

Bandwidth Control in Multiple Video Windows Conferencing System

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Abstract

In recent years, technology has made a dramatic move towards multimedia application especially in conferencing system. Nowadays, the development of conferencing systems that involve multiple parties is very encouraging. As a result, multiple video streams over the Internet are needed in order to provide better visualization.

As compared to audio packets, video packets require more bandwidth during transmission. In order to avoid packet loss or network congestion problem, it is important to have good control of bandwidth in Multimedia Conferencing System that has multiple video windows.

Introduction

Recently, network links have been replaced with more broad ones because of progress of network technologies and lower pricing of broad links. Streaming technology was developed for such network environment to transmit real-time continuous audio and video data.

Both audio and video streaming are the main functions in any multimedia conferencing system to provide good visualization. In order to allow multiple parties to join the conference, streaming of multiple audio and video is a must. It sounds easy but actually it does bring up some problems to be discussed especially during the transmission process. Therefore type of protocol using is important in this purpose.

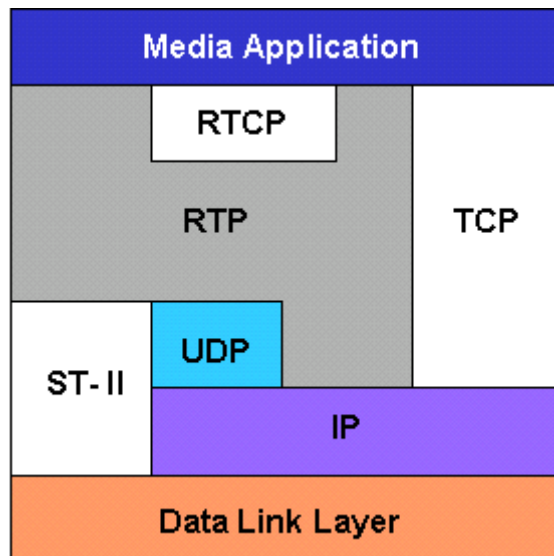


Figure 1: Protocol Architecture

The figure above shows the relationship of the protocols used in routing data over the network, at the highest level the application level protocols communicate between applications such as the client and server application. The application protocols make up part of the data that is transported between hosts inside UDP or TCP network packets or datagrams these network

datagrams are controlled by a set of lower level network protocols. These packets are routed to the correct address using the Internet Protocol (IP).

Typically in a video conferencing scenario a protocol such as RTP will identify the data type being transported, in this example it might be MPEG compressed video, the RTP header will also contain additional information such as timing and synchronization. The RTP header will be 'packaged' along with the video data itself inside a UDP datagram, the UDP datagram will have some error checking services. UDP datagrams are sent out in a stream from sender to receiver, and are routed around the Internet or network by IP (Internet Protocol). On receipt of a UDP datagram, the RTP information and video data are extracted. The RTP header identifies the data as being of type 'MPEG video' to the software application that is playing the video data. The software application will also read the other variables inside the RTP header, the video data will then be used as needed.

It is important to understand the structure of network protocols and its usage because some layers of protocols are needed in bandwidth control. For example, sequence number in Real Time Control Protocol (RTCP) is used to provide feedback of packet loss ratio.

The first part of this paper will be the advantages of having multiple video windows in conferencing system, followed by the drawback of it. The main focus will be in bandwidth usage. Last part presents some of the methods suggested for bandwidth control.

State of the art – Multiple Video Windows in Conferencing System

Advantages of Multiple Video

Users can communicate with many people at the same time – With a multiple video windows conferencing tools, user can collaborate and share information with two or more meeting participants in real-time.

Improve Productivity – As it can bring key people together and keep business moving. From design teams and budget approvals to executive briefings and product launch, video conferencing lets user get things done faster and smarter.

Flexibility – With good control, user are allow to have boardroom meeting of more than 4 parties.

As far as things have been concerned, there also exist some limitations of having multiple video windows in conferencing system.

Drawbacks

On the Internet of today, available bandwidth for each user to receive streaming data is much different and bandwidth even for a user can also be varied with time. It is not realistic, therefore, to transmit a high volume of video data to every receiver with unique transmission rate.

There are few problems of streaming video over the Internet. One of the most serious problems is the high usage of bandwidth. The problem is getting serious if there are multiple video streams transmits over the Internet at the same time.

This happened because the available bandwidth is dynamic. If transmit faster than available bandwidth, congestion and packet loss might occur. Therefore, the video quality will be affected too.

After this problem has been identified, computer scientists have suggested a few methods in bandwidth controlling. Section below stated some of the solving methods that has been proposed as well as implemented successfully in some streaming applications.

Solving method

Multicasting

Multicast techniques are suitable to reduce the load and avoid congestion of network. IP Multicast is a relatively new method of data distribution that is being developed to cater for the growing number of applications that require a single sender to transfer data to multiple users, video conferencing is a good example. The IP Multicast is an efficient standard based solution that extends the IP protocol to cater for multiple user transmission in the same way that a television transmitter broadcasts to a number of homes.

IP works by assigning each host a unique address called an IP address, so that data can easily be transferred to and from individual users. However IP Multicast extends IP by assigning group IP addresses to groups of users. With the standard IP method the same information needs to be transferred across the network several times to each individual user, however with IP Multicast the information needs only to be sent once to the group IP address for all users in that group to receive the data. Below is an example scenario of multicasting:

While conventional packet data is normally sent from one source to one destination, multicast traffic is sent from one source to multiple destinations but without using more bandwidth. With multicast, the source delivers only one packet stream to the switch (for example, at exactly 5 Mbps), and the switch replicates the packets and delivers them to anyone connected to that switch that requests them. In this local Ethernet switch environment, it is rather pointless to worry about bandwidth when everything is happening at wireline speed.

Modern Ethernet switches replicate multicast packets locally without using any additional uplink bandwidth. As a result, sending 5 mbps to every user will have the same network load as sending 5Mbps to one user.

IP Multicast is an important advance in IP networking because it uses network resources and bandwidth more efficiently, which is very important for audio visual data transmission.

Rate Control

To solve bandwidth problem, there is another method called Rate Control. It can be performed at sender side or receiver side and consists of 2 main functions, first is the estimate the available bandwidth, and the second is to match video rate to available bandwidth.

There are two ways to estimate bandwidth. First is by using Probe-based method. The basic idea of Probe-based method is by using probing experiments to estimate the available bandwidth. It adapt sending rate to keep packet loss rate less than a threshold rate. If the packet loss rate less than threshold rate, then increase transmission rate, else decrease transmission rate.

Second way is estimating bandwidth by using Model-based method. The goal of this method is to ensure fair competition with concurrent TCP flows on the network. The basic idea of it is model the average throughput of a TCP flow and transmits video with the same throughput as if it was a TCP flow.

When it is run in the sender side, it explicitly adapts the video rate and the feedback from the receiver is used to estimate the available bandwidth. The feedback information includes packet

loss rate. Whereas when it is run in receiver side, Sender codes video with scalable or layered coder and sends different layers over different multicast groups. Then each receiver estimates its bandwidth and receives an appropriate number of layers up to its available bandwidth.

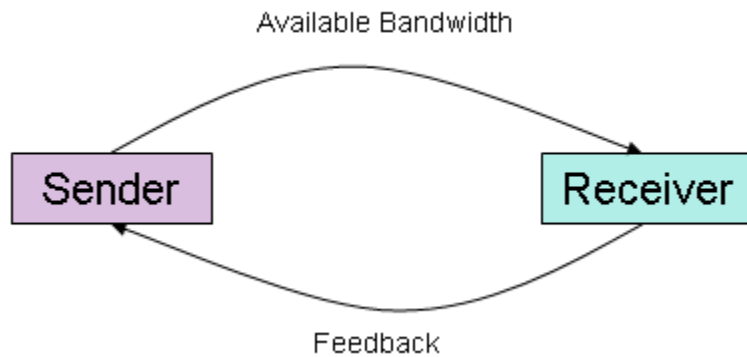


Figure 2 : Source-based rate control

The method used for rate control in receiver side is Receiver-Driven Layered Multicast (RLM). In implementing this approach, sender encodes the video signal, splits it into multiple layers of data, and transmits them for each multicast group. The video quality of the receiver will be improved as the amount of the receiving layers. The receiver can change receiving rate by adding and dropping layers. So, the receiver can adjust receiving transmission rate to the available bandwidth. Initially, a receiver receives only lowest layer. And the receiver adds layers in turn. When network congestion exceeds a threshold and packet loss occurs by adding a layer, the receiver drops the highest layer.

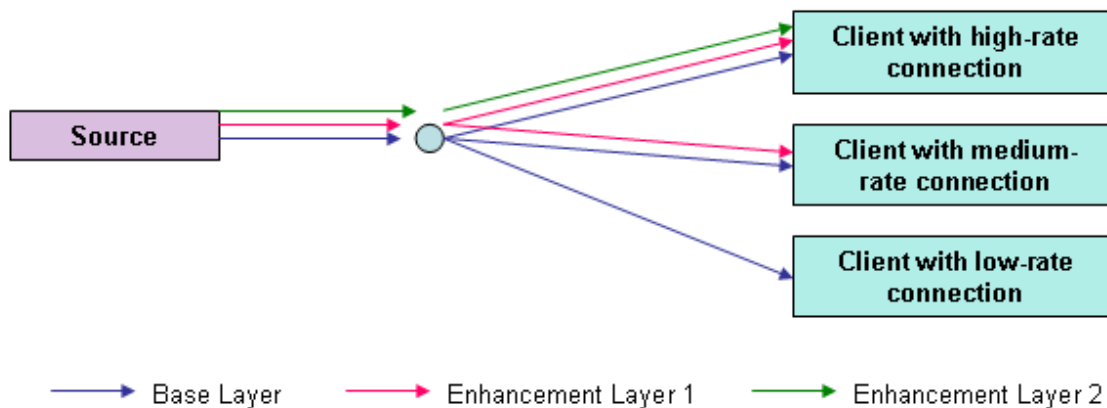


Figure 3 : Receiver-based rate control

However, RLM is not effective for multiple video data streams transmission, because each receiver's control for video data streams is done independently. That is, if packet loss occurs by adding a layer for a receiver, other receivers may drop layers without finding packet loss occurs.

To overcome this limitation, there is a new method to control bandwidth of multiple video data streams sent for each receiver. This method is based on RLM and utilizes priority information on the video quality from the senders. This priority specifies which video data streams should be received with high quality, and a receiver uses RLM for the video data stream. Other video data streams should be received with the lowest quality by selecting the lowest layer only. Therefore,

the video data streams of low priority have constant transmission rate, so they do not influence adjusting transmission rate of the video data stream of high priority.

When priority is changed, the receivers who are receiving high priority video data stream before the priority change can take next action by selecting from the following two actions First is the receivers switch their receiving video data stream to follow high priority video data stream, so that even the receivers who have few decoders can always receive high quality video. Another alternative is the receivers continue to receive the same video data stream after the priority change.

Preemptive Bandwidth Allocation Protocol

For transmitting high quality video and audio streams over networks, several bandwidth reservation schemes have been proposed. For efficient utilization of limited bandwidth, it is desirable that the bandwidth allocated to each stream can be flexibly decreased or increased according to the congestion of networks, keeping the minimum requirement of the quality of services (QoS) of each stream.

The main function of the preemptive bandwidth allocation protocol is to preempt bandwidth if necessary from the current streams along the path. In this protocol, multicast tree evolves dynamically, depending on end-user requirements. The tree construction is based on the shared tree paradigm in the sense that the receivers do not get their connection from the source but from an intermediate node in the tree, called Rendezvous Point.

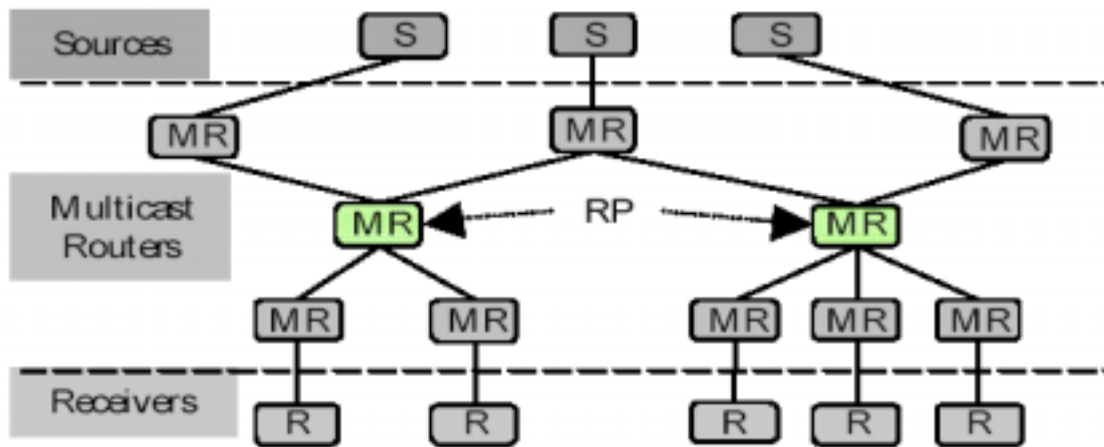


Figure 4 : A shared-tree with multiple Rendezvous Points [3]

To facilitate preemption, end-users has to provide the needs by specifying minimum and maximum requirements in addition to a priority level. This priority level can either be negotiated at the beginning of the session or can initially be set to the lowest value. On each link, two admission controls will be applied to ensure the bandwidth available satisfied the minimum requirement of bandwidth. First admission control will check if there is enough available bandwidth on a given link along the candidate path that satisfied the new node's minimum bandwidth requirement. If first admission condition does not succeed, means that not enough bandwidth is available. Therefore, second admission has to be checked.

Second admission is applied when there is not enough bandwidth from a given link. This admission will check if there is any possible bandwidth preemption from other streams that

traverse the link and that can specify the new users minimum bandwidth requirement. If succeed, the amount of preemptable bandwidth that each streams has to release is calculated. This is to satisfy the users minimum requirement and minimize the total amount of quality lost. If it failed, means that the path cannot offer the users minimum bandwidth requirements even when using preemption.

Conclusion

Transmit of video packets require higher bandwidth compared to other packets. Therefore bandwidth management carries a very important role especially when transmitting multiple video streams through network. Even though there are so many bandwidth allocation approaches have been proposed. The complexity in controlling multiple streams has made the work complicated and that is the reason of performing a well planned bandwidth control in order to develop a good conferencing system.

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