Multimedia Transmission over Mobile Networks

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Abstract: **In this paper we first present the main theories about multimedia transmissions and than we examine some 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.**

1. 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 JavaTM 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

2. 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.

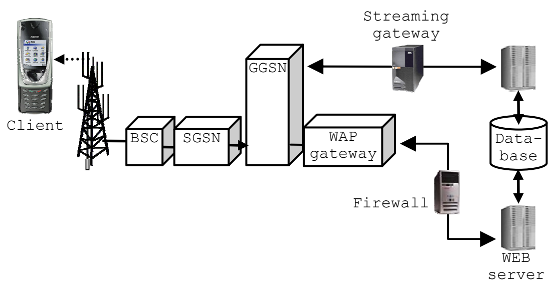

Figure 1 – Streaming Multimedia System

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.

3. 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.

3.1 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.

3.2 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.

3.3 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.

4. 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.

4.1 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.

4.2 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.

4.3 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.

5. 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.

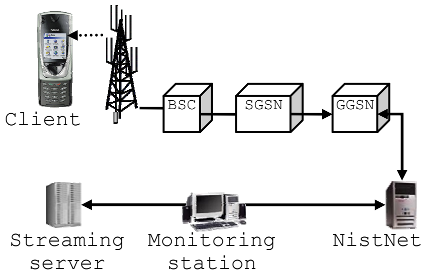

Figure 2 – System model

During our research we developed a simulation environment using NIST Net Network Emulation Tool which is suitable for emulating IP-based networks. Working in this environment we were able to simulate several IP-based connections with variable qualities.

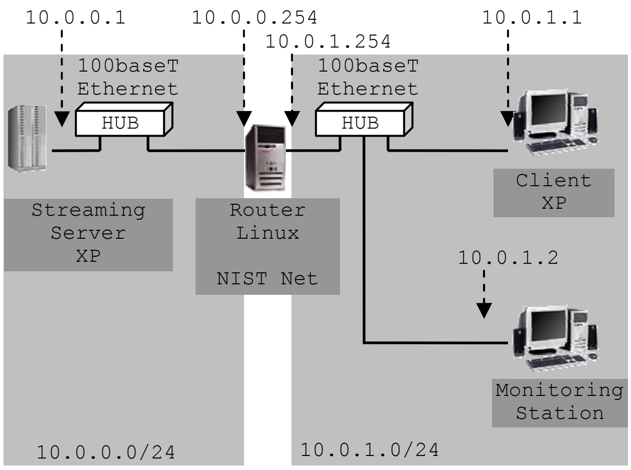

Figure 3 – Simulation environment

- ***The effect of restricted bandwidth***.

First of all, we examined what is happening with an established bit rate flow in case if it is transmitted in a lower bit rate channel. The parameters for this stream were 128 kbps, 12 frame/sec and 1/100 I-frame. Figure 4 shows the number of packets instantaneous in router queue in "Basic" mode.

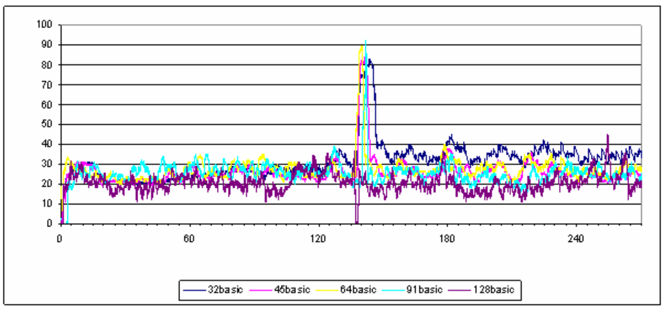

Figure 4 – Router queue size with temporary waiting packets using Basic optimized streaming throughput

In the beginning of transmission the server sends the packets at the nominal code rate, but this causes congestion and the number of waiting packets in the router increases. After a short time (5-10sec), the server realizes that the delay is getting higher and than reduces the code rate fitting to a lower bandwidth. This can reduce the increasing number of waiting packets even to a nearly constant rate. As the simulation result shows, the lower bandwidth at the router can cause more packet congestion therefore during the time the delay will getting higher.

The thorn near the 140 sec is caused by enrolling some still frames in media stream. The explanation of this comes later in this chapter.

The lower bandwidth causes that the system can transmit fewer media informations, which has also effects on the quality of pictures shown in Figure 5.

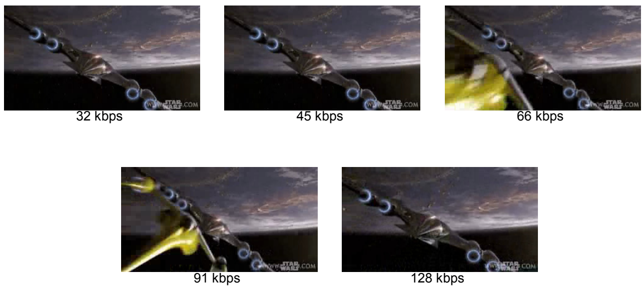

Figure 5 – Media transmission over narrow networks

As we can see, the main effects of lower bandwidth are the motion retardation and quality reduction of a video stream.

- ***The network throughput***

In first step we examined the momentary throughput. The parameters for multimedia streams were: 32, 66, 128, 258 kbps, 12 frames/sec, 1/100 I-frame, "Basic" and "Server" optimized data transmission.

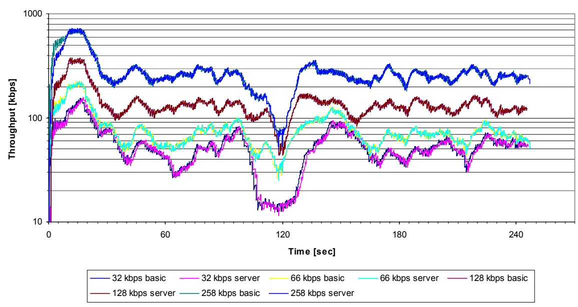

Figure 6 – Basic and server optimized streaming throughput

The value of throughput fluctuates near the nominal coding rate and there are no differences between "basic" and "server" optimized streaming control. In the first 40 seconds it can be on higher bit rate than the nominal, this is caused by the delayed playing. The player program at client uses a buffer for compensating the network fluctuation. During a multimedia streaming process, the server fills up this buffer first and just after that begins the play. At this time there is no need for synchronizing the transmission, so first the server can use the entire available bandwidth for transmitting the data. The size of client buffer is typically about 20 seconds, so after filling this buffer the playing should start and the server must synchronize the multimedia streaming to prevent the overflow in client buffer. This can be established with reducing the bit rate (between 5th and 30th second).

In this multimedia stream we included some still (non moving) pictures for testing reasons (between 120th and 150th seconds). Because of the 20 sec buffer delay, the timestamp of each packet is with 20 seconds more at client side than at the sender side. When the player arrives to the 100th second, the server sends the picture which belong to the 120th second. The coding mechanism realizes that there is no difference between the following frames, converts the data into smaller stream and transmits smaller data packages through the network. This causes the setback of throughput which can be seen in Figure 3 (between 100th and 140th seconds).

During the transmission and playing the still images the buffer will be near empty and the delay between server and client is getting smaller. In this case, in the 140th second the server needs to send the frames for 150th second but this time those packages have big size again. But the transmission mechanism sets back in normal way in a few seconds.

- ***The effect of variable bandwidth***

The third examination was made to find out how is reacting the streaming system to a quick available bandwidth reduction. Figure 7shows the momentary throughput and number of packets in router's buffer. The parameters for multimedia stream is as before: 128 kbps, 12 frames/sec, 1/100 I frame. The black broken line shows the change of available bandwidth size.

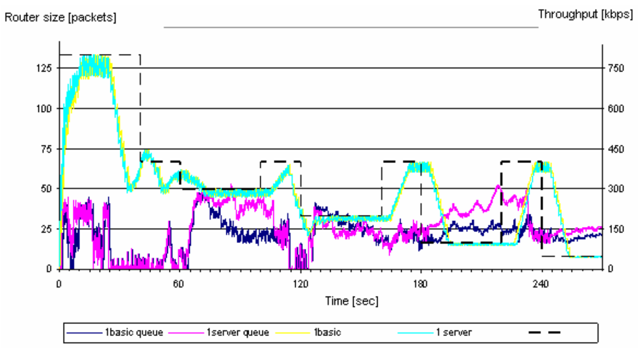

Figure 7 – Effect of quick bandwidth change

In the first 60 seconds the bandwidth remains above the nominal rate and shows that the number of waiting packets is getting stable on a low value. But when the available bandwidth falls under the nominal coding rate the number of waiting packet is increasing. After the intervention of the server this is getting stable on a little bit higher value than before however the overall delay increases too. When the bandwidth rises again above the nominal coding rate (near 120th second) , the buffer will be empty and the delay decreases too a little. As we can see decreasing the available bandwidth means that the number of packets in buffer will be increasing and this may cause higher overall delay in transmission.

6. CONCLUSION

In this paper we developed a system based on streaming technology and we presented the possibilities of multimedia transmissions. We developed also an emulation environment where we could examine the radio access and the IP protocol's main effects on streaming technology.

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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