Chapter 20

ADAPTIVE MULTIMEDIA TRANSMISSION OVER THE INTERNET

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The Internet is a heterogeneous network environment and the network resources that are available to multimedia applications can be modified very quickly. Multimedia applications must have the capability to adapt their operation to network changes. In order to add adaptation characteristics to multimedia applications, we can use techniques both at the network and application layers. Adaptive multimedia transmission techniques have the capability to transmit multimedia data over heterogeneous networks and adapt media transmission to network changes. In order to provide adaptive multimedia transmission, mechanisms to monitor the network conditions and mechanisms to adapt the transmission of the data to the network changes must be implemented. This chapter concentrates on the architecture and design issues of adaptive multimedia techniques that provide the capability to transmit multimedia data over heterogeneous networks and adapt the transmission of the multimedia data to the network changes.

1. Introduction

The Internet is a heterogeneous network environment and the network resources that are available to realtime applications can be modified very quickly. Realtime applications must have the capability to adapt their operation to network changes. In order to add adaptation characteristics to realtime applications, we can use techniques both at the network and application layers. Adaptive realtime 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.

Today, the underlying infrastructure of the Internet does not sufficiently support quality of service (QoS) guarantees. The new technologies that 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 of providing end-to-end QoS is too heavy, and it is better to invest in careful network design and careful network monitoring, in order to identify and upgrade the congested network links [1].

In this chapter, we concentrate on the architecture of an adaptive realtime application that has the capability to transmit multimedia data over heterogeneous networks and adapt the transmission of the multimedia data to the network changes. The chapter covers both unicast and multicast transmission of multimedia data.

The heterogeneous network environment that the Internet provides to realtime applications, as well as the lack of sufficient QoS guarantees, often forces applications to employ adaptation schemes in order to work efficiently. In addition, any application that transmits data over the Internet should not adversely affect the other flows that exist, especially the transmission control protocol (TCP) flows that comprise the majority of flows. We define as *TCP friendly flow* a flow that consumes no more bandwidth than a TCP connection which is traversing the same path as that flow [2].

During multimedia transmission over the Internet, several aspects need to be considered:

- *Transmission rate adaptation*: The sender must adapt the transmission rate based on the current network conditions.
- *TCP friendliness*: During the multimedia transmission over the Internet, the multimedia flows must be TCP-friendly.
- *Scalability*: The performance of the adaptation scheme must not deteriorate with increasing numbers of receivers.
- *Heterogeneity*: The adaptation scheme needs to take into account the heterogeneity of the Internet and must aim at satisfying the requirements of a large part of the receivers if not all possible receivers.

The remainder of this chapter is structured as follows. The next section reviews the various international contributions to research on adaptive multimedia transmission over the Internet. Section 3 presents some thoughts for practitioners. Section 4 describes unicast multimedia adaptive transmission over the Internet, while Section 5 tackles multicast multimedia adaptive transmission. Section 6 considers the future trends in the field, and Section 7 draws conclusions.

2. Background

The subject of adaptive transmission of multimedia data over networks has engaged researchers all over the world for many years [1]. During the design and implementation of an adaptive application special attention must be paid to the following critical modules:

- the module responsible for the transmission of the multimedia data,
- the module responsible for monitoring the network conditions and determines the changing the network conditions,
- the module responsible for the adaptation of the multimedia data to the network changes,
- the module responsible for handling transmission errors during the transmission of the multimedia data.

A common approach for the implementation of adaptive applications is the use of the user datagram protocol (UDP) for the transmission of the multimedia data and the use of the Transmission Control Protocol (TCP) for the transmission of control information [3]. Another approach for the transmission of the multimedia data is the use of RTP over UDP [4, 5]. Most adaptive applications use (Realtime Transmission Protocol/Realtime Control Transmission Protocol (RTP/RTCP) [6]) for the transmission of multimedia data. RTP seems to be the *de facto* standard for the transmission of multimedia data.

For the implementation of the network monitoring module a common approach is to use packet loss as an indication of congestion in the network [4, 5, 7]. Another approach for monitoring the network conditions is the use of the client buffer [8]. An important factor that can be used for monitoring the network conditions, and especially for an indication of network congestion, is the delay jitter.

For the implementation of the adaptation module some common approaches are the use of rate shaping [4, 9], layered encoding, and frame dropping [8], or a combination of the above techniques [10]. The implementation of the adaptation module depends on the encoding method used. For example, to employ the frame dropping technique for the adaptation of an MPEG video stream, a selective frame dropping technique must be used since MPEG video uses inter-frame encoding and some frames contain information relative to other frames.

It is important for adaptive realtime applications to have "friendly" behavior towards the dominant transport protocols (TCP) [11].

For multimedia transmission over the Internet both unicasting and multicasting can be used. In unicast multimedia transmission a separate multimedia stream is transmitted by the sender for each of the receivers. When the number of receivers is large unicasting is not effective (in terms of the required bandwidth) and multicasting can be used. In multicast multimedia transmission a single multimedia stream serves many receivers.

When multimedia data are multicast over the Internet, receivers with heterogeneous data reception capabilities must be accommodated. To accommodate heterogeneity, the sender application may transmit one multicast stream and determine the transmission rate that best satisfies most of the receivers, it may transmit multiple multicast streams with different transmission rates and allocate receivers at each stream, or it may use layered encoding and transmit each layer to a different multicast stream. A detailed survey of adaptive video multicasting over the Internet is given in [12].

The single multicast stream approach has 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 *vice versa*. This problem can be overcome with the use of a multistream multicast approach. Single multicast stream approaches have the advantages of easy encoder and decoder implementation and simple protocol operation since during the single multicast stream approach there is no need for synchronization of receivers' actions (as is required for the multiple multicast streams and layered encoding approaches).

The methods proposed for the multicast transmission of multimedia data over the Internet can generally be divided into three main categories, depending on the number of multicast streams used:

- *The sender uses a single multicast stream for all receivers* [4]. This results in the most effective use of network resources, but gives rise to the fairness problem among the receivers, especially when the receivers have very different capabilities. The sender application must select the transmission rate that satisfies most of the receivers under the current network conditions. Three main approaches can be found in the literature: equation based [2], network feedback based [13, 14] or a combination of the two approaches [15].
- *Simulcast: The sender transmits different versions of the same video encoded with varying degrees of quality*. This results in the creation of a small number of multicast streams with different transmission rates [12, 16]. The different multicast streams carry the same video information but in each one the video is encoded with different bit rates, and even different video formats. Each receiver joins the stream that carries the video quality, in terms of transmission rate, that it is capable of receiving. The main disadvantage here is that the same multimedia information is replicated over the network, but recent research has shown that under some conditions simulcasting has a better behavior than multicast transmission of layered encoded video [17].
- *The sender uses layered encoded video*. This is video that can be reconstructed from a number of discrete data layers, the basic layer and more additional layers. The sender transmits each layer in different multicast streams [18–20]. The basic layer provides the basic quality and the quality improves with each additional layer. Receivers subscribe to one or more multicast streams depending on the available bandwidth of the network path to the source. A survey of adaptive layered video multicasting over the Internet is given in [21].

A comparison of simulcast and layered encoded approaches can be found in [5].

3. Thoughts for Practitioners

The deployment of adaptive transmission applications has the benefit that there is not need for network device changes or any extra configuration of network devices because adaptation takes place to the end points (sender and receiver applications) and not within the network. Adaptive unicast transmission applications require only the presence of an IP network, and adaptive multicast transmission applications require only the presence of an IP multicast network. Even when a part of the network used for adaptive multicasting does not support IP multicasting, the IP tunneling approach can be used to overcome this limitation (something that has been done many times in the MBONE network). This is very important when comparing approaches based on the QoS concept (either InteServ or DiffServ), where network device changes and extra configuration of network devices is needed.

Especially for the unicast adaptive transmission and the single multicast stream adaptive transmission approaches, deployment is narrowed to only the sender application end point since all the required modules for the implementation are located on the sender side alone. This means that any application which is compatible with the transmission of multimedia data through RTP sessions (for example, MBONE tools [10] can access the adaptive transmission service and benefit from its adaptive transmission characteristics.

In implementing adaptive transmission applications, Java technology can be used and, in particular, Java Media Framework (JMF) application programmable interface (API). JMF provides capabilities for importing in Java applications and applets time-based media like video and audio. JMF supports the most common audio and video formats like AIFF, AU, AVI, GSM, MIDI, MPEG, QuickTime, RMF and WAV. In addition, JMF supports the transmission of realtime data (i.e., multimedia) with the use of RTP/RTCP. Moreover, JMF has capabilities for elaboration of audio and video data during transmission over the network with the use of RTP/RTCP. All the above characteristics of JMF make it an attractive platform for the implementation of adaptive transmission mechanisms. More information about JMF can be found in Ref. 22.

The implementation of multistream adaptive multicasting (simulcast and layered encoded approaches) is based on complicated video encoding and decoding mechanisms. The current state-of-the-art technology in such video encoders/decoders is the H.264 video coding standard with the AVC [3] and the SVC [23] extensions. The H.264 AVC extension can be used for the implementation of simulcast multimedia transmission, and the H.264 SVC extension can be used for the implementation of layered encoded multimedia transmission.

4. Unicast Multimedia Adaptive Transmission Over the Internet

4.1. *The Architecture of an Adaptive Streaming Application*

This section presents a typical architecture for a unicast adaptive multimedia transmission application based on the client–server model. (Fig. 1).

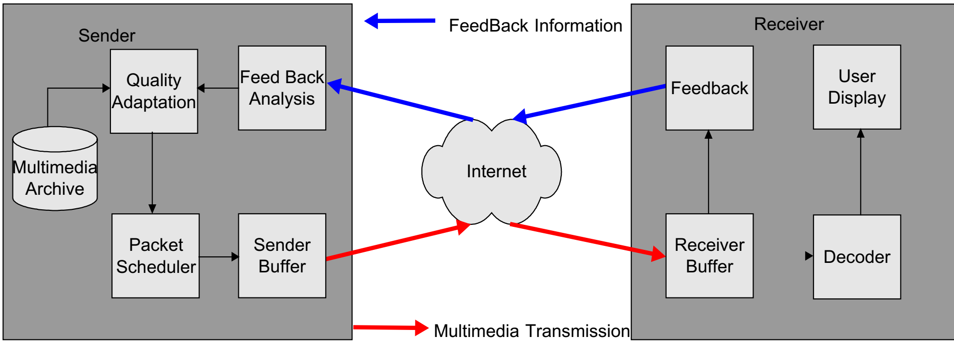

Fig. 1. System architecture.

The sender comprises the following modules:

- *Multimedia archive*: A set of hard disks in which the multimedia files are stored. The adaptive transmission application may support various multimedia formats (e.g., MPEG, JPEG, H.263). It is possible for one multimedia file to be stored in the multimedia 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 users of a local area network (who have faster network connections to the sender) and in H.263 format in order to serve distant users with slower network connections.
- *Feedback analysis*: Responsible for the analysis of feedback information from the network. The role of this module is to determine the network conditions mainly based on packet loss rate and delay jitter information, which are provided by RTCP receiver reports. After the examination of the network conditions, the feedback analysis module informs the quality adaptation module in order to adapt the transmission of the video to current conditions.
- *Quality adaptation*: Responsible for the adaptation of the multimedia transmission quality in order to match with the current network conditions. This module can be implemented using various techniques (rate shaping, layered encoding, frame dropping, etc.).
- *Packet scheduler/sender buffer*: 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 out glitches in the transmission of the multimedia data from the sender to the network, an output buffer is used on the sender.

The receiver of a unicast adaptive multimedia transmission architecture comprises the following modules:

- *Receiver buffer*: The use of a buffer on the receiver in the implementation of such applications is very important. The receiver 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 receiver buffer depends to the delay jitter during the transmission. In any case, the capacity of the receiver buffer must be greater than the maximum delay jitter during the transmission of the data (we suppose that we measure the buffer capacity and the delay jitter in the same units, for example, in seconds).

- *Feedback*: Responsible for monitoring the transmission quality of the multimedia data and informing the sender. The monitoring of the transmission quality is based on the RTCP receiver reports, which the receiver sends to the sender. RTCP receiver reports include information about the packet loss rate and the delay jitter during the transmission of the data. With the above information the feedback analysis module of the sender determines network conditions.
- *Decoder*: Reads the data packets from the receiver 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*: Responsible for the presentation of the multimedia data to the user.

In the following subsections we provide a detailed description of the most important modules in the above architecture.

4.2. *Transmission of Multimedia Data*

The *de facto* protocols currently used for the transmission of multimedia data on the Internet are RTP and RTCP. RTP is used for the transmission of the multimedia data from the sender to the receiver, and the receiver uses the RTCP protocol, to inform the sender of the transmission quality. RTCP is the control protocol of RTP.

RTP and RTCP were designed for the transmission of multimedia data like video and audio. Although they were initially designed for multicast transmission, they have also been used for unicast transmissions. RTP/RTCP can be used for one-way communication like video on demand or for two-way communication like videoconferencing. RTP/RTCP offers a common platform for the representation of the synchronization information that realtime applications need.

RTP is a protocol that offers end-to-end transport services with realtime characteristics over packet-switching networks like IP networks. RTP packet headers include information about the payload type of the data, numbering of the packets and time stamping information.

RTCP offers the following services to applications:

- *QoS monitoring*: Providing — feedback to applications about transmission quality is one of the primary services of RTCP. RTCP uses sender and receiver reports, which contain useful statistical information like total transmitted packets, packet loss rate and delay jitter. 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 an RTP session. In addition, source description packets provide general information about the participants in an RTP session. The service is useful for multicast conferences with many members.
- *Inter-media synchronization*: In realtime applications, it is common to transmit audio and video in different data streams. RTCP provides services like time stamping, which can be used for inter-media synchronization of different data streams (for example, synchronization of audio and video streams).

More information about RTP/RTCP can be found in RFC 3550 [6].

4.3. *Feedback from the Network*

The presentation quality of multimedia data depends on the packet loss rate and the delay jitter during transmission over the network. In addition, packet losses or rapid increases in delay jitter may be considered an indication of problems during transmission. In such cases, the multimedia transmission application must adapt the transmission of the data in order to avoid phenomena like network congestion. Multimedia applications have upper bounds to the packet loss rate and the delay jitter. If packet loss rate or jitter exceeds these upper bounds, transmission of realtime data cannot be continued.

Packet loss rate is defined as the fraction of the total transmitted packets that did not arrive at the receiver. Usually, the main reason for packet losses is congestion.

It is difficult to define delay jitter. Some researchers define delay jitter as the difference between the maximum and the minimum delay during the transmission of packets over a period of time. Other researchers define delay jitter as the maximum difference between the delays of the transmissions of two sequential packets over a period of time. According to RFC 3550 [6] delay jitter is defined as the mean deviation (smoothed absolute value) of the difference in packet spacing at the receiver compared to the sender for a pair of packets. This is equivalent to the difference in the “relative transit time” for the two packets. The relative transit time is the difference between a packet’s timestamp and the receiver’s clock at the time of arrival. If S_i is the timestamp from packet i and j is the time of arrival for this packet, then for two packets i and j, D is defined as

$$D(i,j) = (R_j - R_i) - (S_j - S_i) = (R_j - S_j) - (R_i - S_i). \tag{1}$$

The delay jitter is calculated continuously as each packet i arrives, using the difference D for that packet and the previous packet, according to the following formula:

$$J_i = J_{i-1} + (|D(i-1,j)| - J_{i-1})/16. \tag{2}$$

The above formula states that the new value of delay jitter depends on the previous value of the delay jitter and on a gain parameter, which gives good noise reduction.

Delay jitter occurs when sequential packets encounter different delays in the queues of the network devices. The different delays are related to the server model of each queue and the cross traffics in the transmission path.

Sometimes delay jitter occurs during the transmission of multimedia data that does not originate from the network but from the transmission host (host included delay jitter). This is because during the encoding of the multimedia data the encoder places a timestamp in each packet, which gives information about the time that the packet's information must be presented to the receiver. In addition, this timestamp is used for the calculation of the delay jitter during the transmission of the multimedia data. If a notable time passes from the encoding of the packet and transmission of the packet in the network (because the CPU of the transmitter host is busy), the calculation of the delay jitter is not valid. Host included delay jitter can lead to erroneous estimations of the network conditions.

We can conclude that delay jitter cannot by itself lead to reliable estimations of network conditions. Delay jitter has to be used in combination with other parameters like packet loss rate in order to allow reliable estimations of the network conditions to be made. Thus, the combination of packet loss rate and delay jitter can be used for reliable indications of network congestion [4].

4.4. *Quality Adaptation*

One popular approach for quality adaptation is the rate shaping technique. According to this technique, if we change some parameters of the encoding procedure, we can control the amount of the data that the video encoder produces (either increase or decrease the amount of the data) and as a result we can control the transmission rate of the multimedia data.

The implementation of rate shaping techniques depends on the video encoding used. Rate shaping techniques change one or more of the following parameters:

- *Frame rate*: The rate of the frames that are encoded by the video encoder. Decreasing the frame rate can reduce the amount of the data that the video encoder produces but will reduce the quality.
- *Quantizer*: Specifies the number of DCT coefficients that are encoded. Increasing the quantizer decreases the number of encoded coefficients and reduces image quality.
- *Movement detection threshold*: Used for inter-frame coding, where the DCT is applied to signal differences. The movement detection threshold limits the number of blocks which are detected to be "sufficiently different" from the previous frames. Increasing this threshold decreases the output rate of the encoder.

4.5. *Error Control/Packet Loss*

The packet loss rate depends on various parameters and adaptive multimedia transmission applications must adapt to changes in packet loss. Two approaches are available to reduce the effects of packet loss:

- Automatic repeat request (APQ): APQ is an active technique where the receiver asks the sender to retransmit lost packets.
- Forward error correction (FEC): FEC is a passive technique where the sender transmits redundant information. This redundant information is used by the receiver to correct errors and lost packets.

5. Multicast Multimedia Adaptive Transmission Over the Internet

5.1. *Introduction*

As we have already mentioned, for multimedia transmission over the Internet the unicast approach does not scale well. In unicast multimedia transmission a separate multimedia stream is transmitted by the sender to each of the receivers. When the number of receivers is large, the unicast transmission is not effective (in terms of the required bandwidth) and multicasting can instead be used. During multicast transmission of data, a multicast stream traverses a network node as long as at least one receiver behind that node is listening to that stream. As result, if a receiver stops listening to a multicast stream, the transmission will stop only if that receiver was the only receiver listening to that multicast stream behind that node.

During multicast transmission, receivers with heterogeneous data reception capabilities must be accommodated. To accommodate heterogeneity, the sender application may transmit one multicast stream and determine the transmission rate that best satisfies most of the receivers, it may transmit at multiple multicast streams with different transmission rates and allocate receivers at each stream, or it may use layered encoding and transmit each layer to a different multicast stream.

5.2. *Single Stream Multicasting*

In this scenario, a sender application transmits multimedia data to a group of n receivers with the use of one multicast stream. The sender application uses RTP/RTCP for the transmission of the multimedia data. Receivers receive the multimedia data and inform the sender application about the quality of the transmission with the use of RTCP receiver reports. The sender application collects the RTCP receiver reports, analyzes them and determines the transmission rate that satisfies most receivers given the current network conditions.

During a single stream adaptive multicast transmission, the sender usually runs two algorithms:

- *Feedback analysis algorithm*: Analyzes the feedback information that the receivers send to the sender application (most mechanisms use the RTCP receiver reports

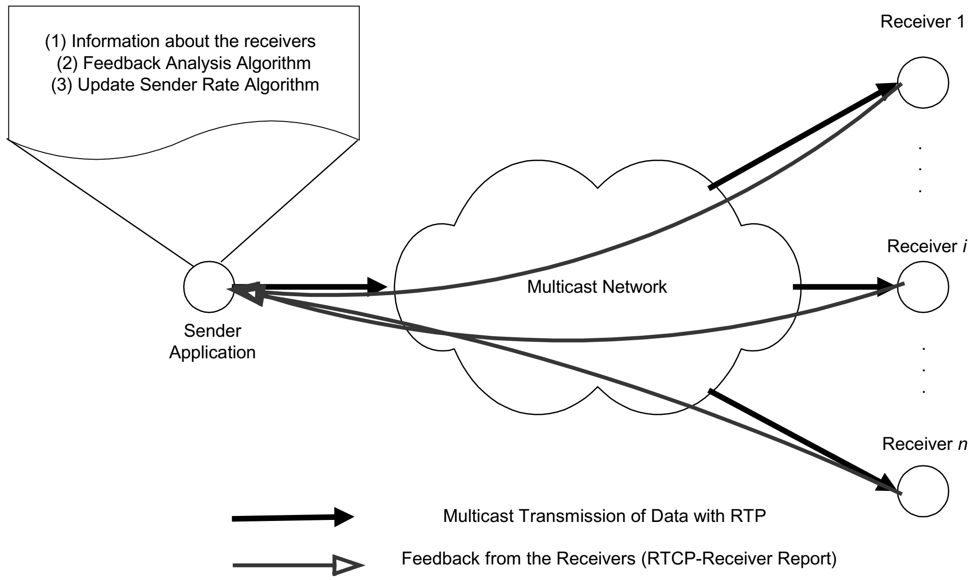

Fig. 2. Architecture of a single stream adaptive multicast transmission mechanism.

for this purpose) concerning the transmission quality of the multimedia data. Every time the sender application receives feedback from a receiver, it runs the feedback analysis algorithm in order to estimate the preferred transmission rate that will satisfy that receiver. The receiver's preferred transmission rate represents the transmission rate that this receiver would prefer if it were the only receiver receiving the multimedia data.

- *Update sender rate algorithm*: The sender application in repeated time periods estimates the transmission rate with the use of this algorithm. The estimation is aimed at increasing the satisfaction of the group of receivers for the given network conditions.

The receiver of a single stream multicast multimedia transmission behaves like the corresponding receiver of the unicast multimedia transmission and via RTCP reports provides the sender with feedback on the quality of the multimedia transmission.

5.3. *Simulcasting*

In the simulcast approach, the sender transmits different versions of the same video encoded with varying degrees of quality. This results in the creation of a small number of multicast streams with different transmission rates [4]. The different multicast streams carry the same video information but in each one the video is encoded with different bit rates, and even different video formats. Each receiver joins the stream that carries the video quality, in terms of transmission rate, that it is capable of receiving.

In such a mechanism the sender is unique and responsible for:

- creating the n different multicast streams (generally a small number of multicast streams, usually three or four are enough),
- setting each stream's bandwidth limits,
- tracking if there are any receivers that are not handled fairly, and
- providing the mechanisms to the receivers to switch streams whenever they consider that they should be in another stream closer to their capabilities.

Figure 3 shows the organisation and the architecture of the sender entity. The sender generates n different stream managers. In each stream manager, an arbitrary number of receiver managers are assigned. Each receiver manager corresponds to a unique receiver that has joined the stream controlled by this stream manager. The synchronization server is responsible for the management, synchronization and intercommunication between stream managers.

The stream manager entity is responsible for the maintenance and the monitoring of one of the n different multicast streams. Also the stream manager entity has all the intra-stream adaptation mechanisms for the adjustment of the transmission rate. The stream manager gathers the states reported by all receiver managers belonging to it at specific, fixed time intervals. It then uses an appropriate algorithm that tries to improve fairness between receivers by determining whether a lower or a higher bit rate is more appropriate. Whenever a receiver cannot be satisfied by a stream because most of the other receivers have much higher or much

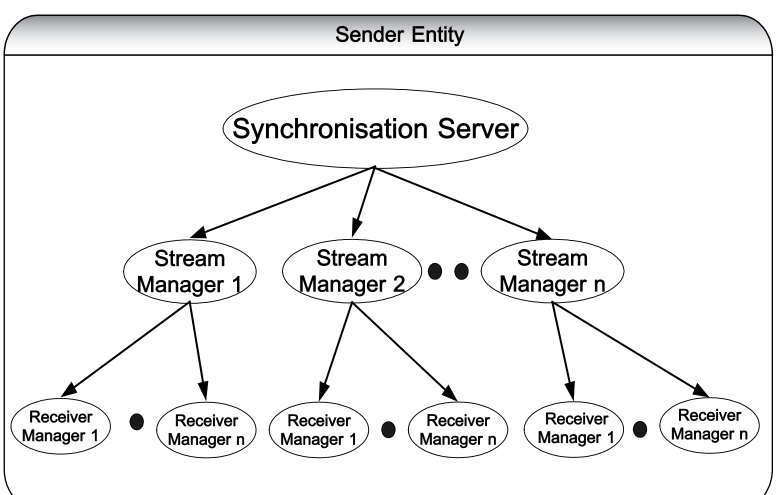

Fig. 3. The architecture and the data flow of the sender for simulcasting.

lower reception capabilities, the stream manager informs it that it has to move to a lower or higher quality stream.

Each receiver manager corresponds to a unique receiver (for scalability reasons a small representative group of receivers may have a corresponding receiver manager). The receiver manager processes the RTCP reports generated by the receiver and can be considered as a representative of the receiver at the side of the sender. It can interact only with one stream manager at a given time, the stream manager controlling the stream from which the receiver is receiving the video. The receiver manager receives the RTCP reports from the receiver and processes them based on packet loss rate and delay jitter information. It then makes an estimation of the state of the receiver based on the current report and a few previous reports that it stores in a buffer.

Receivers join the appropriate streams that better suit their requirements (available bandwidth between the sender and the receiver, etc.) and if during the transmission of multimedia data the network conditions to the path between them and the sender change, the receivers have the capability to change stream in order to better fulfil their requirements. Each receiver measures the characteristics of the path which connects it with the sender and informs the sender with the use of RTCP receiver reports. The communication between the sender and the receivers is based on RTP/RTCP sessions. The sender uses RTP to transmit the video streams and the participants (the sender and the receivers) use RTCP to exchange control messages.

5.4. *Layered Encoding*

In this mechanism the sender transmits multimedia data to a group of m receivers with the use of multicast. The sender uses the layered encoding approach, and transmits the video information in n different layers (the basic layer and $n-1$ additional layers). The sender transmits each layer in a different RTP/RTCP multicast session. The transmission rate within each layer adapts within its limits (each layer has an upper and lower limit in its transmission rate) according to the capabilities of the receivers. Receivers join the appropriate number of layers which better suit their requirements (available bandwidth between the sender and the receiver, etc.) and if during the transmission of multimedia data the network conditions to the path between them and the sender change, the receivers have the capability to receive more or fewer video layers in order to better fulfill their requirements. The communication between the sender and the receivers is based on RTP/RTCP sessions. The sender again uses the RTP to transmit the video layers and the participants (the sender and the receivers) use RTCP to exchange control messages.

Figure 4 shows the organization and the architecture of the sender entity. The sender generates n different layer managers. Each layer manager is responsible for the transmission of a video layer. The sender creates a new receiver manager

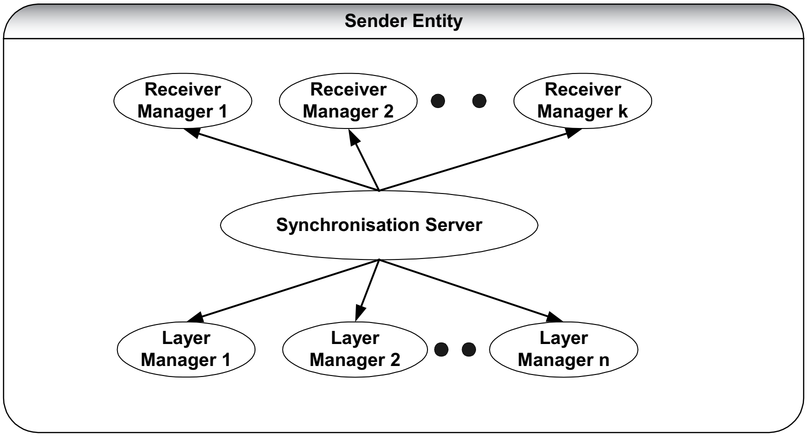

Fig. 4. The architecture and the data flow of the sender for layered encoding.

every time it receives an RTCP report from a new receiver. Each receiver manager corresponds to a unique receiver (for scalability reasons a small representative group of receivers may have a corresponding receiver manager). The receiver manager processes the RTCP reports generated by the receiver and can be considered as a representative of the receiver at the side of the sender. In addition, the synchronization server is responsible for the management, synchronization and intercommunication between layer managers and receiver managers. If a receiver manager does not receive RTCP reports from the receiver which it represents for a long time, it stops its operation and releases its resources.

Each receiver measures the characteristics of the path which connects it with the sender and informs the sender with the use of RTCP receiver reports.

The receivers join the appropriate layers that better suit their requirements (available bandwidth between the sender and the receiver, etc.) and if during the transmission of multimedia data the network conditions to the path between them and the sender change, the receivers have the capability to receive more or fewer video layers in order to better fulfill their requirements.

5.5. *Evaluation Parameters*

The following criteria are used to evaluate the approaches discussed:

- *Network congestion*: The goal of the adaptive transmission mechanisms is to increase the usage of the available bandwidth and decrease the packet losses of all the applications that transmit data in the same network path carrying the multicast data.
- *Scalability*: During transmission, the multimedia data may be received by a large number of receivers. The performance of the selected mechanism must not be degraded as the number of the receivers increases. This means that the complexity

and the performance of the mechanism used must be acceptable even when there are a large number of receivers.

- *Adaptation speed*: This refers to the time taken from the beginning of the multicast transmission until the selected mechanism achieves a stable operation. The smaller this time, the higher the performance of the mechanism.
- *TCP friendliness*: Most of the Internet traffic is TCP traffic. Any application that transmits data over the Internet should not adversely affect the other flows that exist and should be especially friendly towards TCP flows.
- *User satisfaction*: It is difficult to measure user satisfaction. For example, studies have show that during the transmission of MPEG video, just 3% packet loss can result in an up to 30% reduction in the presentation quality. As a result the satisfaction of the end user is influenced greatly by packet loss.

6. Directions for Future Research

The mechanisms described in this chapter have been proposed for installation and operation over the Internet. One interesting extension is tailoring these mechanisms to operate over mobile networks. The adaptive transmission of multimedia data over mobile networks is a challenge because one of their basic characteristics is the continuously changing environment. In order to tailor the above mechanisms for usage over mobile networks various issues must be considered, including:

- designing effective TCP models for wireless/mobile environments,
- creating more efficient encoding /decoding techniques, and
- power consumption.

As wireless communications and networking are quickly occupying centre stage of research and development activity in the area of communication networks, the suitability of the layered protocol architecture is coming under close scrutiny from the research community. Although layered protocol architectures have served wired networks well, they are not suitable for wireless networks. To illustrate this point, researchers usually present what they call a cross-layer design proposal. Thus, there have been a large number of cross-layer design proposals in the literature recently, with some of them focusing on multimedia transmission. Generally speaking, cross-layer design refers to actively exploiting the dependence between protocol layers to obtain performance gains. This is unlike layering, where the protocols at the different layers are designed independently.

7. Conclusion

Multimedia transmission is an important component of many current and future Internet applications, but high quality transmission is challenging because of the heterogeneous environment of the Internet. Multimedia transmission must be adapted to the various network changes due to the heterogeneous group of multimedia receivers and the lack of end-to-end QoS.

Many researchers have suggested that due to the use of new technologies that offer QoS guarantees, adaptive multimedia transmission applications will not be used in the future. We believe that this will not be the case 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 behavior.
- Some networks may never be able to provide specific QoS guarantees to users.
- Even if the Internet eventually supports reservation mechanisms or differentiated services, it is more likely to be on a per-class than a 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.

Terminology

Adaptive multimedia applications: Applications that have the capability to transmit multimedia data over heterogeneous networks and adapt media transmission to network changes.

Quality of service (*QoS*): The capability of a network to provide better service to selected network traffic.

Multimedia data: Data that consist of various media types like text, audio, video and animation.

RTP/RTCP: Protocols which are used for the transmission of multimedia data. RTP performs the actual transmission and RTCP is the control and monitoring Protocol.

Transmission Control Protocol (*TCP*): A connection-oriented, reliable protocol of the TCP/IP protocol suite used for managing full-duplex transmission streams.

TCP-friendly flow: A flow that consumes no more bandwidth than a TCP connection which is traversing the same path as that flow.

User Datagram Protocol (*UDP*): A connectionless, unreliable protocol of the TCP/IP protocol suite used for sending and receiving datagrams over an IP network.

Multicast: Transmitting data simultaneously to many receivers without the need to replicate the data.

Simulcast: Transmission of the same multimedia data in multiple multicast streams with different transmission rates.

Layered encoding: Transmission of the multimedia data in n different layers the basic layer and $n - 1$ additional layers.

Frame rate: The rate of the frames that are encoded by a video encoder.

Exercise

1. What is the difference between unicast and multicast multimedia transmission?
2. Which protocols are the *de facto* protocols for multimedia transmission and what are their services?
3. What is delay jitter and how is it calculated?
4. What are the most common approaches to handling packet loss during multimedia transmission?
5. What modules comprise the sender in a unicast adaptive multimedia transmission architecture?
6. What are the different approaches for multicast multimedia adaptive transmission over the Internet?
7. What is the difference between the simulcast and layered encoding approaches?
8. What is the most effective approach (in terms of bandwidth consumption) for multicast multimedia adaptive transmission over the Internet?
9. What is the most complicated approach (in terms of encoding and decoding issues) for multicast multimedia adaptive transmission over the Internet?
10. What are the evaluation parameters for multicast multimedia adaptive transmission over the Internet?

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