MHTP - A Multimedia High-Speed Transport Protocol

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Abstract - In considering transport layer protocol for multimedia applications in high speed networks, several characteristics of multimedia applications have to be considered. These characteristics include high speed transmission capability, real-time constraints, large bandwidth required by applications, and packet lost of some percentage is acceptable in some multimedia applications. In this paper, we propose a transport layer protocol for high speed multimedia applications, namely MHTP (Multimedia High-speed Transport Protocol), to resolve problems we will face in interworking multimedia systems using high speed networks.

1. Introduction

In the era of high speed networking, people usually ask the question of what applications require such a high transmission capability and *multimedia* often comes out as the answer. There have been many research works [1 - 6] in designing transport protocols for high speed networks, but none of them have taken into account the characteristics of multimedia applications in their designs.

Four major characteristics of multimedia applications have to be considered in the design of network protocols. They are (1) real-time constraints, (2) large bandwidth required (eg., video), (3) synchronization among media, (4) a certain percentage of packet lost is acceptable and (5) packets in the same connection may require different reliabilities. The first three characteristics are easy to understand and only the last two deserve further explanations.

An example to explain the fourth characteristic is the voice traffic. It has been shown in many experiments that digitized voice packet can suffer a packet lost probability up to 2 percent without hurting the quality [7,8]. The packet lost may be due to packet error, packet arrival exceeding the real-time constraint, or buffer overflow. This characteristic allows the receiver not to require retransmissions of some packets with error as long as the packet lost probability is within the acceptable range.

Lastly, let us take the layered video coding technique [9] as an example to explain the characteristic that packets in the same connection may require different reliabilities. In a 2-layer video coding, the video data is classified into two categories. The first category of data constructs the fundamental structure of the video frame and the second category of data can improve the quality of the video frame constructed by the first category. Under such a coding technique, if the second

category of data is lost, the quality of the video is still acceptable with a minor degradation. However, the first category of data cannot be lost since it constitutes the backbone of the video frame. Hence, in such an application, the first category of data requires an error-free transmission while the second category of data can suffer packet lost. This video coding technique allows MHTP to drop packets of second category when a congestion situation is detected such that some packets have to be dropped. In order to achieve this function, MHTP should provide a sequence numbering mechanism to identify which packets require reliable transmissions and which packets do not require reliable transmissions in the same transport connection. This feature has never been provided in previous works.

There are other design issues for multimedia transport protocols such as (1) the ease of add/drop a medium during a connection without interrupting the transmission of other existing media transmission, (2) the handling of packets which fail to meet the real-time constraint due to congestion or packet error, and (3) the admission control of connections before connection establishment. This paper presents detailed design issues of MHTP, which include the layering concept, the packet format, the connection management, the error control, the flow control, the acknowledgement mechanism, and other considerations.

2. The Conceptual Layering and packet format of MHTP

MHTP, in abstract, consists of a multimedia connection manager and a list of sub-protocols as shown in Fig.1. Each sub-protocol will establish a unique sub-protocol connection to transfer only one type of media. However, these sub-protocols all belong to the same multimedia connection manager of a multimedia application. For example, a multimedia application may want to transmit voice and text media across the network. During the connection establishment, the application may specify that the voice medium requests a connectionless service while the text medium requests a connection-oriented service. Thus two sub-protocols, voice sub-protocol and text sub-protocol, will possess different management activities but still belong to the same multimedia application and the same "multimedia connection". The other reason why MHTP consists of a list of sub-protocols is that we can add/drop any sub-protocol to/from a multimedia connection arbitrarily [10].

Fig. 1 The MHTP conceptual layering

The MHTP packet format is presented in Fig.2, SOURCE PORT and DESTINATION PORT, similar to that of TCP, are used to identify the multimedia connection (application). SEQUENCE NUMBER field presents the "byte" sequence number of transmitted media traffic as also used in TCP for the convenience of internetworking. Since a multimedia connection may contain several sub-protocols and each sub-protocol manages its own sequence numbering; hence, more than one sequence numbering may be contained in a single multimedia connection.

SP MEDIA TYPE field(SP means Sub-Protocol) contains MEDIA, C/D, COUNT, and SPID subfields as shown in Fig.3(a). MEDIA field specifies the type of medium (eg., video, voice, text, and image etc.) of the sub-protocol connection. C/D flag is used to indicate that the current packet is either a data packet or a control packet. COUNT field is designed to indicate that how many sub-protocols are contained in one multimedia connection. Since there can be more than one sub-protocol connection for one type of media, SPID (Sub-Protocol Identifier) field is used to distinguish among them. The recognition of a specific sub-protocol can be achieved according to SOURCE PORT, DESTINATION PORT, MEDIA, and SPID.

The QOS(Quality of Service) field[11] is to specify the quality of service requirements of this sub-protocol as shown in Fig.3(a). BANDWIDTH field allows the application to declare the required bandwidth for the application. This field is useful in admission control before connection establishment. ERROR field is set in the outgoing packets to notify the remote application about what degree of packet lost probability this sub-protocol can tolerate. Different media types and different applications may require different levels of packet loss probability. DELAY field is used to specify the real-time constraint of this sub-protocol connection.

OPTION MODES field (see Fig.3(b)) contains different QOS control flags, which are used to indicate which control activities should be activated to satisfy the requirements of diverse multimedia applications.

Fig. 2 The MHTP packet format

a. SP MEDIA TYPE field and Quality of Services (QOS) field

c. Control flags in CODE BITS field

Fig.3 SP MEDIA TYPE field, QOS field, OPTION MODES field and CODE BITS field.

DC (DCHECK) flag is set to 1 to disable the checksum calculation. For example, since the fiber-optic transmission bit error rate is as low as $10^{-9} \sim 10^{-12}$ level and some types of media traffic (e.g. voice) can tolerate some damaged packets, these applications can set this DC flag to disable checksum calculation to gain higher transmission speed and better performance.

DE (DERROR) flag is set to 1 to disable packet retransmissions for packets with error. DE flag is required since it is possible for a sub-protocol to operate with DC=0 and DE=1 (e.g. voice applications). That is, even if a checksum error is found in a packet, the sender will not retransmit it. DF (DFLOW) is set to 1 to disable the flow control; hence, the sender transmission rate will be unconstrained.

RE (REAL-TIME) flag presents whether the packet has a real-time constraint or not and needs to specify what degree of maximum acceptable delay in DELAY field when RE=1. BURST flag and RATE flag indicate that whether the BURST SIZE field and RATE field are valid in current packet. The use of FA (FAST CONNECTION SETUP) and LESS (CONNECTION-LESS) flags will be described in section 3.1.

CODE BITS field (see Fig.3(c)) contains several control flags for connection establishment, data transfer schemes, connection closing, acknowledgement schemes and other features. More details will be described in section 4.2.

RELIABILITY PACKET NUMBER field is to allow unreliable transmission of some specified packets in connection-oriented service such as layered video coding. This field and the FRAG flag in OPTION MODES fields will be described in section 4.3 in details.

There are three ACKNOWLEDGE NUMBER fields to acknowledge the received data packets from the receiver. BURST SIZE field and RATE field, used for flow control, are often used by the receiver to notify the transmitter about the transmission rate and burst size which the receiver is willing to accept. More details about these ACK fields, BURST SIZE field and RATE field will be discussed in sections 4.1 and 4.2. CHECKSUM field is used to check the entire packet, including the header and the user data. CHECKSUM field is placed at the trailer of the packet to achieve the parallel processing purpose and to speed up the transmission [4].

3. MHTP Connection Setup

3.1 MHTP Multimedia Connection Setup Management

MHTP is designed to offer three kinds of connection services by FA and LESS flags in Option Modes field as shown in Fig.3(b).

. FA=0, LESS=0 : Connection-Oriented Service (3-way handshaking based)

. FA=0, LESS=1 : Connection-Less Service

. FA=1, LESS=0 : FAST Connection Setup and Release Service

Since the functions required at the transport layer can vary substantially in order to achieve good performance in different environments, the transport protocol should support several mechanisms to flexibly adjust their processing. Thus MHTP provides diverse connection services selectivity to sub-protocols in the same multimedia connection.

More clearly, because different sub-protocols require diverse connection services, MHTP supports both connection-oriented service and connection-less service. Let us take video phone application, which transmits video and voice traffics, as an example. Since the voice traffic can tolerate a larger percentage of packet lost than that of video, the application might allocate one sub-protocol for video using connection-oriented service and another for voice using connection-less service. Fast Connection Setup scheme is also included in MHTP to reduce the overhead introduced by 3-way

handshaking. Under this scheme, the connection is opened when the first setup packet is received and is closed under timer control.

3.2 Admission Control

In a multimedia high speed network, many traffic sources are bursty and require a large bandwidth with the real-time constraint, we should not accept a new connection to join the network without appropriate admission control. If admission control is not exercised, the number of accepted connections might be too high such that the required quality of applications might be jeopardized because of the real-time constraints and bandwidth requirements. Hence, the effective approach is to control traffic flow at entry points to the network.

Initially, when a new multimedia connection requires an admission, MHTP needs the information about the bandwidth requirement, the media types and the real-time constraints from the applications for the end-to-end admission control. Granting the admission or not depends on issues such as the processing power, buffer size, the acceptable packet loss probability, the bandwidth required, and the acceptable transmission delay. A survey of techniques of admission control can be found in [12], we will not repeat this issue here.

4. Data Transfer

4.1 Flow Control

It has been observed that the major source of the packet loss in high bandwidth networks comes from a link overrunning a node when the packets arrive in burst instead of the transmission bit error. Moreover, in long distance transmission with high bandwidth, sliding window flow control is not a good candidate since the window size will be too large to saturate the buffer of the receiver [4]. In order not to waste the bandwidth and to provide better throughput, rate control is needed. The flow control mechanism in MHTP is based on the "rate control", which is described below.

Rate control is also used to restrict the size and time spacing of bursts of packets from the sender. Within a small time period, the number of bytes that the sender can transmit must not exceed the processing rate of the receiver. Otherwise these packets will be dropped.

In Fig.2, BURST SIZE field and RATE field are used for the purpose of rate control. The value of RATE field (bytes/sec) means that how fast the data can be transmitted by the sender and the BURST SIZE field means the maximum amount of data which might be sent in a burst of packets. The values of both fields are provided by the receiver to control the transmission rate of the sender.

Burst size also indicates the buffer size the receiver can process within a time interval, which is called "burst interval". The length of a "burst interval" is given by the "burst size" divided by the "rate" (burst size / rate = burst interval (sec/burst)). In other words, the maximum number of bytes that can be transmitted in each burst interval is specified by the burst size. Thus, the sender will send packets at its peak rate until it has sent the agreed burst size within one burst interval. It will then wait the specified waiting time determined by the burst interval before transmitting the next burst. Based on the number of active connections and available buffers, the MHTP receiver will assign appropriate burst size and rate to its corresponding sender (see Fig.4).

* RATE = 1Mbytes/sec
 * BURST SIZE = 100Kbytes/a Burst of Packets
 * Burst Interval = 0.1 sec/a Burst of Packets = BURST SIZE/RATE

* 0.1 sec = Burst Interval.

Fig. 4 Rate Control Issue Explanation

4.2. Retransmission Mechanism

In MHTP, due to considering different characteristics of multiple types of media, the retransmission mechanism is designed to combine the Go-back-N and the Selective Retransmission schemes to satisfy different requirements and diverse transmission conditions of multiple type of media packets. As shown in Fig.2, the MHTP packet format contains three ACKNOWLEDGE NUMBER fields and two control flags in CODE BITS field (GO-B flag and SELE flag):

RETX BEGIN ACKNOWLEDGE NUMBER field and
RETX END ACKNOWLEDGE NUMBER field:

For selective retransmission to specify the beginning and ending sequence number of the retransmitted data sources.

GO-BACK-N ACKNOWLEDGE NUMBER field:

For Go-back-N:

- (1) To specify the beginning sequence number of the retransmitted traffic (NACK flag = 1, GO-B flag = 1).
- (2) To acknowledge the received data by the receiver (NACK flag = 0, GO-B flag = 1, SELE flag = 0).

We give an example to explain how to use these fields and flags. In Fig.5, we present the method which combines the "Go-back-N" and "Selective Retransmission". In case (a), the receiver does not receive the data whose sequence number is from 3001 to 5000 due to packet lost or packet error. However, the receiver has received the data with sequence number from 5000 to 10000. Moreover, the data with sequence number greater than 10000 are all lost because of buffer overflow. In this case, we set the RETX BEGIN ACK

NUMBER field = 3001 and RETX END ACK NUMBER = 5000 to perform the selective retransmission. Further, MHTP also performs the Go-back-N retransmission from 10001 by giving GO-BACK-N ACK NUMBER field = 10001. Cases (b) and (c) explain Go-back-N retransmission and selective retransmission respectively.

The retransmission mechanism of MHTP can concurrently satisfy real-time constraints by selective retransmission and save the buffer storage to avoid occupying the buffer for an unacceptable time period by Go-back-N method. Therefore, MHTP is more flexible than other transport protocols in retransmission mechanism and can achieve better performance in the multimedia high speed network.

(a) Go-back-N & Selective Retransmission

(b) Go-back-N

(c) Selective Retransmission

Fig. 5 The Mechanism of Combining "Go-back-N" and "Selective Retransmission"

4.3 Sequence Numbering

One important feature of MHTP is that both reliable and unreliable data transferring can be supported in a single sub-protocol connection. Under such a scheme, the users can require some packets to be transmitted reliably and some unreliably through the same sub-protocol connection as mentioned earlier.

Although MHTP provides the ERROR field to specify the packet lost probability the sub-protocol can tolerate, it cannot be used to require the retransmission of a specific packet. However, once the packets are lost, how can the receiver find out that the lost packets require reliable or unreliable transmissions? For example, assuming the receiver receives packets 1, 2, 3 and 5 with packet 4 being lost, the receiver has no way of judging whether packet 4 is a reliable packet (which needs a retransmission) or an unreliable packet (which does not need a retransmission). In order to resolve the unreliable problem, we require a field, namely "Reliability Packet Number"(RPN).

In MHTP, all reliable packets have an odd RPN starting from 1 and all unreliable packets have an even RPN starting from 0. Further, each odd RPN corresponds to exactly one reliable packet while an even RPN can correspond to several unreliable packets. As illustrated in Fig.6, a session consists of 8 packets and only the packets with the sequence number 2001-4000, 6001-7000 and 9001-10000 are the reliable packets such that their RPN numbers are 1,3 and 5 respectively. Other packets with the even RPN number are all unreliable packets. In Fig.6, if the packets with the sequence number 1001-2000(RPN=0), 2001-4000(RPN=1), 4001-5000(RPN=2) and 7001-8000(RPN=4) are received, then the receiver knows that a packet with RPN=3 is lost and which requires a retransmission since the RPN is an odd number. Then a packet requests the retransmission with sequence number 5001-7000 will be sent by the receiver and the sender will retransmit the packet with sequence number 6001-7000.

Sequence Number :							
1001-2000	2001-4000	4001-5000	5001-6000	6001-7000	7001-8000	8001-9000	9001-10000
Reliability Packet Number (RPN) :							
0	1	2		2		2	
4	4	2					

Fig.6 An example to explain the usage of Reliability Packet Number (RPN)

In addition, due to the "byte-oriented" issue is adapted for MHTP, MHTP packet format has a FRAG flag in OPTION MODES field, namely Fragmentation bit, to indicate the current packet is fragmented or not. If the packet is fragmented into several packets, this FRAG flag must be set in these fragmented packets. More importantly, these fragmented packets will also possess the same RPN in order to inform the receiver that these packets with the same RPN number initially belong to a single packet.

5. Conclusions

In this paper, a multimedia high speed transport protocol is proposed which considers the characteristics of both high speed transmission and multimedia applications. Since the

characteristics of multimedia applications are quite different from traditional data transmission, some design issues have to be considered in the protocol such as real-time constraints and limited level of packet loss probability. Since more issues are considered in this protocol which makes the protocol more complex, a fixed header format is proposed for easier VLSI implementation.

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