Buffer Management Scheme to Transmit Real-time Audio in IP Networks

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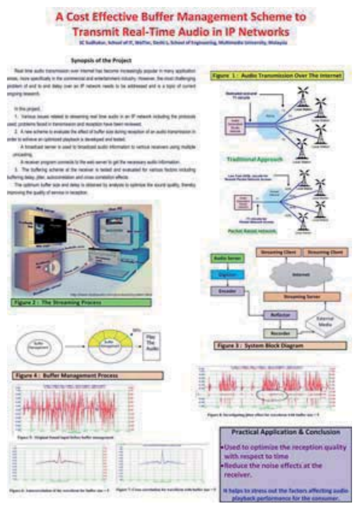

Abstract
Real time audio transmission over the internet has become increasingly popular in many application areas, in particular in the commercial and entertainment sectors. However, the most challenging problem of end-to-end delay over an IP network needs to be addressed and is a topic of current ongoing research. In this work various issues related to streaming real time audio in an IP network including the protocols used, problems faced in transmission and reception have been addressed. A new scheme to evaluate the effect of buffer size during reception of an audio transmission in order to achieve an optimized playback is developed and tested. This method can be used to optimize the reception quality with respect to time and provide a way to reduce the noise effects at the receiver. It helps to stress out the factors affecting audio playback performance for the consumer.

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Keywords
Voice over Internet Protocol (VoIP), Real Time Audio, Streaming, IP Networks, Buffer Management.

Introduction
When we consider best effort delivery for data transmission, several problems that need to be addressed are referred below (Kenneth L, 2004):

- End-to-End delay
- Packet Jitter
- Throughput
- Packet Loss

Experimental Tests
Signal Quality
We consider two scenarios in our subjective tests: transmission of real time audio with buffer size set at 5 and at 10. The variation lies between factors 0-1, shows that change in buffer size does not vary the quality of reception. Similar results were obtained while changing the buffer size to other values.
Buffering Delay
When the signal transmitted is received by the client, the signal is buffered before being played by the client

application. Change in the buffer size at the receiver application varied the delay in reception

Jitter

The inter-packet arrival time between each buffering process defined as the jitter (Karlson, 1996), occurs at every interval when the buffer algorithm at the receiver starts to play the first packet and buffers the second. For instance, it is seen that when the buffer size is set to 5, jitter occurs at every 5 seconds at the client application. When a sample file of approximately 40 seconds is played, jitter occurs 7-8 times approximately during the playback. Various tests for different buffer sizes showed that as the buffer size increases, the jitter at the reception is minimized. By setting the play-out buffer, at various sizes it is envisaged that jitter effects will be minimized. A large buffer allows significant jitter to be compensated for, but it implies a large delay, which is to be avoided for real time multimedia data.

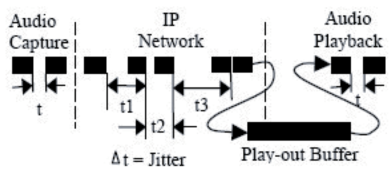

Figure 1: Relationship between play-out buffer usage and Jitter Inter-arrival time

Results

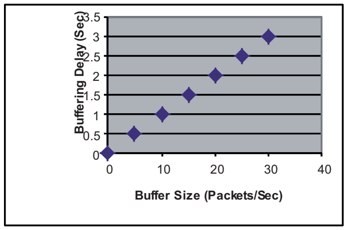

Figure 2: Buffer Size and Buffering Delay

Discussion and Conclusions

Although work exists for telephony related systems, this work presents results based on IP or packet switched networks. Studies indicate that the most crucial phenomena of buffering as applied to signals in an IP based network lies in the delay of the received signal and the jitter caused during reception. Correlation of the signal spectrum indicates that there is negligible degradation in signal quality and does not affect the quality of service. Delay increases proportionally with respect to variation in buffer size. However, delay in reception is accepted by the receiver, as it involves many criteria like network speed, bandwidth, system performance and application performance. The effect of jitter caused due to break in communication which could be populated by noise at the receiver, is a major consideration in determining the appropriate buffer size for transmission. It has been observed that increasing the buffer size decreases the jitter. However, a large buffer size implies a large delay which is to be avoided for real time multimedia data. Therefore, applications that allow variation of the buffer size dynamically to a optimum value would highly improve on the signal quality at the receiver, dependent on the bandwidth of the network, traffic and system performance. Work at this end is in progress to identify a tool that would optimize the buffer size of a client to compensate for jitter, and reduce the delay in reception, thereby increasing the quality of service.

Bibliography

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