Adaptive Transmission of Data over the Internet

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***Abstract—* Internet is a heterogeneous network environment and the network resources that are available to real time applications can be modified very quickly. Today, the underlying infrastructure of the Internet does not sufficiently support quality of service (QoS) guarantees. The new technologies, which are used for the implementation of networks, provide capabilities to support QoS in one network domain but it is not easy to implement QoS among various network domains, in order to provide end-to-end QoS to the user. In addition, some researchers believe that the cost for providing end-to-end QoS is too big, and it is better to invest on careful network design and careful network monitoring, in order to identify and upgrade the congested network links [4]**

Real time applications must have the capability to adapt their operation to network changes. In order to add adaptation characteristics to real time applications, we can use techniques both at the network and application layers

Keywords- QoS.

I. INTRODUCTION

Adaptive real time applications have the capability to transmit multimedia data over heterogeneous networks and adapt media transmission to network changes. In order to implement an adaptive multimedia transmission application, mechanisms to monitor the network conditions, and mechanisms to adapt the transmission of the data to the network changes must be implemented.

In this paper, we concentrate on the architecture of an adaptive real time application that has the capability to transmit multimedia data over heterogeneous networks and adapt the transmission of the multimedia data to the network changes. Moreover in this article, we concentrate on the unicast transmission of multimedia data.

II. RELATED WORK

The subject of adaptive transmission of multimedia data over networks has engaged researchers all over the world. During the design and the implementation of an adaptive application special attention must be paid to the following critical modules:

- The module, which is responsible for the transmission of the multimedia data
- The module, which is responsible for monitoring the network conditions and determines the change to the network conditions
- The module, which is responsible for the adaptation of the multimedia data to the network changes
- The module, which is responsible for handling the transmission errors during the transmission of the multimedia data. A common approach for the implementation of adaptive applications is the use of UDP for the transmission of the multimedia data and the use of TCP

III. PROPOSED ARCHITECTURE OF STREAMING APPLICATION

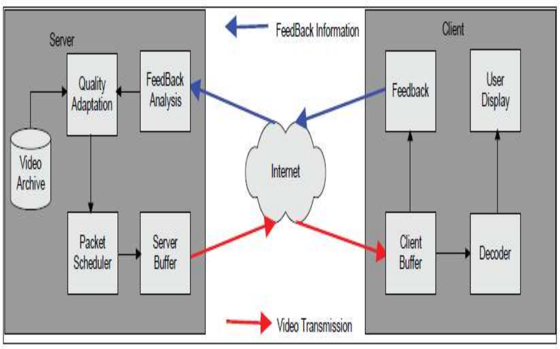
A. Server side modules are:

This section presents a typical architecture for an adaptive streaming application, based on the client server model. Figure 1 displays the architecture of such an adaptive streaming application. The server of the adaptive streaming architecture consists of the following modules:

• *Video archive:*

Video archive consists of a set of hard disks in which the video files are stored. The adaptive streaming application may support various video formats (for example MPEG, JPEG, H.263, etc.). It is possible for one video file to be stored in the video archive in more than one format in order to serve different target user groups. For example, it is possible to store the same video in MPEG format in order to serve the users of the local area network

Figure 1. System architecture



• ***Feedback analysis:***

This module is responsible for the analysis of feedback information from the network. The role of this module is to determine the network condition mainly based on packet loss rate and delay jitter information, which are provided by RTCP receiver reports. After the examination of network condition, the feedback analysis module informs the quality adaptation module, in order to adapt the transmission of the video to current network conditions.

• ***Quality adaptation:***

It is responsible for the adaptation of the video transmission quality in order to match with the current network conditions.

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This module can be implemented using various techniques (rate shaping, layered encoding, frame dropping, etc.).

• ***Packet scheduler/Server buffer:***

This module is responsible for the encapsulation of multimedia information in the RTP packets. In addition, this module is responsible for the transmission of the RTP packets in the network. In order to smooth accidental problems to the transmission of the multimedia data from the server to the network, an output buffer is used on the server.

A. The client side modules are:

• ***Client buffer:***

The use of the buffer on the client for the implementation of streaming applications is very important. The client application stores the incoming data to the buffer before starting to present data to the user. The presentation of the multimedia data to the user starts only after the necessary amount of the data is stored in the buffer. The capacity of the client buffer depends to the delay jitter during the transmission of the multimedia data. In any case the capacity of the client buffer must be greater than the maximum delay jitter during the transmission of the data.

• ***Feedback:***

This module is responsible of monitoring the transmission quality of the data and informing the server. The monitoring of the transmission quality is based on RTCP receiver reports that the client sends to the server. RTCP receiver reports include information about the packet loss rate and the delay jitter during the transmission of the data. With the previous information, the feedback analysis module of the server determines the network's condition.

• ***Decoder:***

This module reads the data packets from the client buffer and decodes the encoded multimedia information. Depending on the packet losses and the delay during the transmission of the packets, the quality of the multimedia presentation can vary. The decoding and the presentation of the multimedia data can stop, if the appropriate amount of data does not exist in the buffer.

• ***User display:***

It is responsible for the presentation of the multimedia data to the user.

IV. TRANSMISSION OF MULTIMEDIA DATA

The transmission of the multimedia data is based on the protocols RTP/RTCP. The protocol RTP is used for thetransmission of the multimedia data from the server to the client and the client uses the RTCP protocol, in order to inform the server of the transmission quality. The RTP/RTCP protocols have been designed for the transmission of real time data like video and audio. Although the RTP/RTCP protocols were initially designed for multicast transmission, they were also used for unicast transmissions. RTP/RTCP can be used for one-way communication like video on demand or for two-way communication like videoconference. RTP/ RTCP offers a common platform for the representation of synchronisation information that real time applications needs. The RTCP protocol is the control protocol of RTP. The RTP protocol has been designed to operate in cooperation with the RTCP protocol, which provides information about the transmission quality. RTP is a protocol that offers end to end transport services with real time characteristics over packet switching networks like IP networks.

• ***QoS monitoring:***

This is one of the primary services of RTCP. RTCP provides feedback to applications about the transmission quality. RTCP uses sender reports and receiver reports, which contain useful statistical information like total transmitted packets, packet loss rate and delay jitter during the transmission of the data. This statistical information is very useful, because it can be used for the implementation of congestion control mechanisms.

• ***Source identification:***

RTCP source description packets can be used for identification of the participants in a RTP session. In addition, source description packets provide general information about the participants in a RTP session. This service of RTCP is useful for multicast conferences

V. FUTURE SCOPE

The most prominent enhancement of the adaptive real time applications is the use of multicast transmission of the multimedia data. The multicast transmission of multimedia data over the Internet has to accommodate clients with heterogeneous data reception capabilities. To accommodate heterogeneity, the server may transmit one multicast stream and determine the transmission rate that satisfies most of the clients [2], and may transmit multiple multicast streams with different transmission rates and allocate clients at each stream or may use layered encoding and transmit each layer to a different multicast stream[2]. An interesting survey of techniques for multicast multimedia data over the Internet is presented by Li, Ammar, and Paul (1999).

Single multicast stream approaches have the disadvantage that clients with a low bandwidth link will always get a high-bandwidth stream if most of the other members are connected via a high bandwidth link and the same is true the other way around. This problem can be overcome with the use of a multi-stream multicast approach. Single multicast stream approaches have the advantages of easy encoder and decoder implementation and simple protocol operation, due to the fact that during the single multicast stream approach there is no need for synchronization of clients' actions (as is required by the multiple multicast streams and layered encoding approaches).

The subject of adaptive multicast of multimedia data over networks with the use of one multicast stream has engaged researchers all over the world. During the adaptive multicast transmission of multimedia data in a single multicast stream, the server must select the transmission rate that satisfies most the clients with the current network conditions. Three approaches can be found in the literature for the implementation of the adaptation protocol in a single stream multicast mechanism: equation based [10], network feedback based, or based on a combination of the previous two approaches.

VI. CONCLUSION

Many researchers urge that due to the use of new technologies for the implementation of the networks, which offer QoS guarantees, adaptive real time application will not be used in the future. We believe that this is not true and adaptive real time applications will be used in the future for the following reasons:

• Users may not always want to pay the extra cost for a service with specific QoS guarantees when they have the capability to access a service with good adaptive behaviour.

- Some networks may never be able to provide specific QoS guarantees to the users.
- Even if the Internet eventually supports reservation mechanisms or differentiated services, it is more likely to be on per-class than per-flow basis. Thus, flows are still expected to perform congestion control within their own class.
- With the use of the differential services network model, networks can support services with QoS guarantees together with best effort services and adaptive services.

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