Encyclopedia of Internet Technologies and Applications

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Multicast of Multimedia Data

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INTRODUCTION

The heterogeneous network environment that Internet provides to real time applications as well as the lack of sufficient QoS (quality of service) guarantees, many times forces applications to embody adaptation schemes in order to work efficiently. In addition, any application that transmits data over the Internet should have a friendly behaviour toward the other flows that coexist in today's Internet and especially toward the *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 with that flow (Pandhye, Kurose, Towsley, & Koodli, 1999).

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

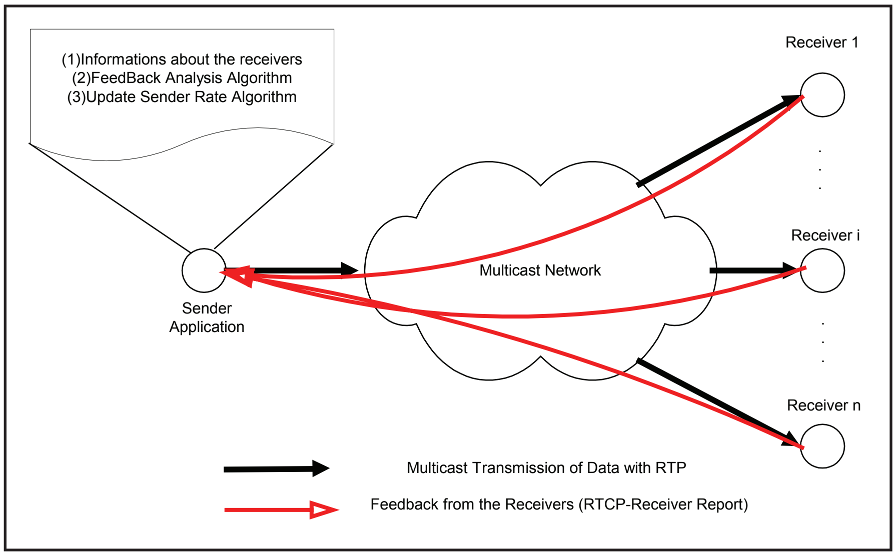
- **Transmission rate adaptation:** The sender must adapt the transmission rate based on the current network conditions.
- **TCP friendliness:** During the multicast transmission over the Internet, the multicasts flows must be TCP-friendly.
- **Scalability:** The performance of the adaptation scheme 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.

BACKGROUND

When someone multicasts multimedia data over the Internet, he or she has to accommodate receivers with heterogeneous data reception capabilities. To accommodate heterogeneity, the sender application may transmit one multicast stream and determine the transmission rate that better satisfies most of the receivers, may transmit at multiple multicast streams with different transmission rates and allocate receivers at each stream, or may use layered encoding and transmit each layer to a different multicast stream.

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. The previously described 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

Figure 1. Architecture of a single stream multicast transmission mechanism

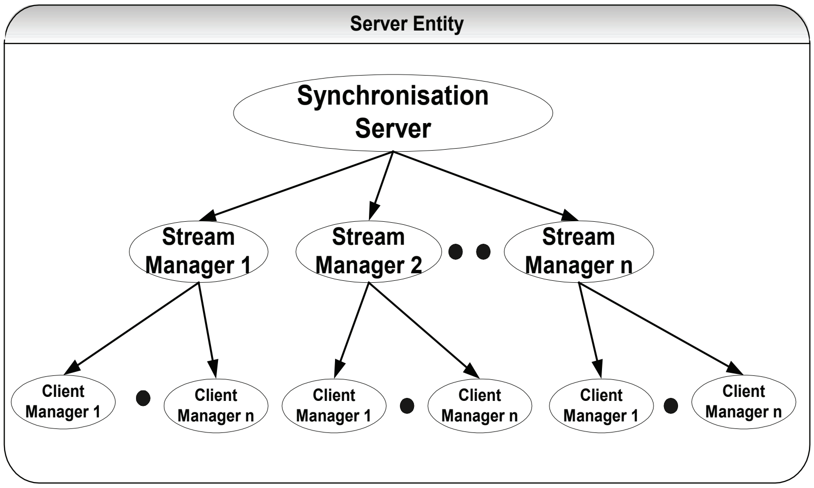


during the single multicast stream approach there is no need for synchronisation of receivers' actions (as is required for the multiple multicast streams and layered encoding approaches).

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

- The sender uses a single multicast stream for all receivers (Bouras & Gkamas, 2003). This results to the most effective use of the network resources, but on the other hand the fairness problem among the receivers arises, especially when the receivers have very different capabilities. The subject of adaptive multicast of multimedia data over networks with the use of one multicast stream has engaged many researchers. During the adaptive multicast transmission of multimedia data in a single multicast stream, the sender application must select the transmission rate that satisfies most of the receivers 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 (Pandhye et al., 1999), network feedback based (Jiang, Ammar, & Zegura, 1998; Sisalem, 1998) or based on a combination of the previous two approaches (Sisalem & Wolisz, 2000a).
- **Simulcast:** The sender transmits versions of the same video, encoded in varying degrees of quality. This results to the creation of a small number of multicast streams with different transmission rates (Bouras, Gkamas, Karaliotas, & Stamos, 2001). 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 in the stream that carries the video quality, in terms of transmission rate, that it is capable of receiving. The main disadvantage in this case is that the same multimedia information is replicated over the network but recent research has shown that under some conditions simulcast has better behavior that multicast transmission of layered encoded video (Kim & Ammar, 2001).
- The sender uses *layered encoded* video, which is video that can be reconstructed from a number of discrete data layers, the basic layer, and more additional layers, and transmits each layer into different multicast stream (Legout & Biersack, 2000; Sisalem & Wolisz, 2000b). The basic layer provides the basic quality and the quality improves

Figure 2. The architecture and the data flow of the server



with each additional layer. The receivers subscribe to one or more multicast streams depending on the available bandwidth into the network path to the source.

SINGLE STREAM MULTICAST TRANSMISSION OF MULTIMEDIA DATA

In such mechanism a sender application transmits multimedia data to a group of n receivers with the use of multicast in one stream. The sender application is using *RTP*/*RTCP* protocols for the transmission of the multimedia data. Receivers receive the multimedia data and inform the sender application for the quality of the transmission with the use of RTCP receiver reports. The sender application collects the RTCP receiver reports, analyses them and determines the transmission rate that satisfy most the group of receivers with the current network conditions.

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

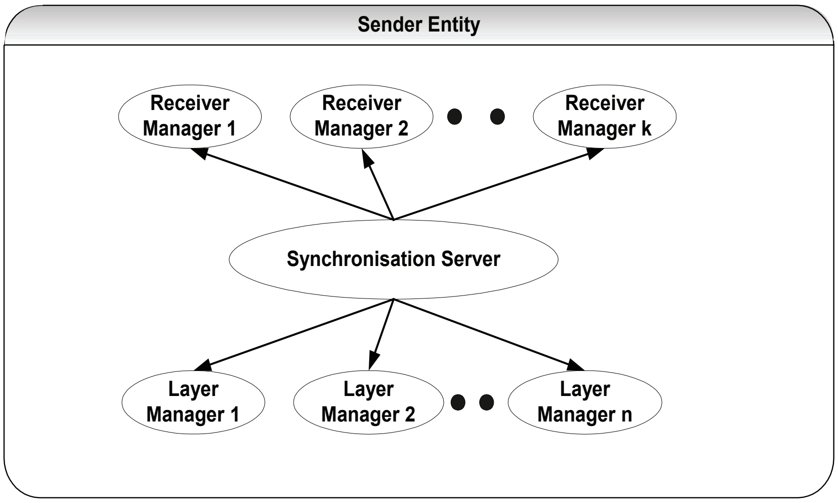
- **Feedback analysis algorithm:** Feedback analysis algorithm analyses the feedback information that the receivers sends to the sender application (most mechanisms use RTCP receiver reports for this purpose), concerning the transmission quality of the multimedia data. Every time the sender application receives feedback from a receiver, it runs the feedback analysis algorithm in order to estimate the preferred transmission rate, which will satisfy that receiver. The receiver's preferred transmission rate represents the transmission rate that this receiver will prefer if it was the only one receiver in the multicast transmission of the multimedia data.
- **Update sender rate algorithm:** The sender application in repeated time periods estimates the transmission rate for multicasting the multimedia data with the use of the update sender rate algorithm. The estimation of the sender application transmission rate is aiming to increase the satisfaction of the group of receivers. When the sender application estimates the new transmission rate, it tries to provide to the group of receivers the best satisfaction that the current network conditions allow.

SIMULCAST

In such a mechanism, the server is unique and responsible of:

- Creating the n different multicast streams (in most mechanisms a small number of multicast streams, usually 3 or 4 are enough)

Figure 3. The architecture and the data flow of the sender



- Setting each stream's bandwidth limits
- Tracking if there are any clients that are not handled with fairness
- Providing the mechanisms to the clients to switch streams whenever they consider that they should be in another stream closer to their capabilities

Figure 2 shows the organisation and the architecture of the server entity. The server generates n different stream managers. In each stream manager, an arbitrary number of client managers is assigned. Each client manager corresponds to a unique client that has joined the stream controlled by this stream manager. The synchronisation server is responsible for the management, synchronisation, 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 periodically gathers the states reported by all client managers belonging to it at the end of a specific, fixed time period. It then uses an appropriate algorithm that tries to improve fairness between clients by determining whether a lower or a higher bit rate is more appropriate. Whenever a client cannot be satisfied by a stream due to the fact that most of the other clients have much higher or much lower reception capabilities, the stream manager informs it that it has to move to a lower or higher quality stream.

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

LAYERED ENCODING

In such mechanism, the sender transmits multimedia data to a group of m receivers with the use of multicast. The sender is using 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 into a different RTP/RTCP multicast session. The transmission rate within each layer is adapting within its limits (each layer has an upper and lower limit in its transmission rate) according to the capabilities of the receivers listening up to it. The 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 less video layers in order to accomplish better their requirements. The communication between the sender and the receivers is based on RTP/RTCP sessions and the sender is using the RTP protocol to transmit the video layers and the participants (the sender and the receivers) use the RTCP protocol in order to exchange control messages.

Figure 3 shows the organisation 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 every time receives a RTCP report from a new receiver. Each receiver manager corresponds to a unique receiver (for scalability issues a small representative group of receivers may have a corresponding receiver manager). It 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 synchronisation server is responsible for the management, synchronisation 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 receiver reports.

EVALUATION PARAMETERS

During the multicast transmission of multimedia data over the Internet the overall target is the optimal usage of the network resources and for this reason an appropriate mechanism is used. In order to evaluate those mechanisms there are the following criteria:

- **Network congestion:** The goal of the multicast 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 with the network path of the multicast data.
- **Scalability:** During the multicast transmission of multimedia data, the multimedia data may be received by a large number of receivers. The performance of the selected mechanism must not be downgraded when the number of the receivers of the multicast data is increased. This means that the complexity and the performance of the used mechanism must be acceptable even when a large number of receivers receive the multimedia data through the multicast transmission.
- **Adaptation speed:** With the term adaptation speed we refer to the time needed from the begging of the multicast transmission of the multimedia data until the selected mechanism achieves a stable operation. This time must be relatively small and the performance of the mechanism is better when this time is small.
- **TCP friendliness:** Most of the Internet traffic is TCP traffic. Any application that transmits data over the Internet should have a friendly behaviour toward the other flows that coexist in today's Internet and especially toward the TCP flows that comprise the majority of flows.
- **User satisfaction:** It is difficult to measure the user satisfaction. For example, studies has show that during the transmission of MPEG video, just 3% packet loss can result up to 30% reduction of the presentation quality. As a result the satisfaction of the end user is influenced very much from the packet loss.

TRANSMISSION OF MULTIMEDIA DATA

The transmission of the multimedia data is based on the protocols RTP/RTCP. The protocol RTP is used for the transmission 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 need. 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. RTP packet headers include information about the payload type of the data, numbering of the packets and timestamping information.

RTCP offers the following services to applications:

- **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 with many members.
- **Inter-media synchronisation:** In real time applications it is common to transmit audio and video in different data streams. RTCP provides services like timestamping, which can be used for inter-media synchronisation of different data streams (for example synchronisation of audio and video streams).

More information about RTP/RTCP can be found in RFC 3550 (Schulzrinne, Casner, Frederick, & Jacobson, 2003).

FUTURE TRENDS

The mechanisms described in the previous paragraphs have been proposed for installation and operation over the Internet. One interesting extension of the previous mechanisms is the adaptation of the previous mechanisms to operate over mobile networks. The multicast transmission of multimedia data over mobile networks is a challenge due to the fact the one of the basic characteristics of mobile networks is the continuously changing environment. In order to adapt the previously described mechanisms for usage over mobile networks various issues must be considered such as more efficient encodings.

CONCLUSION

The multicast transmission of real time multimedia data is an important component of many current and future emerging Internet applications such as video-conferencing, distance learning, and video distribution. The heterogeneous nature of the Internet makes the multicast transmission of real time multimedia data a challenge. Different receivers of the same multicast stream may have different processing capabilities, different loss tolerance and different bandwidth available in the paths leading to them.

When multicast multimedia data is transmitted over the Internet, receivers with heterogeneous data reception capabilities have to be accommodated. To accommodate heterogeneity, the sender application may transmit one multicast stream and determine the transmission rate that 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.

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KEY TERMS

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

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

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

Quality of Service (QoS): Quality of service refers to the capability of a network to provide better service to selected network traffic.

RTP/RTCP: Protocol that is used for the transmission of multimedia data. The RTP performs the actual transmission and the RTCP is the control and monitoring transmission.

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