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IMPLEMENTATION ON ADAPTIVE TRANSMISSION OF REAL TIME MULTIMEDIA DATA

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ABSTRACT

Several mechanisms for adaptive transmission of multimedia data, which is based on real time protocols RTP/RTCP.RTCP offers capabilities for monitoring the transmission quality of multimedia data. When multimedia data (audio, video) are transmitted over a network with low Quality of Service Management there is a high possibility of router queue overflow. To overcome this problem Adaptive Transmission Rate Algorithm is used. To do this Packet size of each source is adjusted according to network bandwidth. A controller is used to trace the data transmission rate at the router. An algorithm is developed and coded in Tool Command Language. Simulation is performed on NS2. The present work is developed using JMF over RTP only.

KEYWORDS: NS2, JMF, RTP, RTCP, TCL, Adaptive Transmission Rate Algorithm, QOS, TCP, UDP

1. INTRODUCTION

Real time multimedia data is very much important in web content, biometric information and entertainment world. In this paper we first identify and study basic concepts of real time multimedia data and then we compare transmission rate for multimedia data (video) using UDP protocol and TCP for control information and RTP over UDP. Finally develop and populate an adaptive transmission rate algorithm on Tool Command Language. An adaptive real time application is an application that has the capability to transmit multimedia data over heterogeneous networks and adapt the transmission of multimedia data to the network changes.

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 sufficiently. In addition any application that transmits over the internet should have a friendly behavior toward the other flows that coexist in today's Internet and especially toward the TCP flows that comprise the majority of flows. We define a TCP friendly flow that consumes no more bandwidth than a TCP connection which is traversing the same path with that flow.

During the multicast transmission over the Internet several aspects need to be considered. 1) Transmission Rate Adaptation-The sender must adapt the transmission rate based on the current network conditions. 2) TCP friendliness-During the multicast transmission over the internet, the multicast flows must be TCP friendly. 3) Scalability is performance of the adaptation scheme must not deteriorate with increasing number of receivers. 4) 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 receivers if not all possible receivers.

When audio and video are transmitted over a network with low QOS management there is a high possibility of router Queue Overflow. When the total transmission rate is higher than the network bandwidth packets are lost because they are dropped when a router queue overflows or when a client buffer overflows.

According to Adaptive Transmission Rate Algorithm the packet size of each source is adjusted according to the network bandwidth. In a network there are some quality requirements in order to successfully run an application. These requirements are commonly referred to as Quality of Service (QOS). The term QOS is a subjective term. QOS is a service provided by the service plane to an end user (e.g.., a host or a network element) which utilizes the IP transfer capabilities and associated control and management functions for delivery of the user information specified by the service level agreements. An optimal quality of service is the one in which the behavior of data transmission in a network (e.g. access to a file) would be the same as in a local disk: for instance, the user would read a remote file like if it was reading from a locally stored file.

Real Time Protocol (RTP) is a point to point used to carry multimedia traffic, namely audio and video, over IP networks. This protocol provides also network transport functions intended for applications with real time requirements, videoconference or simulation data, over multicast or unicast services.

Real Time Control Protocol: RTP data transport is improved with a control (RTCP) protocol, which provides feedback on the quality of data transmission to the RTP session. The transmitted packets must be multiplexed into data and control packets. The RTCP packets contain information about QOS monitoring and congestion control, session size estimation and scaling [6]. Basically RTCP appears as a solution that gives reliability to a RTP data flow. RTP/RTCP provides functionality and control mechanisms necessary to carry real time content done at the application level. The flow control congestion information is provided by the RTCP sender and receiver reports.

The architecture of an adaptive real time application that has the capability to transmit multimedia data over heterogeneous networks and adapt the transmission of multimedia data to the network changes [2]. During the design and implementation of adaptive application special attention must be paid to the following critical modules.

The module is responsible for the transmission of multimedia data. The module is responsible for monitoring the network conditions and determines the change to the network conditions. The module is responsible for adaptation of multimedia data to the network changes. The module is responsible for handling the transmission errors during the transmission of multimedia data.

In the present work a common approach for the implementing adaptive applications is the use of UDP for transmission of multimedia data and TCP for the transmission of control information. Another approach for the transmission of multimedia data is the use RTP over UDP. Most adaptive applications use RTP/RTCP for the transmission of multimedia data. RTCP offers capabilities for monitoring the transmission quality of multimedia data [10].

In first implementation based on the above concept assumptions are made for the transmission of real time multimedia data over RTP using JMF takes three parameters in the constructor input media locator, destination IP address, Destination Port number.

When the total transmission rate is higher than the network bandwidth packets are lost because they are dropped when router queue overflows or when a client buffer overflows. There is a high possibility for a router queue overflow when audio and video data are transmitted over a network with low QOS management. In second implementation based on the above concept assumptions are made to create agent for UDP protocol and cbr0. In section 2 represents the literature survey, section 3 shows design, solution and Software Architecture, section 4 indicates experiments, results and interpretations, section 5 signifies conclusion, section 6 contains references.

2. LITERATURE SURVEY

We discuss here 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 multimedia data to the network changes. More over in this article we concentrate on the unicast transmission of multimedia data.

During the design and the implementation of an adaptive transmission [1] of multimedia data special attention must be paid to the following critical modules- 1) the module is responsible for the transmission of Multimedia data. 2) The module, which is responsible for monitoring network conditions and determines the change to the network conditions. 3) The module, which is responsible for adaptation of the multimedia data to the network changes. 4) The module 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 TCP for the transmission of control information. Another approach for the transmission of multimedia data is the use of RTP over UDP. Most adaptive applications use RTP/RTCP for the transmission of Multimedia data. RTCP offers capabilities for monitoring the transmission quality of multimedia data [8]. For the implementation of the network-monitoring module; a common approach is to use the packet loss as an indication of congestion. One other approach for monitoring network conditions is the utilization of client buffer. An important factor that can be used for monitoring the network conditions and especially for indication of network conditions is the use of delay jitter during the transmission of multimedia data.

For the implementation of the adaptation module some common approaches are use of rate shaping, the use of layered encoding, and the use of frame dropping [9]. The implementation of adaptation module depends on the encoding method that is used for the transmission of multimedia data. In order to use frame-dropping technique for the adaptation of a mpeg video stream a selectable frame dropping technique must be used. Mpeg video uses inter frame encoding & some frames contain information relative to other frames.

Adaptive real time applications have friendly behavior to the dominant transport protocols (TCP) of the Internet. The server of the adaptive streaming architecture consists of the following modules –Video archive: -video archive consists of the set of hard disks in video files are stored. The adaptive streaming application may support various video formats (for ex 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. Feedback Analysis: -The module is responsible for the analysis of feedback information from n/w. 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.

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

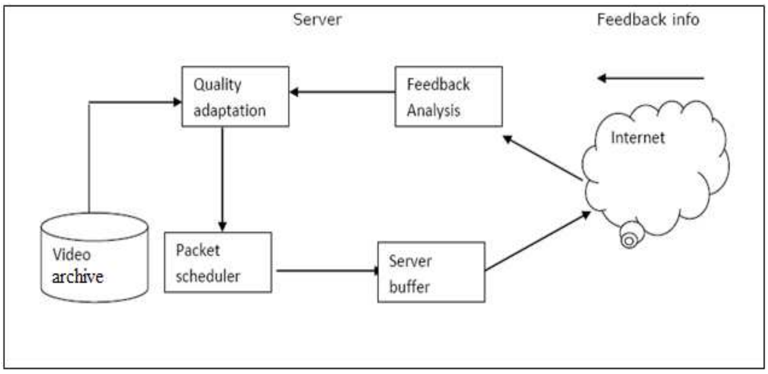

Figure 1

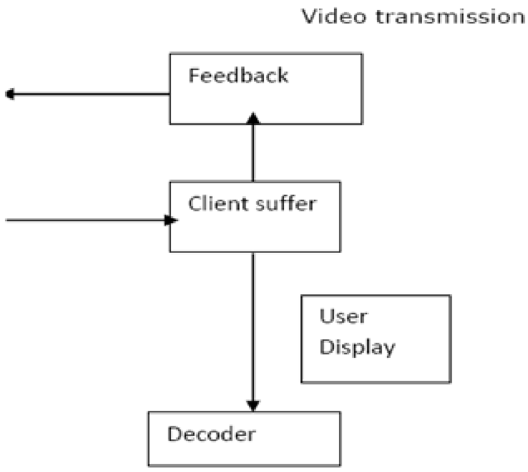

Figure 2

2.1 Transmission of Multimedia Data

It is based on RTP/RTCP. The protocol RTP is used for transmission of the multimedia data from server to the client & client uses RTCP protocol in order to inform the server of the transmission quality.

RTP/RTCP is one-way communication video on demand or two-way communication video conference. RTCP is the control information of RTP.RTP is a protocol that offers end-to-end transport services with real time characteristics over packet switching network like IP networks. RTP packet headers include information about numbering of packets.

In this research [2] suggests the most prominent enhancement of adaptive real time applications is the use of multicast transmission of multimedia data. The multicast transmission of multimedia data over i/p has to accommodate clients with reception capabilities. To accommodate heterogeneity the server may transmit one multicast stream & determine the transmission rate that satisfies most of the clients and may transmit multiple multicast streams with different transmission rates and allocate client at each stream. Or may use layered encoding and transmit each layer to a different multicast stream. Single multicast stream approaches have the disadvantages that clients with a low bandwidth link will always get a high bandwidth link. The problem can be overcome with the use of a multi stream multicast data. Single multicast streams have advantages of easy encoder & decoder implementation & simple protocol application.

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 data over Internet should have a friendly behaviors toward the other flows that co-exist in today's Internet and especially toward 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.

According to Bert J. Dempsey [3] When someone multicasts multimedia data over the Internet, he or she 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 [12], may transmit at multiple multicast streams with transmission rates and allocate receivers at each stream or may use layered encoding and transmit each layer to a different multicast stream [11].

The single multicast stream approach has the disadvantages 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 during single multicast stream approach, there is no need for synchronization of receiver's actions (as is required for multiple multicast streams and layered encoding approaches).

In this paper [4] the methods proposed for the multicast transmission of multimedia data over the Internet can be generally divided into three main categories depending on the no of multicast streams used.

The sender uses a single multicast stream for all receivers. 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, network based, or a combination of the previous two approaches.

3. DESIGN / SOLUTION / SOFTWARE ARCHITECTURE

Following are the procedures for transmission of video using JMF:

Input media locator. It can be a file or http or capture source. Then it starts the transmission. It returns Null if transmission started ok. Otherwise it returns string with the reason why the set up failed.

Create a processor for the specified media locator and program it to output JPEG/RTP. Create an RTP session to transmit the output of the processor to the specified IP address and port no.

Start the transmission. It stops the transmission if already started. Try to create a processor to handle the input media locator. Try wait for it to configure.

To get the tracks from the processor search through the tracks for a video track. If find a video track try to program it to output JPEG/RTP

Make sure the sizes are multiple of 8's set the output content descriptor to RAW-RTP. Realize the processor.

This will internally create a flow graph and attempt to create an output data source for JPEG/RTP. Set the JPEG quality to 0.5.

To get the output data source of the processor create an RTP transmit data sink. This is the easiest way to create an RTP transmitter. The other way is to use RTP session manager API. Using the RTP session manager gives you more control if you wish to fine tune your transmission and set other parameters. Create a media locator for the RTP data sink. Create a data sink, open it and start transmission. It will wait for the processor to start sending data so we need to start the processor itself which is done after this method returns.

Setting the encoding quality to the specified value on the JPEG encoder. Loop through the controls to find the quality control for JPEG encoder. Check to see if the owner is a codec then check for the output format.

Call the required method on the processor and wait until we get an event that confirms the success of the method or a failure event. If there was an error during configure or realize, the processor will be closed. All controller events send a notification to the waiting thread in wait for state method we need three parameters to do the transmission. Create a video transmit object with the specified parameters.

Start the transmission result will be non-null if there was an error. The return value is a string describing the possible error.

Transmit for 60 seconds and then close the processor. This is safeguard when using a capture data source.

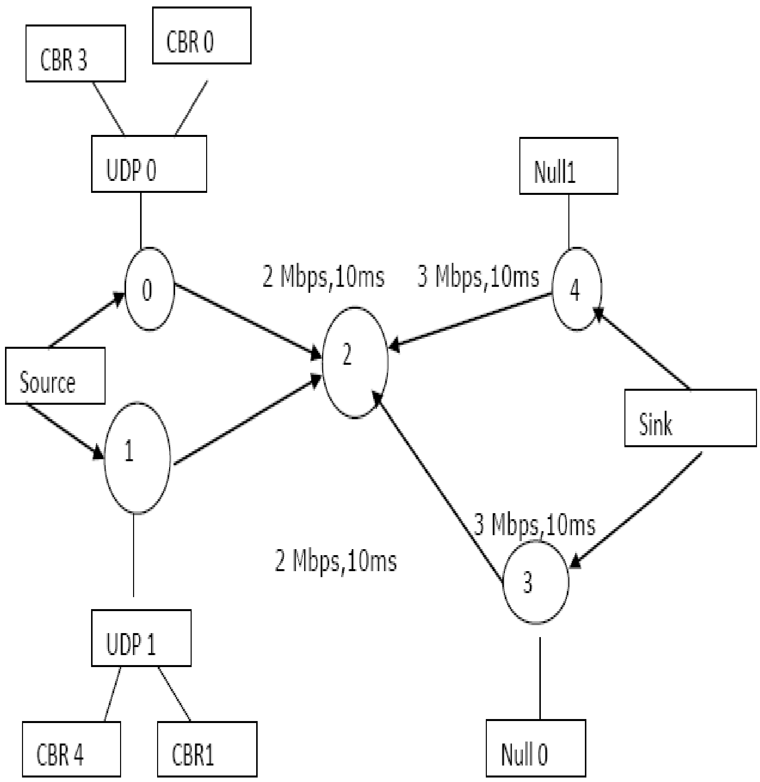

Figure 3: Topology for Adaptive Transmission Rate Algorithm

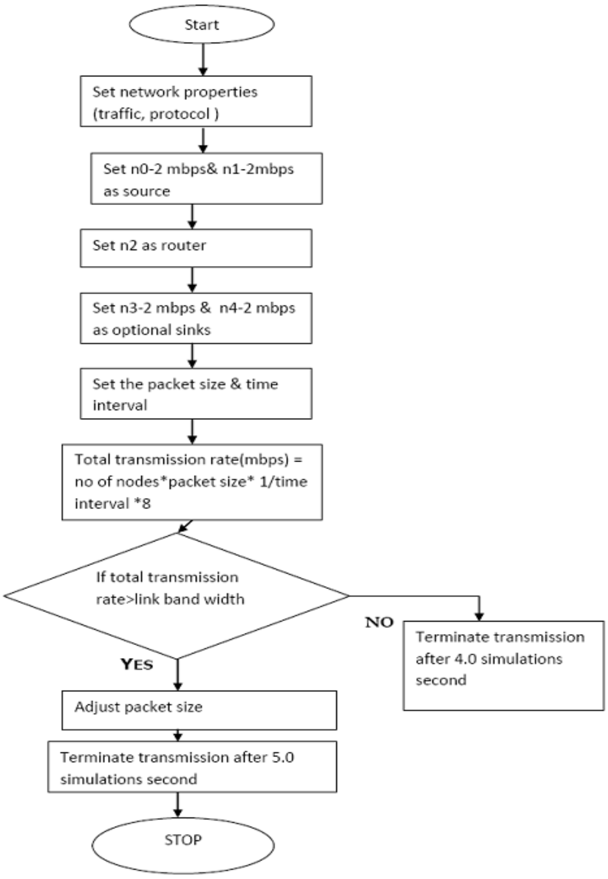

Figure 4: Flow Chart for Adaptive Transmission Rate

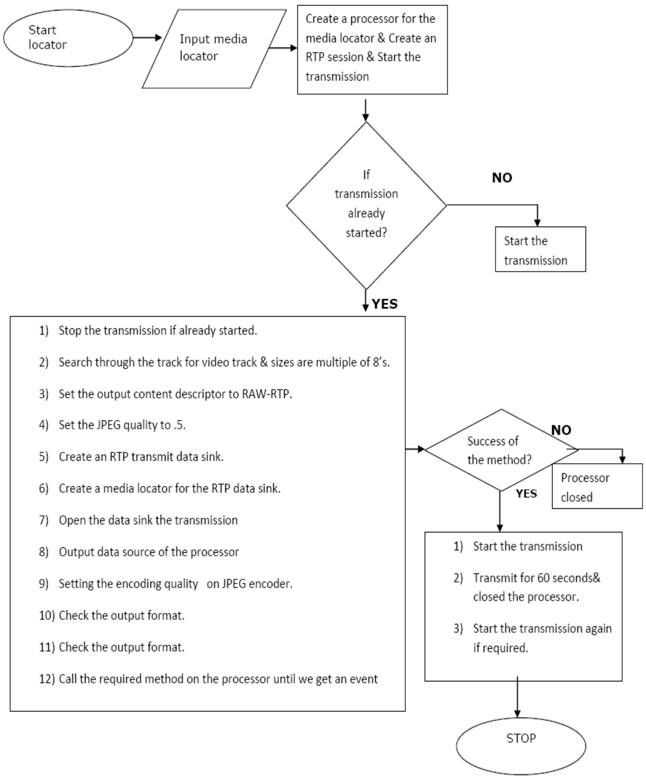

Figure 5: Flow Chart for Video Transmission

4. EXPERIMENTATIONS, RESULTS & INTERPRETATION

After transmitting video named Wall-E.avi (701 MB) video is being transmitted easily and it is easy to calculate frame rate and it is easy to convert it in JPEG/RTP format.

The following are some of the results from the developed system and their interpretation:

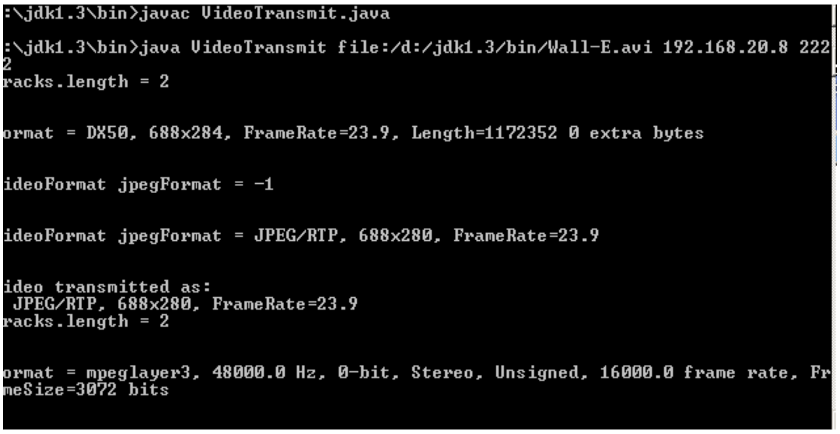

Figure 6

The above screen is the output for the video transmission. The video is transmitted as JPEG/RTP. Frame rate and track length is calculated here.

Figure 7

The above screen is waiting for data. JM studio is used for transmitting through IP address and port no.

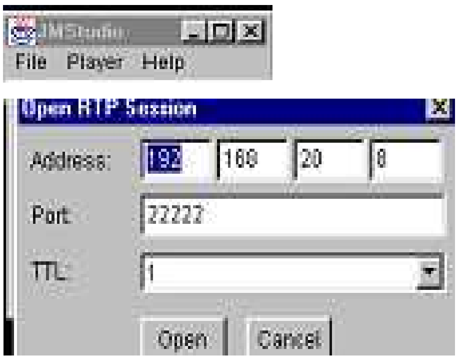

Figure 8

The above screen is for setting IP address, port number for the transmitting computer for transmission of video using JM studio.

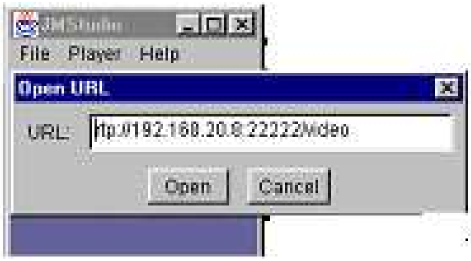

Figure 9

The above Screen is for setting URL using rtp://<source IP >:<port> /video

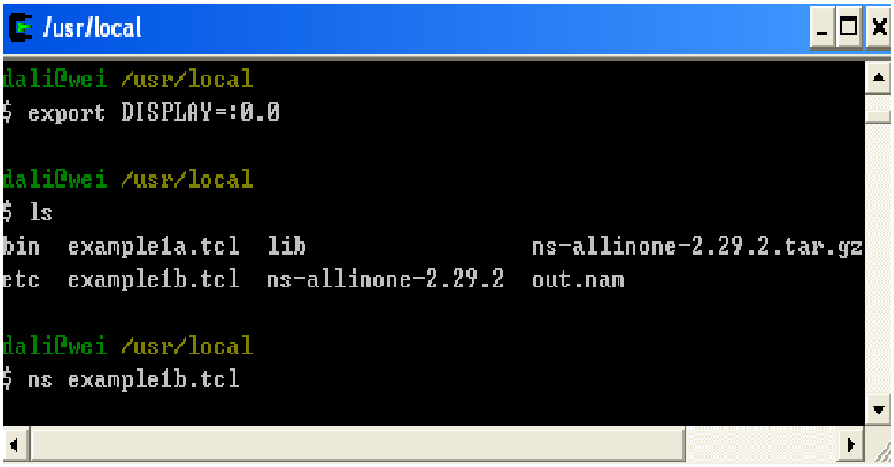

Figure 10

The above screen is for the implementation of adaptive transmission rate algorithm.

5. CONCLUSIONS

In this work it has been focused to transmit the real time multimedia data in the easiest way based on the disadvantages of the earlier works carried out using JMF (Java Media Framework). There is a high possibility for a router queue overflow when audio and video data are transmitted over a network with low QOS management. A controller is used to monitor and adjust the data flow from the router to the sinks. With such mechanisms QOS requirements can be achieved by avoiding the occurrence of data losses. Within the following way the main objectives are being fulfilled:-

- Using this technique Router Queue Overflow can be easily avoided.
- In present work it is easy to handle and easy to transmit video using JM studio.
- In present work it is easy to calculate frame rate, track length of video.

6. REFERENCES

1. Cristina Aurrecoechea, A. Campbell, L. Hauw, A Survey of QoS Architectures, ACM/Springer-Verlag Multimedia Systems Journal, Special Issue on QoS Architecture, Vol. 6/3, May-1998, pp. 138-151.
2. Gabriel Muntean, Liam Murphy, A Novel Feedback-Controlled Multimedia Transmission Scheme, 8th IEEE International Conference on Telecommunication, June-2001, Bucharest-Romania, Vol. III, pp. 123-128.

3. Bert J. Dempsey, W. Timothy Strayer, Alfred C. Weaver, “Adaptive Error Control for Multimedia Data Transfers”
4. Vickers, B. J., Albuquerque, C. V. N., Suda, T. Adaptive Multicast of Multi-Layered Video: Rate-Based and Credit Based Approaches, Proc. of IEEE Infocom, March 1998.
5. Christos Bouras and A. Gkamas, “A Mechanism for Multicast Multimedia Data with Adaptive QoS Characteristics”, M.J. van Sinderen and L.J.M. Nieuwenhuis (Eds.): PROMS 2001, LNCS 2213, pp. 74-88, 2001.
6. Lorenzo Vicisano, Jon Crowcroft, “TCP-like Congestion Control for Layered Multicast Data Transfer”
7. Lorenzo Vicisano, Jon Crowcroft, “TCP-like Congestion Control for Layered Multicast Data Transfer”
8. Time, the packet loss rate is greatly decreased. H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, “RTP: A Transport Protocol for Real –Time Applications,” *RFC 1889*, January 1996.
9. Hugo Pereira, Marília Curado, Paulo de Carvalho, “QoS in real time data transmission”, The University of Melbourne; IEEE Communications Magazine; December 2002.
10. Shu-Ching Chen, Mei-Ling Shyu, Irina Gray1, Hongli Luo, “An Adaptive Multimedia Transmission Protocol for Distributed Multimedia Applications”.
11. Manimegalai Parry and N.Gangatharan, “Adaptive Data Transmission in Multimedia Networks”, American Journal of Applied Sciences, 2005.
12. Christos Bouras and A. Gkamas, “A Mechanism for Multicast Multimedia Data with Adaptive QoS Characteristics”, M.J. van Sinderen and L.J.M. Nieuwenhuis (Eds.): PROMS 2001, LNCS 2213, pp. 74-88, 2001.