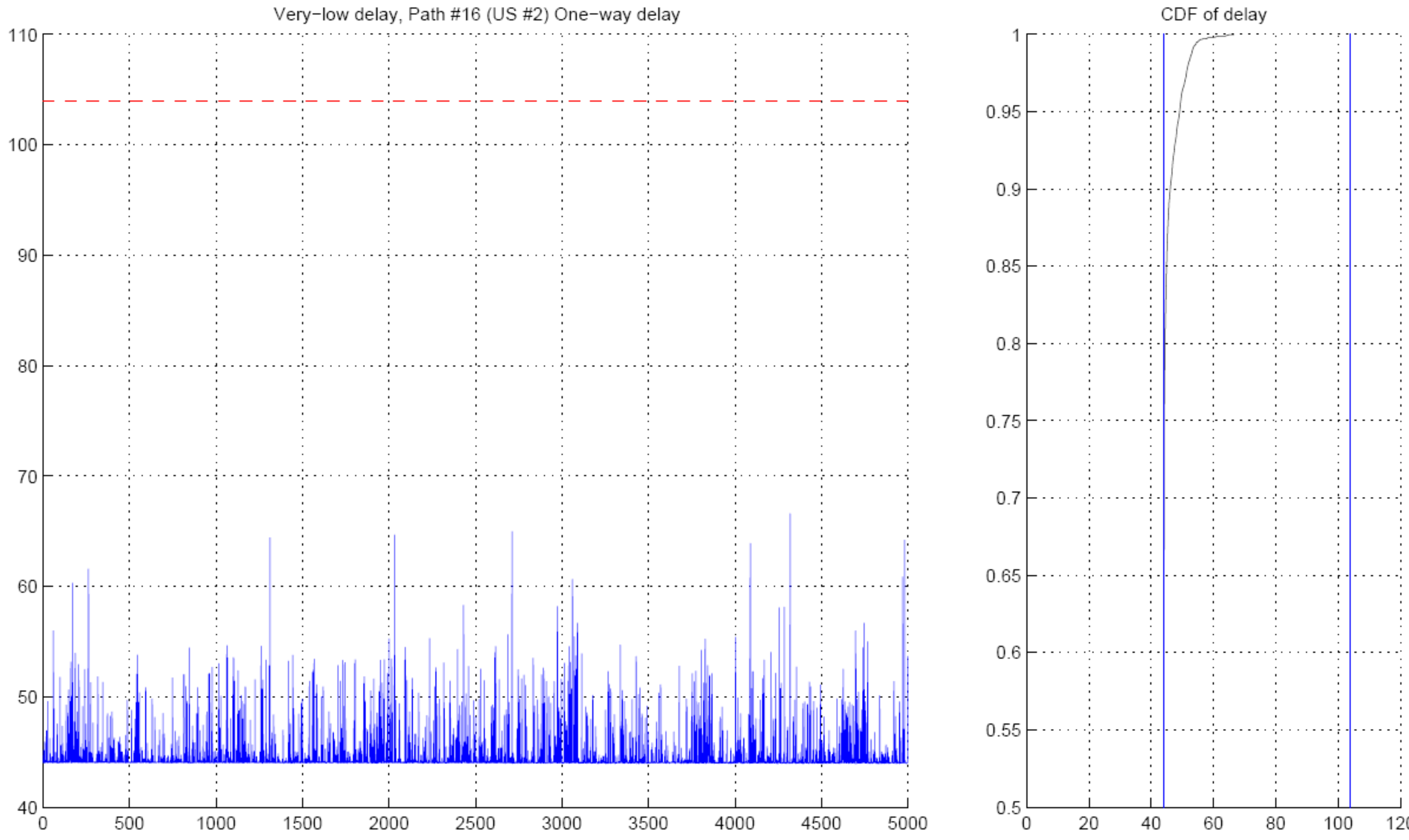
Bursty Losses: Single: 1.5%, Double: 0.1%, Triple: 0.1%

Connection with High Delay



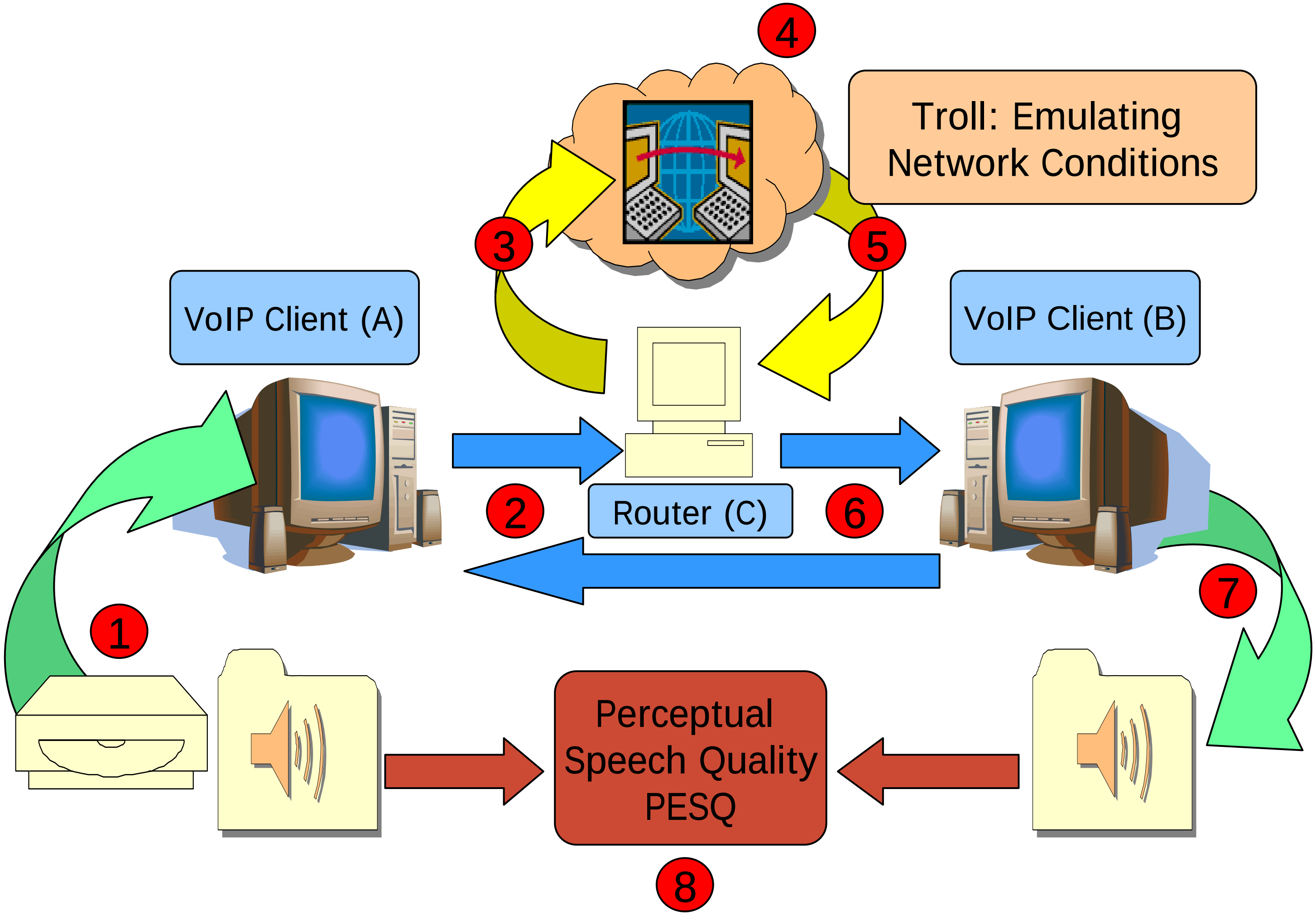
Out-of-order packets: 13%

Connection with Very Low Delay



Out-of-order packets: 0%

Experimental Setup



Outline

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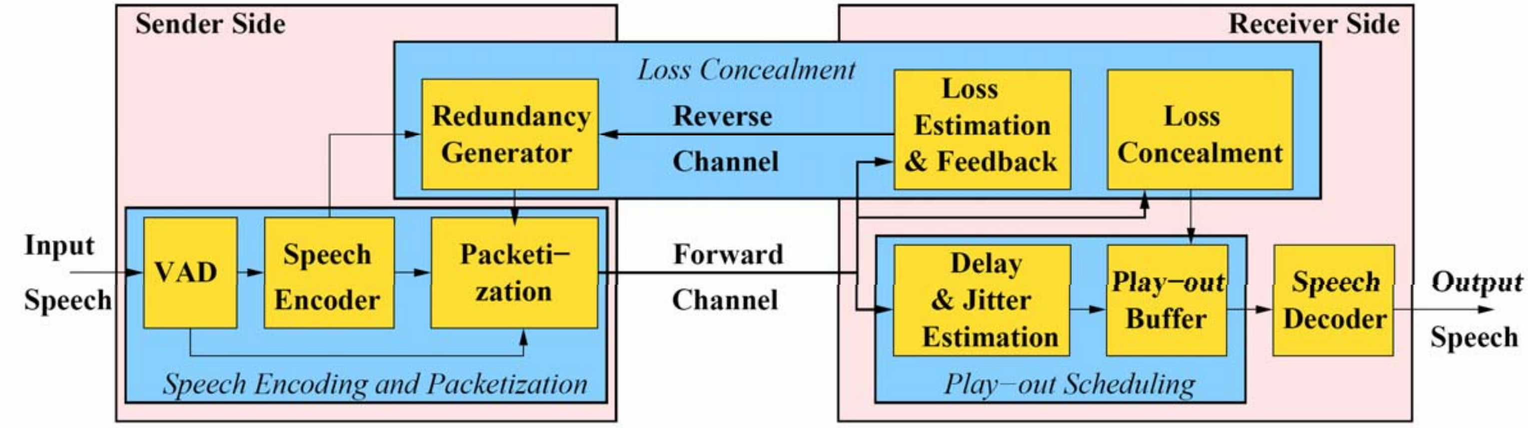
  - Network evaluations
- Architecture of VoIP Systems



- Experimental Results
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Speech Codecs

ITU G.729 low bit-rate codec

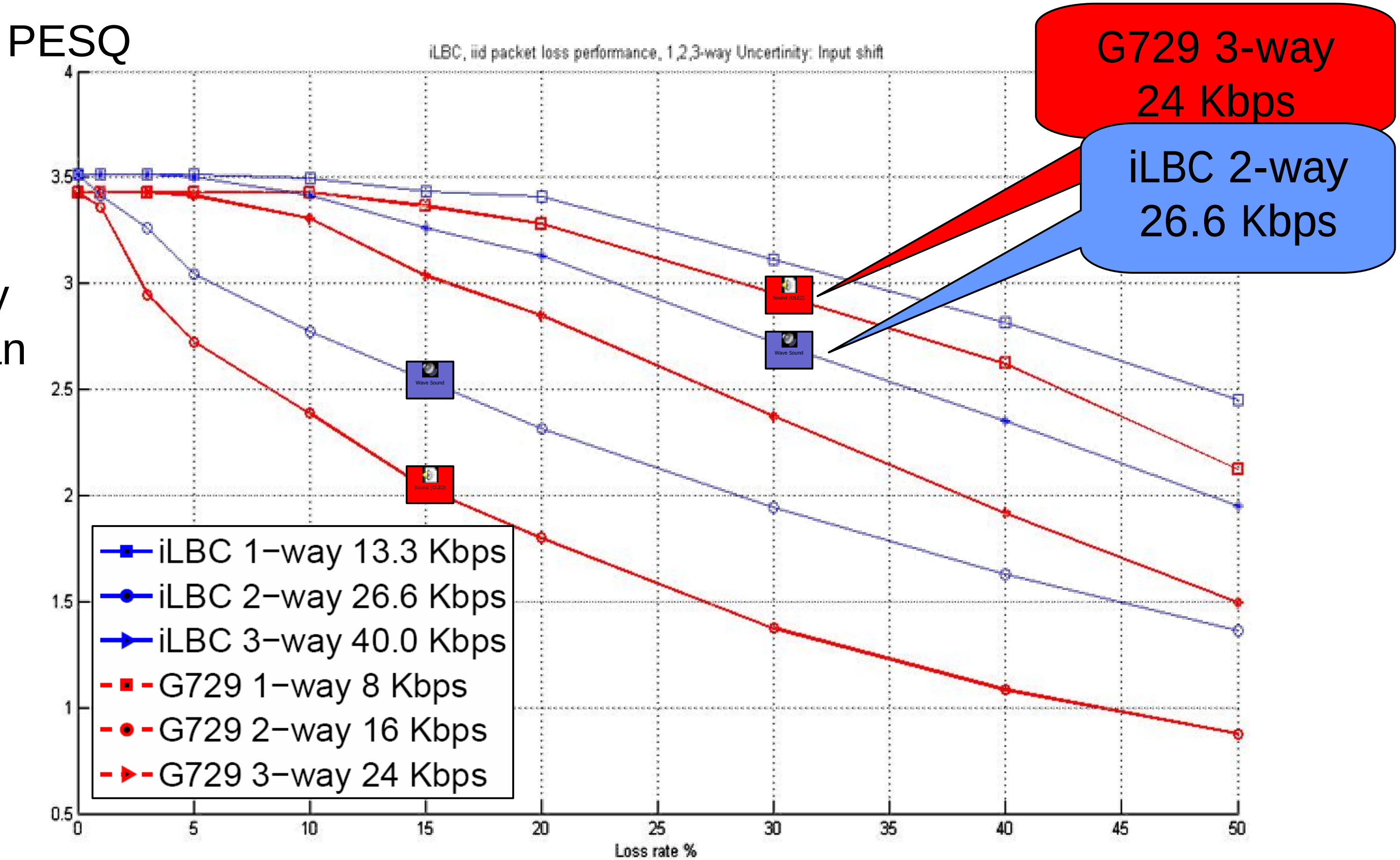
- High efficiency by removing redundancies across frames
  - Long propagation of errors once a frame is lost

Internet Low Bit-Rate Codec (iLBC—RFC3951)

- Encode self-decodable frames and remove internal states in receiver
  - 66 - 90% higher bit-rate than G.729
  - Achieve shorter recovery time and robustness to losses
- Inadequate without redundancies

Performance Under IID Losses

G729 3-way is better than iLBC 2-way with similar bit rate



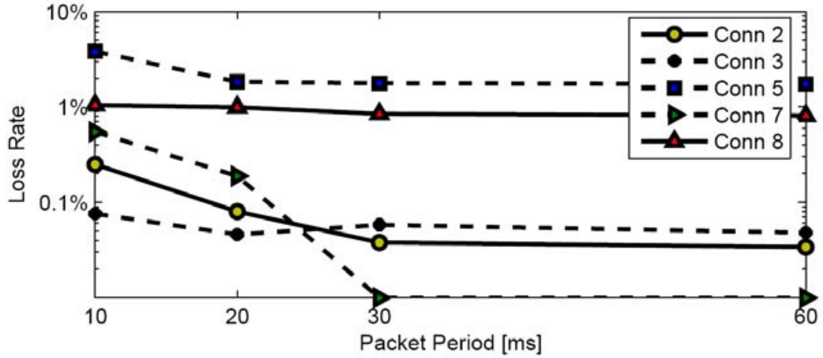
For loose end-to-end delay conditions, G729 performs better than iLBC for LR>10%

Packet Delays and Redundancy

- Skype and Google Talk use 60-ms packet period
  - Skype packs two 30-ms iLBC frames in a packet **(**14.4 Kbps)
  - Google Talk packs variable bit-rate PCM (24 Kbps)

Frame Per Packet	Packet Period	No Redundancy	2-way	3-way	4-way
1	30 ms	0 ms	30 ms	60 ms	90 ms
2	60 ms	30 ms	90 ms	150 ms	210 ms

- 3- and 4-way redundancies
  - Cannot be used with 60-ms periods
  - Can be used with 30-ms periods without extensive delay

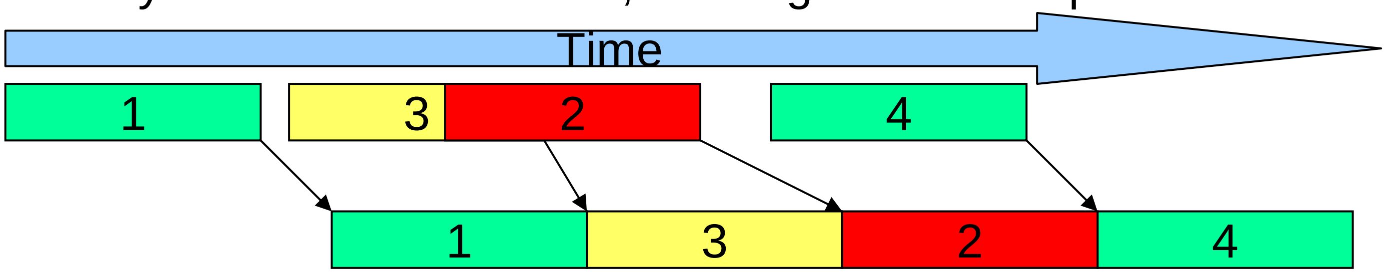


Loss rate not affected by reducing packet period from 60 to 30 ms

Play-out Scheduling

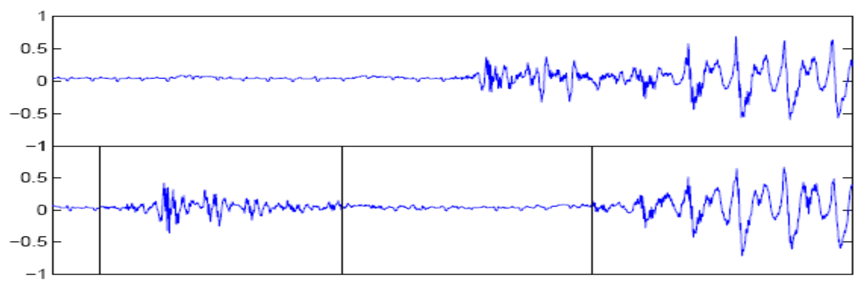
Skype uses simple FIFO play-out buffer

- 1st packet not received 60-ms later than the expected arrival time
  - Apply iLBC-level loss concealment (RFC3951)
- Subsequent packets not received beyond the 60-ms expected time
  - Played-out when received, causing the whole speech to shift



Original Waveform

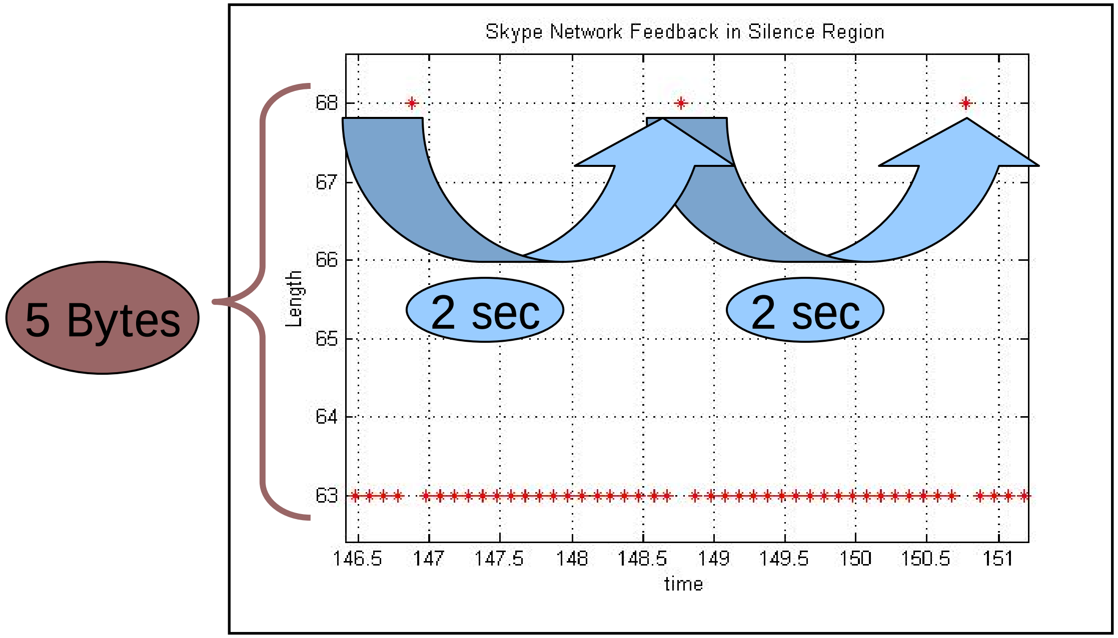
Degraded Waveform



Google Talk employs a jitter buffer similar to that of Skype

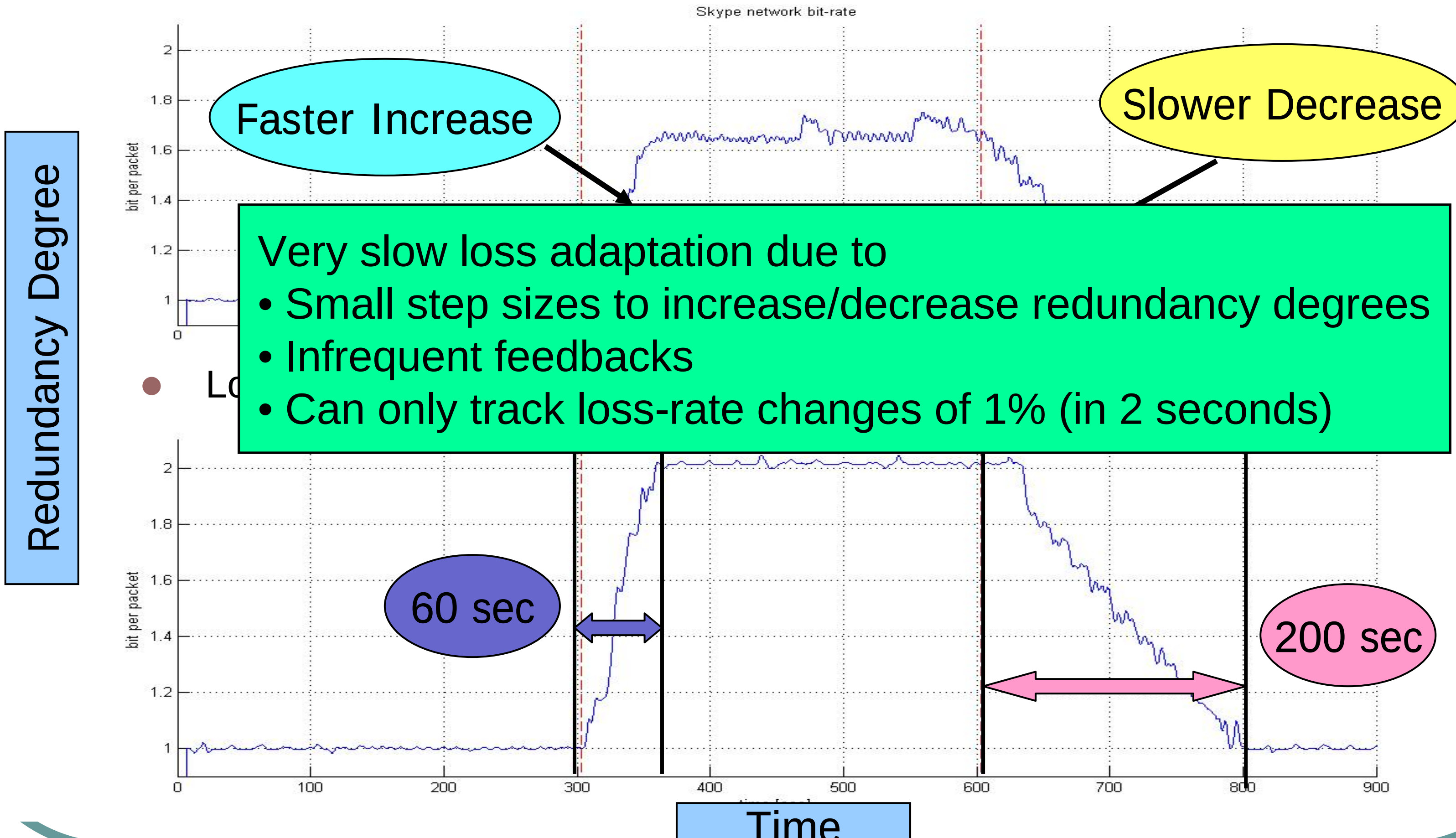
Skype: Loss Adaptation & Concealment Feedback Control

- Packets in Skype not arriving 600 ms beyond expected time are considered lost
  - Receiver sends decision to increase or decrease redundancy degree to sender every 2 seconds



Skype: Adaptation to Loss-Rate Changes

- Loss rate: Step function [0% - 15% - 0%] 5 min steps

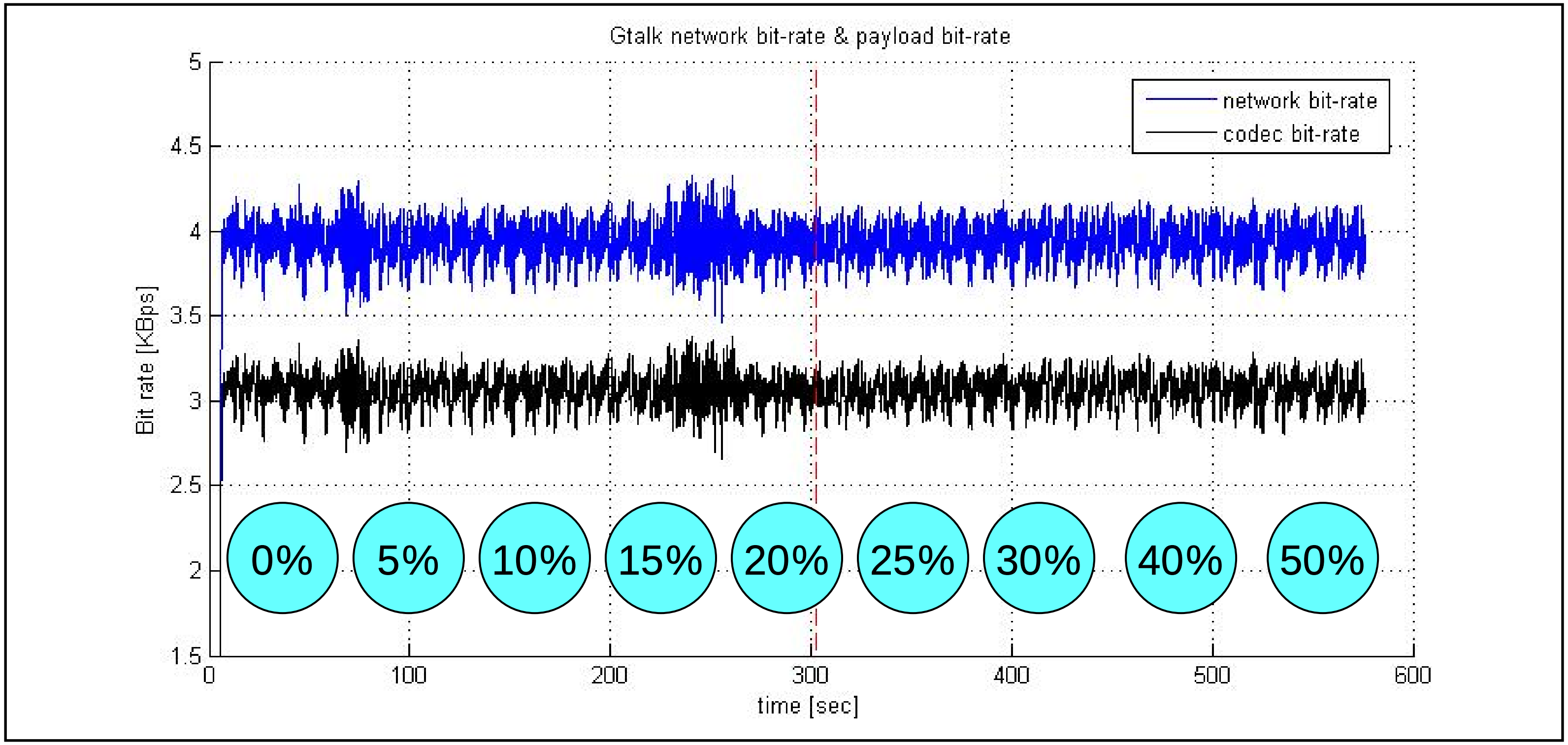


- Lo

Very slow loss adaptation due to
- Small step sizes to increase/decrease redundancy degrees
- Infrequent feedbacks
- Can only track loss-rate changes of 1% (in 2 seconds)

Google Talk: No adaptation to Loss-Rate Changes

- No change in packet period and payload size with loss-rate changes
- No network feedback



Proposed iLBC Prototype

- 30-ms packet period
- Packets are sequenced
- 1-way to 4-way redundancy decision made at receiver
  - Relayed with each reverse-path UDP speech packet
  - Extra 90ms delay for 4-way concealment in the most conservative case
  - Packets late over 60ms are discarded
- Near-term improvement opportunities
  - Decide on redundancy rate based on loss rate, UCFLR (unconcealable frame loss rate), jitter-buffer size
  - Relay decision when condition changes (repeated to assure reception)
  - Adaptively change play-out schedule to minimize delay with a prescribed level of UCFLR

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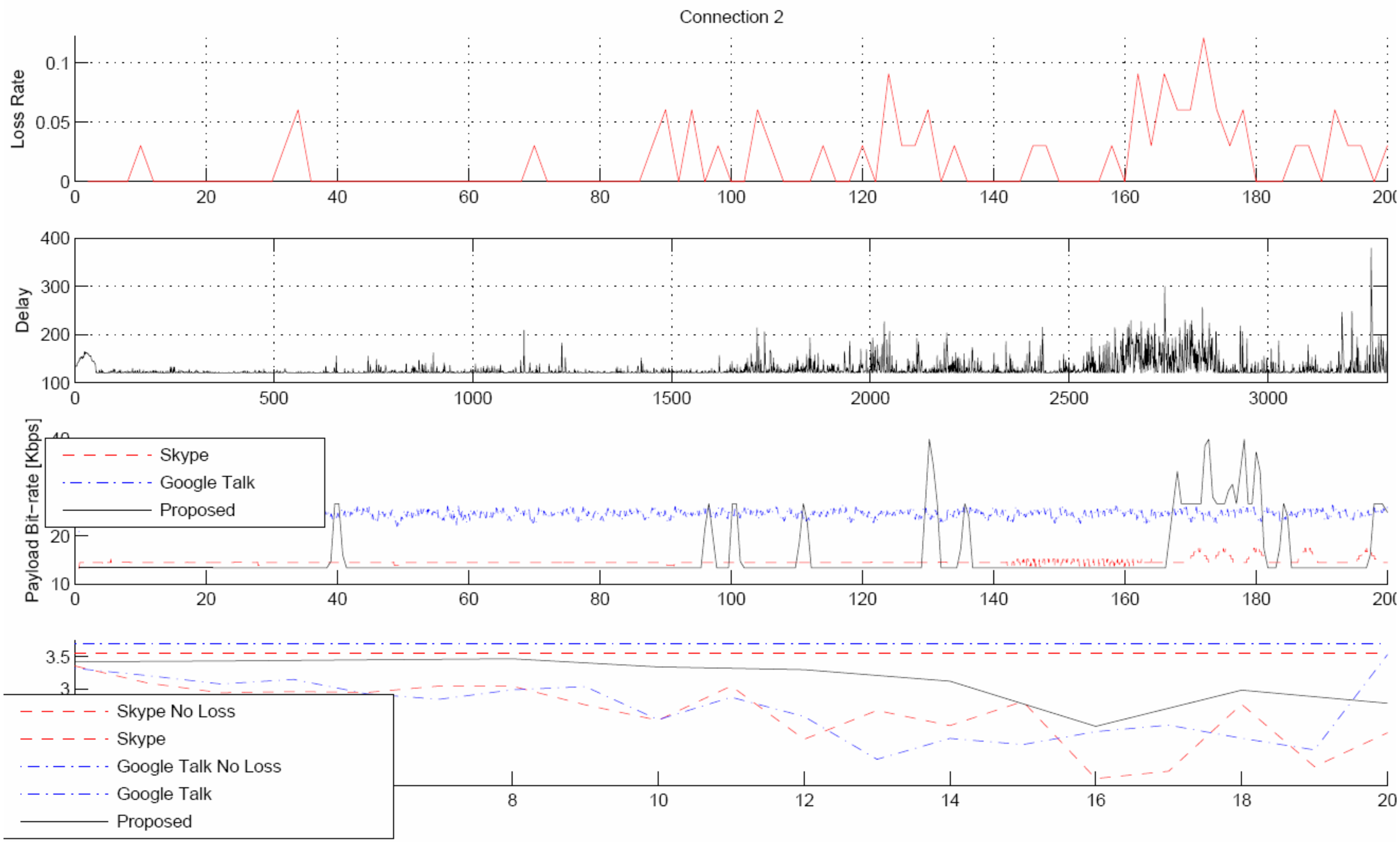
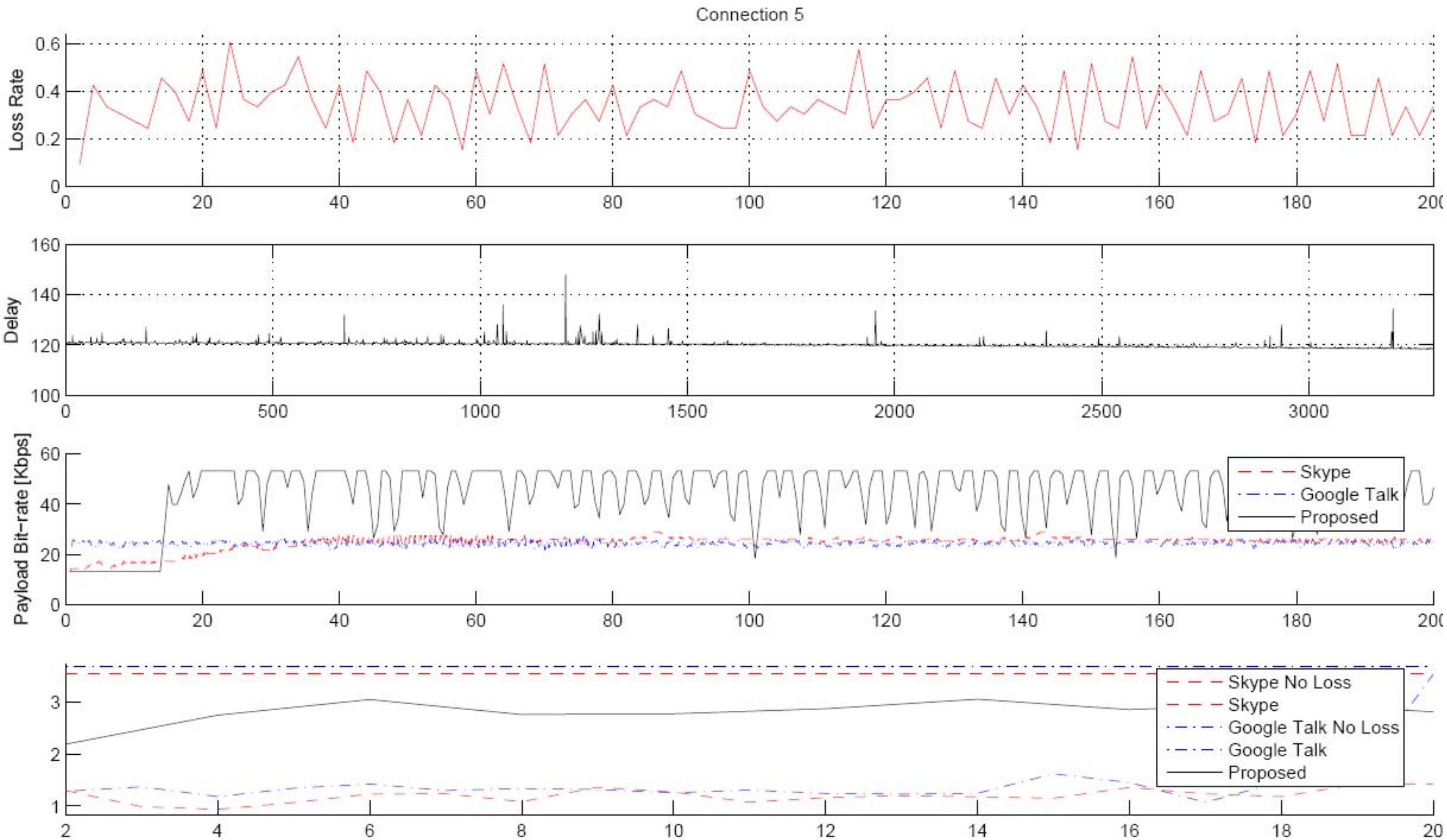
Internet Traces (#2): Low Loss Rate



Internet Traces (#5): High Loss Rate



Conclusions

- Internet delays and losses are diverse
- Skype and Google Talk
  - Too conservative loss adaptation and feedback control
  - Ineffective play-out scheduling
- Proposed prototype
  - More aggressive frame rate and feedbacks
  - 1-4 ways of redundancy