Encyclopedia of Internet Technologies and Applications

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Adaptive Transmission of Multimedia Data over the Internet

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INTRODUCTION

Internet is a heterogeneous network environment and the network resources that are available to real time applications can be modified very quickly. 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. 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.

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 (Diot, 2001).

In this article, 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.

BACKGROUND

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 for the transmission of control information (Parry & Gangatharan, 2005; Vandalore, Feng, Jain, & Fahmy, 1999). Another approach for the transmission of the multimedia data is the use of RTP over UDP (Bouras & Gkamas, 2003; Byers et al., 2000). Most adaptive applications use RTP/RTCP (real time transmission protocol / real time control transmission protocol) (Schulzrinne, Casner, Frederick, & Jacobson, 2003) for the transmission of the multimedia data. The RTP protocol seems to be the de facto standard for the transmission of multimedia data over the Internet and is used both by mbone tools (vit, vat, etc.) and ITU H.323 applications. In addition RTCP offers capabilities for monitoring the transmission quality of multimedia data.

For the implementation of the network monitoring module, a common approach is to use the packet loss as an indication of congestion in the network (Bouras et al., 2003; Byers et al., 2000). One other approach for monitoring the network conditions is the use of utilization of the client buffer (Rejaie, Estrin, & Handley, 1999; Walpole et al., 1997). An important factor that can be used for monitoring the network conditions, and especially for indication of network congestion, is the use of delay jitter during the transmission of the multimedia data.

For the implementation of the adaptation module, some common approaches are the use of rate shaping (Byers et al., 2000; Bouras et al., 2003), the use of layered encoding (Rejaie et al., 1999), the use of frame dropping (Walpole et al., 1997) or a combination of the previous techniques (Ramanujan et al., 1997). The implementation of the adaptation module depends on the encoding method that is used for the transmission of the multimedia data. For example, in order to use the frame dropping technique for the adaptation of a MPEG video stream, a selective frame dropping technique must be used, due to the fact that MPEG video uses inter-frame encoding and some frames contain information relative to other frames. In Vandalore et al. (1999), a detailed survey of application level adaptation techniques is given.

It is important for adaptive real time applications to have "friendly" behavior to the dominant transport protocols (TCP) of the Internet (Floyd & Fall, 1998). In Widmer et al. (2001), a survey on TCP-friendly congestion control mechanisms is presented.

ADAPTIVE TRANSMISSION OF MULTIMEDIA DATA OVER THE INTERNET

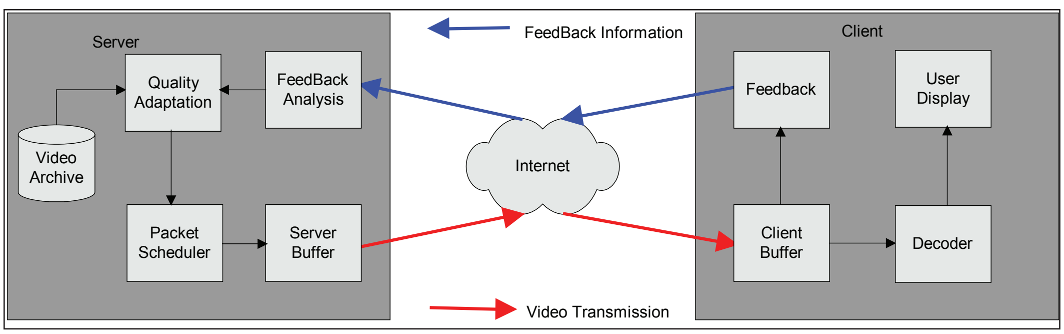
The Architecture of an Adaptive Streaming Application

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 (who have faster network

Figure 1. System architecture



connection with the server) and in H.263 format in order to serve distant users with slow network connections. In this article, we do not investigate the problem of video storage in video archives in order to achieve the optimal performance of the server.
- **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. 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.

The client of the adaptive streaming architecture consists of the following modules:

- **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 (we suppose that we measure the buffer capacity and the delay jitter in the same units, e.g. in seconds).
- **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.

In the following paragraphs, we give a detailed description of the most important modules of the previously described architecture.

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 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. 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 et al., 2003).

Feedback from the Network

The presentation quality of real time data depends on the packet loss rate and the delay jitter during the transmission over the network. In addition, packet losses or rapid increases of delay jitter may be considered as an indication of problems during the transmission of data over the network. In such a case, the adaptive streaming application must adapt the transmission of the data in order to avoid phenomenon like network congestion. Real time applications have upper bounds to the packet loss rate and to the delay jitter. If packet loss rate or jitter gets to be over these upper bounds, the transmission of real time data can not 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 of 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 the packets for a period of time. Some other researchers define delay jitter as the maximum difference between the delay of the transmission of two sequential packets for a period of time. According to RFC 3550 (Schulzrinne et al., 2003), delay jitter is defined to be the mean deviation (smoothed absolute value) of the difference D 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 R_i is the time of arrival for this packet, then for two packets i and j, D is defined as: $D(i,j) = (R_j - R_i) - (S_j - S_i) = (R_j - S_j) - (R_i - S_i)$. 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:

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

The previous 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 queue of the network devices. The different delays are related to the serve model of each queue and the cross traffics in the transmission path.

Sometimes delay jitter occurs during the transmission of real time data, which does not originate from the network but is originated from the transmission host (host included delay jitter). This is because during the encoding of the real time 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 real time 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 estimation for the network conditions.

We can conclude that delay jitter can not lead to reliable estimation of network condition by itself. Delay jitter has to be used in combination with other parameters, like packet loss rate, in order to make reliable estimations of the network conditions. In Bouras et al. (2003), it is shown that the combination of packet loss rate and delay jitter can be used for reliable indication of network congestion.

Quality Adaptation

Quality adaptation module is based on the rate shaping technique. According to the rate shaping 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. Rate shaping techniques change one or more of the following parameters:

- **Frame rate:** Frame rate is the rate of the frames, which are encoded by video encoder. Decreasing the frame rate can reduce the amount of the data that the video encoder produces but will reduce the quality.
- **Quantizer:** The quantizer specifies the number of DCT coefficients that are encoded. Increasing the quantizer decreases the number of encoded coefficients and the image is coarser.
- **Movement detection threshold:** This is 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.

Error Control/Packet Loss

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

- **APQ (Automatic Repeat Request):** APQ is an active technique where the receiver and ask the sender to retransmit some lost packets.
- **FEC (Forward Error Correction):** 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.

FUTURE TRENDS

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 (Byers et al., 2000; Rizzo, 2000; Widmer et al., 2001), 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 (Byers et al., 2000). 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 (Rizzo, 2000; Widmer et al. (2001), network feedback based (Byers et al., 2000), or based on a combination of the previous two approaches (Sisalem & Wolisz, 2000).

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 applications 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.

REFERENCES

Bouras, C., & Gkamas, A. (2003). Multimedia transmission with adaptive QoS based on real time protocols. *International Journal of Communications Systems, Wiley InterScience*, *16*(2), 225-248

Byers, J., Frumin, M., Horn, G., Luby, M., Mitzenmacher, M., Roetter, A., & Shaver, W. (2000). FLID-DL: Congestion control for layered multicast. In *Proceedings of NGC* (pp. 71-81).

Cheung, S. Y., Ammar, M., & Xue, L. (1996). On the use of destination set grouping to improve fariness in multicast video distribution. In *Proceedings of INFOCOM 96*, San Francisco.

Diot, C. (2001, January 25-26). On QoS & traffic engineering and SLS-related work by Sprint. *Workshop on Internet Design for SLS Delivery*, Tulip Inn Tropen, Amsterdam, The Netherlands.

Floyd, S., & Fall, K. (1998, August). Promoting the use of end-to-end congestion control in the Internet. In *IEEE/ACM Transactions on Networking*.

Li, X., Ammar, M. H., & Paul, S. (1999, April). Video multicast over the Internet. *IEEE Network Magazine.*

Parry, M., & Gangatharan, N. (2005). Adaptive data transmission in multimedia networks. *American Journal of Applied Sciences*, *2*(3), 730-733.

Ramanujan, R., Newhouse, J., Kaddoura, M., Ahamad, A., Chartier, E., & Thurber, K. (1997). Adaptive streaming of MPEG video over IP networks. In *Proceedings of the 22nd IEEE Conference on Computer Networks,* 398-409.

Rejaie, R., Estrin, D., & Handley, M. (1999). Quality adaptation for congestion controlled video playback over the Internet. In *Proceedings of ACM SIGCOMM '99,* 189-200. Cambridge.

Rizzo, L. (2000) pgmcc: A TCP-friendly single-rate multicast congestion control scheme. In *Proceedings of SIGCOMM 2000*, Stockholm.

Schulzrinne, H., Casner, S., Frederick, R., & Jacobson, V. (2003). *RTP: A transport protocol for real-time applications*, RFC 3550, IETF.

Sisalem, D., & Wolisz, A. (2000). LDA+ TCP-friendly adaptation: A measurement and comparison study. The *Tenth International Workshop on Network and Operating Systems Support for Digital Audio and Video*, Chapel Hill, NC.

Vandalore, B., Feng, W., Jain, R., & Fahmy, S., (1999). A survey of application layer techniques for adaptive streaming of multimedia. *Journal of Real Time Systems (Special Issue on Adaptive Multimedia)*.

Vickers, B. J., Albuquerque, C. V. N., & Suda, T. (1998). Adaptive multicast of multi-layered video: Rate-based and credit-based approaches. In *Proceedings of IEEE Infocom,* 1073-1083.

Walpole, J., Koster, R., Cen, S., Cowan, C., Maier, D., McNamee, D., et al. (1997). A player for adaptive mpeg video streaming over the Internet. In *Proceedings of the 26th Applied Imagery Pattern Recognition Workshop AIPR-97*, SPIE, (Washington DC), 270-281.

Widmer, J., Denda, R., & Mauve, M., (2001). A survey on TCP-friendly congestion control mechanisms. *Special Issue of the IEEE Network Magazine Control of Best Effort Traffic, 15,* 28-37.

Widmer, J., & Handley, M. (2001). Extending equation-based congestion control to multicast applications. In *Proceedings of the ACM SIGCOMM* (San Diego, CA), 275-285.

KEY TERMS

Adaptive Real Time Applications: Adaptive real time applications are application that have the capability to transmit multimedia data over heterogeneous networks and adapt media transmission to network changes.

Delay Jitter: Delay jitter is defined to be the mean deviation (smoothed absolute value) of the difference in packet spacing at the receiver compared to the sender for a pair of packets.

Frame Rate: Frame rate is the rate of the frames, which are encoded by video encoder.

Movement Detection Threshold: The movement detection threshold is a parameter that limits the number of blocks which are detected to be "sufficiently different" from the previous frames.

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

Packet Loss Rate: Packet loss rate is defined as the fraction of the total transmitted packets that did not arrive at the receiver.

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

Quantizer: Quantizer specifies the number of DCT coefficients that are encoded.

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