

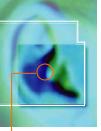
Playout Controller – QoS at the IP Edge Points

For networks with packet loss and jitter Henrik Åström

GLOBAL IP SOUND

Speech Processing, Transmission and Quality Aspects (STQ) Workshop

Feb 2003



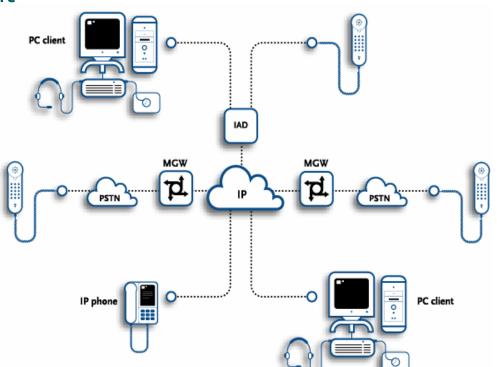
Outline

- VoIP Quality Issues
- Traditional Algorithms
- NetEQ Playout Controller
 - Speech Quality
 - Delay
 - Total Quality (with E-model)
- Conclusion



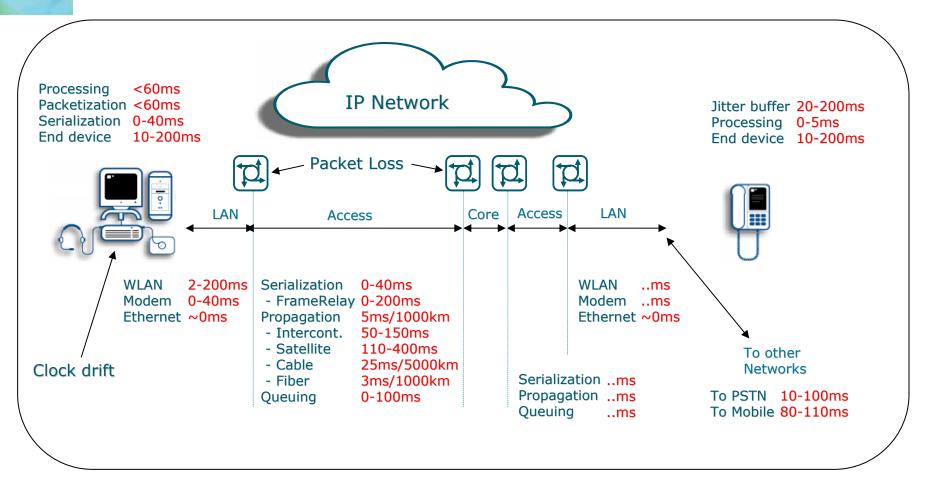
VoIP and Quality

- VoIP flexibility, cost savings and new features
- Quality issues delay, jitter, packet loss, clock drift
- Efficient algorithms ensure the speech quality in VoIP
- Overprovisioning and tight network management expensive and inflexible
- Edge point Playout Controller Solution to lots of the Problems



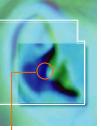
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Delay & Packet Loss Sources



The packets can be lost due to queing effects at the access points and in the routers. Packet loss also occurs when there are bit errors during the transmission (mostly for WLAN and satellite transmissions)

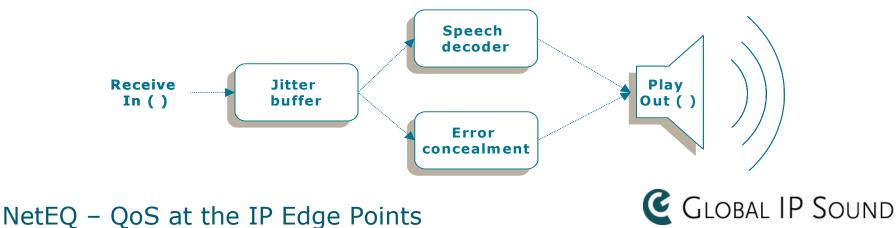


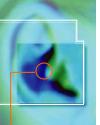


Traditional Jitter Buffer & PLC Approach

Traditionally - Jitter Buffer and PLC, two separate units

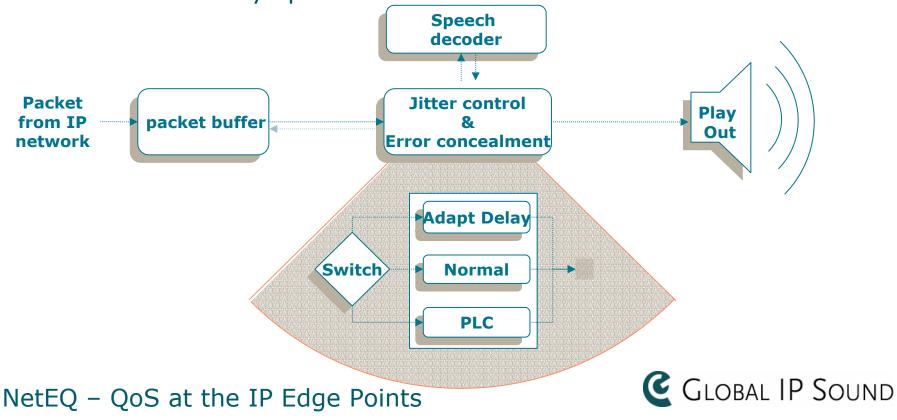
- Jitter Buffer
 - Fixed simple, high delay, not flexible
 - Adaptive lower delay, each adaptation gives distortions
- Packet Loss Concealment
 - Zero Stuffing bad quality
 - Packet Repetition up to 2-3% packet loss
 - G.711 Appendix I better quality
 - Built-in PLC codec dependent

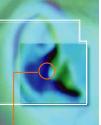




NetEQ Approach

- Playout Controller Combined Delay Adaptation and PLC
- Integrated Design Cooptimizations
- Signal Processing techniques enables good quality
- Better performance, Delay and Packet Loss Concealment
- Works with any speech codec





NetEQ vs Traditional Methods

Traditional Algorithms

- 1. Operation @ Packet level (10-60 ms)
- 2. Delay Modified by Adding/Dropping Packets
- 3. Error Concealment on Packets
- 4. Clock Drift/Skew can cause Delay and Quality Problems

Operation on Decoded Speech Resolution (1-2 ms)

NetEQ Playout Controller

Signal Processing to Increase/Decrease Delay

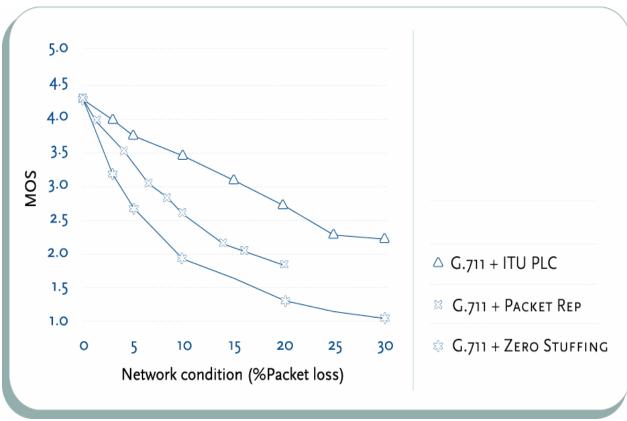
Error Concealment on Speech/Audio

Automatic and Instant Adjustment to Clock Drift/Skew



Voice Quality During Packet Loss

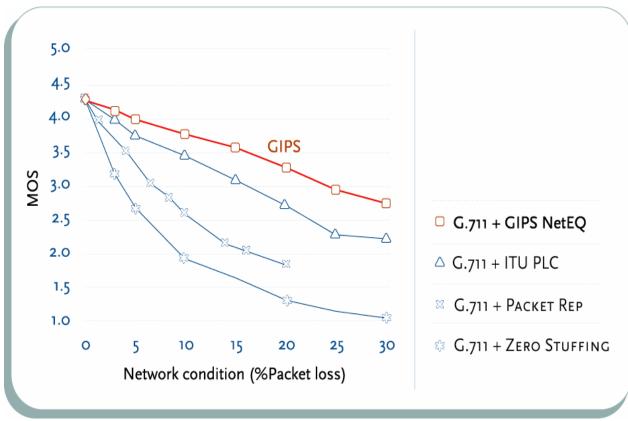
Results of Listening Test at COMSAT





Voice Quality During Packet Loss

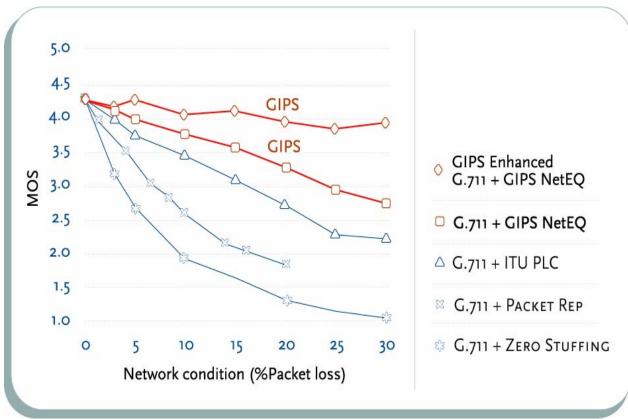
Results of Listening Test at COMSAT





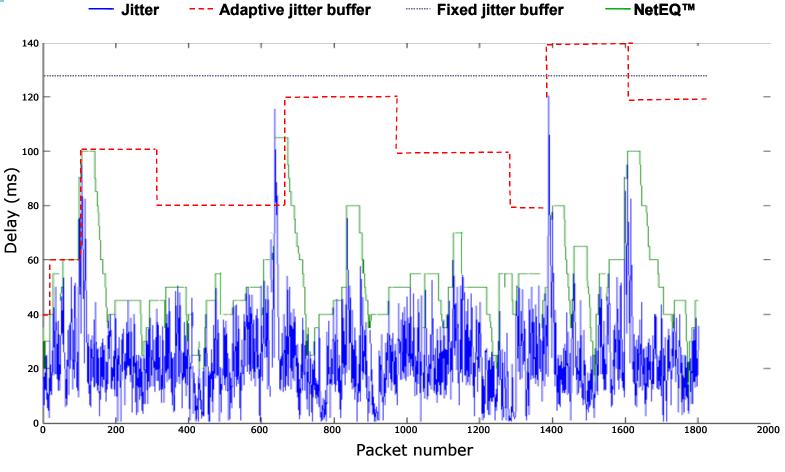
Voice Quality During Packet Loss

Results of Listening Test at COMSAT





Delay Performance



Fixed one-way delay 80 ms.

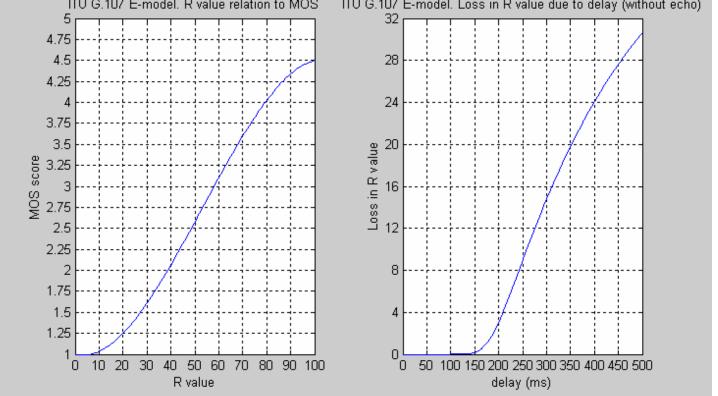
NetEQ's Delay ~150ms, Adaptive Jitter Buffer ~190 ms

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ITU E-Model

Delay also affects the quality. This is not shown in a MOS test

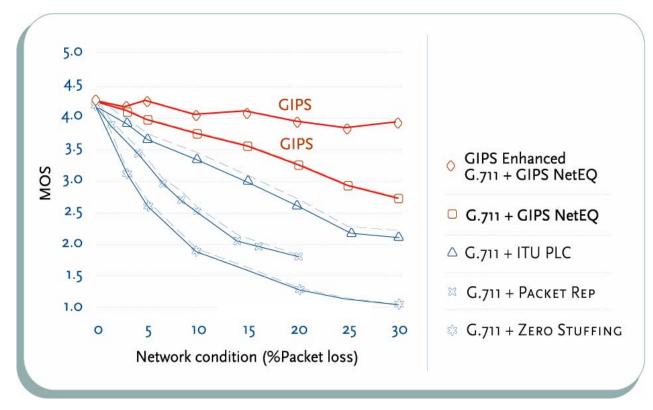
R = Ro - Is - Id - Ie,eff + A ITU G.107 E-model. R value relation to MOS ITU G.107 E-model. Loss in R value due to delay (without echo)



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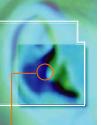
Delay Adjusted MOS Score

- The E-model + normal MOS chart give delay modified MOS
- NetEQ has lower delay, this keeps the total quality higher



Using values from jitter chart. NetEQ has average 150 ms delay, standard adaptive jitter buffer has 190 ms delay





Summary

- Important Low Delay and good PLC
- Advanced Playout Controller Higher Quality and Better Bandwidth Utilization
- Demo

