THE IMPLEMENTATION AND ANALYSIS OF INTERACTIVE MULTIMEDIA MOBILE SERVICES

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Abstract

In this paper we examined the QoS parameters of multimedia streaming systems over mobile networks. We used a network emulation environment for monitoring, testing and examining the IP based streaming protocols.

Introduction

In the recent years we were witnesses of a rapid development of the world of mobiles. New, transmission technologies (GPRS, EDGE) and up-to-date devices appeared on the market, thus it was possible to introduce those services which have only been used through wired environment, like the IP-based video-conference, the real-time or on Demand sound and motion broadcast. The appearance of WAP and Java™ technologies on mobile devices encouraged the interactive operation and the implementation of user-friendly applications.

The need for mobile multimedia on the market is at present not very high for service providers to open up for this however there is already demand for development in this field Therefore, throughout our research area, we have worked on the implementation of a system, where we have examined the possible implementation and the operation of the multimedia content provider

The system

The basis of the completed system is a WEB-server, which ensures the availability for the users in the form of WML content: registration, check log-in, the easy availability of the content (picture, sound, movie), value-added services as well as user rights and limitations. The users can connect to the system with the browser of their mobile terminal and log into a private environment. There they are able to upload pictures and videos made with their device and share with other users. With the possession of the application rights, they have the potential to reach and inspect the content provided by the system like TV-channels, weather forecast, news, music and video files.

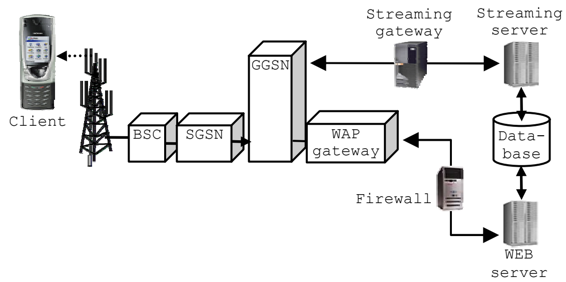

Figure 1.: Streaming multimedia system

We applied the streaming technology for the transfer of sound and motion pictures which we have examined from different angles [5] throughout the construction of our system (Helix, QuickTime, WindowsMedia, PacketVideo). We found Helix's system [3],[4] the most suitable, which provides much in its services: mirror servers, media gateway and proxy support, management of access, wide range in player programs and resource producing.

Streaming protocols

Today's IP-based streaming procedures mainly apply Real-time Transfer Protocols (RTP) and Real-Time Transfer Control

Protocols (RTCP) for the transmission of media stream. The protocol, apart from its transferring task, it transfers the QoS parameters which were measured by the application of the stream and ensures feedback potential for the best application to the momentary status of the transfer layer. And also supports the multicast and the usage of gateways.

Real-time Transport Protocol (RTP). The idea behind RTP [2] is that certain data needs to be delivered from a server to a client in a real time manner. RTP is an application layer component that utilizes UDP as transport mechanism and an RTP packet consists of sequence numbers, timestamps, and payload. RTP enables a client application to monitor the packet losses, and to “re-order” those packets that arrive out of order at the client. But RTP does not address resource reservation and does not guarantee Quality-of-Service for real-time services.

Real-time Transport Control Protocol (RTCP). RTCP is a sub-component to RTP that is used to control performance information between server and client. This information could be such as the percentage of RTP packet loss during a video session, which is crucial to managing the quality and throughput of the video data from the server. Both RTCP and RTP are designed to be independent of the underlying transport and network layers.

Real-Time Streaming Protocol (RTSP). RTSP [7] is a session-oriented protocol that is transported over TCP between server and client. The purpose of RTSP is to provide a language for communicating standard video-on-demand requests. RTSP establishes and controls either a single or several time-synchronized streams of continuous media such as audio and video. It does not deliver the continuous media stream itself, although interleaving of the media stream with the control stream is possible. In other words, RTSP acts as a “network remote control” for multimedia servers. Such control actions include pause/resume, repositioning of playback, fast forward and rewind.

There is no notion of an RTSP connection; instead a server maintains a session labelled by an identifier. During a RTSP session, an RTSP client may open and close many reliable transport connections to the server to issue RTSP requests. Alternatively, it may use a connectionless transport protocol such as UDP. Consequently, RTSP does not define how audio and video are encapsulated in packets for transmission; instead this is defined via RTP.

In a wired network bit errors are very rare and network congestion is the most likely source of packet loss [7]. But the way TCP provides reliable end-to-end service in the Internet can result in problems when TCP/IP is run over wireless links. Error recovery in the Internet is typically handled at the transport layer by TCP, with IP providing a basic unreliable service at the Internet layer. This allow applications that not require reliable service to use another end-to-end protocol such as the UDP. However, research indicates that link layer error recovery schemes over wireless Internet links can improve the performance of higher layer end-to-end protocols.

The link layer approach to error recovery is both potentially faster than end-to-end recovery, and adaptable to the wireless link characteristics. The approach is to handle wireless link errors at the link layer by implementing a protocol that hides errors from the higher layers.

Basic Problems in Video Streaming

In the first place we looked for the development of the determination and the measurement methods of those QoS parameters with which the IP-based network becomes measurable for the multimedia traffic. Video streaming over the Internet [6] is difficult because the Internet only offers best effort service. That is, it provides no guarantees on bandwidth, delay jitter, or loss rate [1]. Specifically, these characteristics are unknown and dynamic.

Bandwidth. The bandwidth available between two points in the Internet is generally unknown and time-varying. If the sender transmits faster than the available bandwidth then congestion occurs, packets are lost, and there is a severe drop in video quality. If the sender transmits slower than the available bandwidth then the receiver produces sub-optimal video quality. The goal to overcome the bandwidth problem is to estimate the available bandwidth and than match the transmitted video bit rate to the available bandwidth. Additional considerations that make the bandwidth problem very challenging include accurately estimating the available bandwidth, matching the pre-encoded video to the estimated channel bandwidth, transmitting at a rate that is fair to other concurrent flows in the Internet, and solving this problem in a multicast situation where a single sender streams data to multiple receivers where each may have a different available bandwidth.

Delay (jitter). The end-to-end delay that a packet experiences may fluctuate from packet to packet. This variation in end-to-end delay is referred to as the delay jitter. Delay jitter is a problem because the receiver must receive/decode/display frames at a constant rate, and any late frames resulting from the delay jitter can produce problems in the reconstructed video, e.g. jerks in the video. This problem is typically addressed by including a play-out buffer at the receiver. While the play-out buffer can compensate for the delay jitter, it also introduces additional delay.

Packet losses. The third fundamental problem is losses. A number of different types of losses may occur, depending on the particular network under consideration. For example, wired packet networks such as the Internet are afflicted by packet loss, where an entire packet is erased (lost). On the other hand, wireless channels are typically afflicted by bit errors or burst errors. Losses can have a very destructive effect on the reconstructed video quality. To combat the effect of losses, a video streaming system is designed with error control. Approaches for error control can be roughly grouped into four classes: (1) forward error correction (FEC), (2) retransmissions, (3) error concealment, and (4) error-resilient video coding.

Examinations

In wired environment the quality of streaming multimedia depends on temporary network traffic. The effects on transmission become from QoS parameters of network layer and not from MAC or physical layers – these may only influence the bandwidth but not in remarkable way. In mobile networks the situation changes and we have a time-variable physical layer which influences much more the quality of applications. So in mobile environments the quality of streaming applications depends a lot on properties of physical layer which constrains the QoS parameters ensured by network layer.

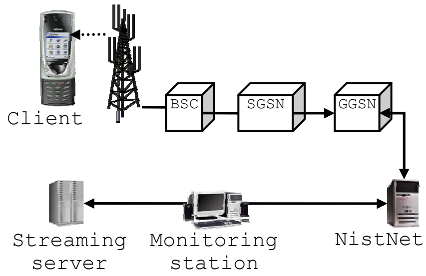

Figure 2.: Examination system

We made two kind of measurements [8]. First of all we examined how is the GPRS link layer influencing the IP.

- The network connection. What kind of bandwidth is available using GPRS? How is the channel coding influence the transmission?
- Network parameters. Delay, jitter and packet loss.

Next step was to examine how is the IP traffic influences the QoS of streaming applications.

- The bandwidth. How manages the streaming system the network bandwidth and how is this influencing the buffer of data.
- Rate control. How handles the rate control the fluctuation of network throughput?
- Packet loss. How solves the system the problem of losing packets?
- Delay jitter. How manages the system the delay jitter?

Some measurement results:

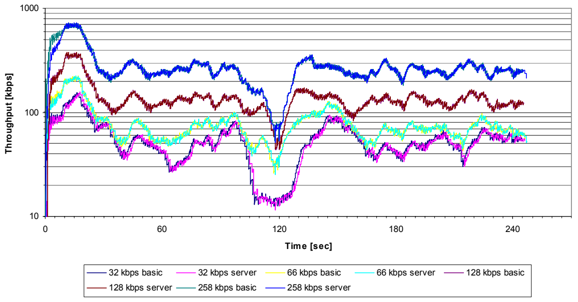

Figure 3.: Basic and server optimized streaming throughput

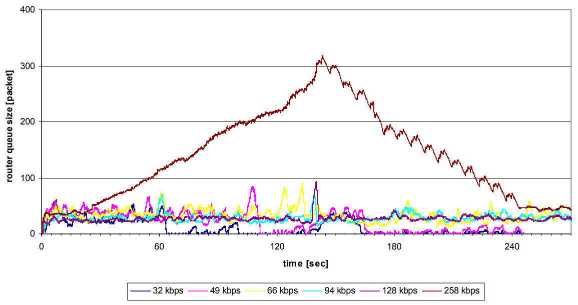

Figure 4.: Router queue size with different target bit rate

We are intended to examine also the MPEG-4 video coding scheme too.

- Challenges in MPEG-4. In what way assists this video standard the transmission over mobile networks?
- Error resilience tools. How is this method helping the coded video replacing into regular sized packets?
- Adaptive intra refresh. How is this method preventing the propagation errors?

Conclusions

Furthermore, we are intending to examine the integration of mobile providers within the network and the field of communication across firewalls. Unfortunately, at present all mobile providers in Hungary make network over GPRS only available through firewalls. We indicated the compressed parameter measurement of the given bandwidth connections and the given media types (sound, slow motion picture, film) as an important research field. Additionally, we feel the need to build a model of GPRS and EDGE radio transfer channel, as well as the simulation for getting a clear picture about the special emerging characteristics throughout the transfer affecting the media stream.

Up to this point, with our achieved results, we can state, that there are no technological barriers for the expansion of mobile multimedia and we hope, it will soon be available adjusting itself to the market demands in Hungary too.

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