Docket #: S01-088

Adaptive Playout Scheduling for VoIP and Multimedia over IP

Stanford researchers developed and patented a receiver-based playout scheduling scheme in conjunction with improved playout algorithms to reduce packet buffering delay and packet loss for VoIP communication. Quality of service is a considerable obstacle to real-time voice communication and media streaming over the Internet. Network delay jitter, excessive delay, and packet loss contribute to impaired Voice over Internet Protocol (VoIP) quality. Stanford researchers have developed and patented a method that improves the quality of service for Internet telephony and multimedia over IP. This novel method estimates network delay from past statistics resulting in adaptively adjusted playout of voice packets that does not impair audio quality.

Applications

- Voice over Internet Protocol (VoIP)
- Streaming media over IP - video, music, and multimedia

Advantages

- Improves conversation quality in VoIP & Internet based video conferencing
- Reduces end-to-end voice/audio/video latency
- Reduces media packet loss

Publications

- Liang, Yi J., Nikolaus Farber, and Bernd Girod. "Adaptive playout scheduling and loss concealment for voice communication over IP networks." Multimedia, IEEE

Transactions on 5, no. 4 (2003): 532-543.

Patents

- Issued: 7,324,444 (USA)

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