The Research on RTCP Feedback Mechanism with Network Tomography

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***Abstract*—The Network Tomography technology based on RTCP feedback for network monitoring and management is significant. The current RTCP feedback mechanism is defined in RFC3550. In this paper, the RTCP Extended Report (XR) feedback mechanism has been studied. The paper analyses the relationship between the feedback interval and RTCP packet size and the size of the multicast session, points out the shortage of the current feedback mechanism, and puts forward animproved encoding method. Based on the RTCP unicast feedback model, using reasonable encoding XR packets can reduce the feedback bandwidth and promote the feedback rate.**

Keywords- RTCP; feedback mechanism; XR packet; encoding

I. INTRODUCTION

With the rapid development of network technology, transmitting real-time audio and video streaming in the IP network has become a concerned network application. The standard protocols for real-time audio and video streaming transmitting include the Real-Time Transport Protocol (RTP) and the Real-Time Transport Control Protocol (RTCP). RTP is the Internet standard published by the IETF, the current version of the document is RFC3550 [1]. RTP is only responsible for real-time data transmission;the main function of RTCPis providing periodic reports for the reception quality feedback and flow controlling. Traditional RTCP feedback model is that all the receivers periodically send RTCP packets to provide feedback about the current network quality of service with multicast model. With the increasing size of the multicast session, for not occupyingtoo much network resources,RTCP packets may maintain a fixed proportion of the bandwidth.However that will extend the RTCP packet transmission interval and reduce the feedback rate.

Because of the limit of the RTCP feedback mechanism mentioned before, IETF proposed the RTCP unicast feedback which suits a single source session with a single sender such as SSM. In the RTCP unicast feedback model, the feedback target is between the receivers and the source, The Feedback Target (FT) is a logical function which may be integrated with the Source in the same entity or disjoint from the source. Receivers send RTCP packets to the FT, after that the FT reflects or combines the feedback packets and multicasts forwarding to the multicast group. In the RTCP unicast feedback model, using the RTCP Extended Reports (XR) can provide more detailed network monitoring informationto the source and reduce the network bandwidth and catch up well feedback effects.

II. EXTENTED REPORT ANALYSIS

The RTCP implements the quality feedback and network monitoring functions by sending Sender Report (SR), Receiver Report (RR) and Extended Report (XR) [2]. The three main feedback reportsform a compound RTCP packet combined with Network Tomography [3] technology. The XR packet includes seven extended report blocks: Loss RLE Report Block, Duplicate RLE Report Block, Packet Receipt Times Report Block, Receiver Reference Time Report Block, DLRR Report Block, Statistics Summary Report Block, VoIP Metrics Report Block.

The XR provides more detailed feedback in several categories. Loss RLE reports use run-length encoding (RLE) to allow receivers to report on loss of individual data packets, providing the sender with a complete view of which packets arrived. Packet-receipt time reports allow the receiver to report reception times of individual packets, giving a more detailed view of packet timing than standard jitter reports and other time information. Figure1 shows the XR network tomography message format including the loss RLE report block [4] and the Packet Receipt Times Report block.

BT=1	Rsvd.	T	Block length
SSRC of source			
Begin_seq			End_seq
Chunk 1			Chunk 2
......			......
Chunk n-1			Chunk n

a. Loss RLE Report Block

BT=3	Rsvd.	T	Block length
SSRC of source			
Begin_seq			End_seq
Receipt time of packet begin_seq			
Receipt time of packet (begin_seq +1) mod 65536			
......			
Receipt time of packet (end_seq -1) mod 65536			

b. Receipt Times Report Block

Figure 1. Experiment environment

The loss RLE report block type permits detailed reporting upon individual packet receipt and loss events. With Network Tomography, one application could discover the topology of the multicast tree used for distributing a source's RTP packets, and of the loss rates along links within that tree or could

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manage the network. The Packet Receipt Times Report block type permits per-sequence-number reports on packet receipt times for a known source's RTP packet stream. Such information can be used to measure partial path characteristics and to model distributions for packet jitter.

Loss RLE report is composed by the fixed 12 Bytes header and a number of packet loss report blocks. The size of the loss RLE block depends on the number of packets lost and packet loss pattern and the network status. In the best case, we consider there was no loss. So that, the loss RLE block can be omitted. In the worst case, packets are lost uncertainly. The encoding itself consists of a series of 16 bit units called chunks that describe the Loss RLE report blocks' packet loss information. There are three kinds of encoding formats about chunks. Each chunk either specifies a run length chunk or a bit vector chunk, or is a null chunk.

1) A run length chunk describes between 1 and 16,383 events that are all the same, either all receipts or all losses.

2) A bit vector chunk describes 15 events that may be mixed receipts and losses. Typically network status is uncertain, so a bit vector chunk encoding forma is used often. A bit vector chunk is divided into 15 pieces, each piece has 2^{Tl} (0 <Tl<15) packets. Tl is the constraint parameter, one for each 2^{Tl}packets is reported, use the Boolean value indicating that the serial number of the reception of packets, 0 loss, 1 for normal reception.

3) A null chunk, an empty chunk, which size is 0, ignored by the application.

Packet Receipt Times Report reports on packet receipt time for each packer arriving the receiver， so you can calculate the delay time, and can measure the status of a particular path segment. Packet Receipt Times Report does not use RLE encoding, but uses dilution parameters Tr, that is, does not report each packet's arriving time, but one reporting for each 2^{Tr} packets.

III. EXPRESSION DERIVATION

A. Loss RLE Reports Block Size Expression

Loss RLE Reports Block is divided into 15 pieces in a chunk, each chunk length is 2 Bytes, so the Loss RLE Report length$L_{LossRLE}$can be expressed as: 12+2*γ(γ is chunk number). Without compression encoding, chunks reports for every RTP packets, assuming the number of RTP packets gonging reporting is the K(i), the chunk number γ=K(i)/15, then $L_{LossRLE}$=12+2*K(i)/15;

To report the RTP packet by once per 2^{Tl}packet, the chunk numberγ =2*K(i)/(15*2^{Tl}), then $L_{LossRLE}$ =12 +2*K(i)/(15*2^{Tl}).

So that loss RLE report packet length is calculated as:

$$L_{LossRLE} = 12 + 2 \times \frac{K(i)}{15\times 2^{Tl}} = 12 + 2 \times \frac{V(i)*T}{15\times 2^{Tl}} \qquad (1)$$

B. Packet Receipt Times ReportBlock Size Expression

Packet Receipt Times Report includes the fixed 12 Bytes header and a number of blocks composed of a block length of 4 Bytes, if the number of the RTP packets to be reported is K(i),assuming every packet to be reported, then all the block length is 4*K(i). Packet Receipt Times Report length could be expressed as $L_{Receipt}$= 12+4*K(i). If one report for every 2^{Tr} packets, then the block number is K(i)/2^{Tr}, therefore $L_{Receipt}$ = 12+4*K(i)/2^{Tr}. So that the Packet Receipt Times Report length is calculated as follows:

$$L_{Receipt} = 12 + 4 \times \frac{K(i)}{2^{Tr}} = 12 + 4 \times \frac{V(i)*T}{2^{Tr}} \qquad (2)$$

C. RTCP Report IntervalExpression

In RFC3550,only a small part of a fixed portion of the session bandwidth distributes to RTCP control traffic in a RTP session, usually 5%. 25% of the RTCP bandwidth is distributed to transmit data of the participants. When senders share the ratio ofthe number of participants greater than 25%, the sender's RTCP bandwidth is corresponding to the proportion. Assuming the total membership of the session is R, the number of sender is S, Receiver number is N, the average RTCP packet size is L_{RTCP}, the session bandwidth is M, the RTCP bandwidth is M_{RTCP}.

The RTCP report interval T [5] is calculated as follows:

if senders < (25% of total number of participants)
then {
sender Interval = L_{RTCP}*S/25%*M_{RTCP};
receiver Interval = L_{RTCP}*N/75%*M_{RTCP};
}
else { Interval = S_{RTCP}*R/M_{RTCP}; }

So$T = \frac{N\times L_{RTCP}}{0.75\times f\times M} = \frac{N\times L_{RTCP}}{0.0375\times M}$

IV. EXPERIMENTAL SIMULATION AND ANALYSIS

For a 2 Mbps MPEG-2 [6] media streaming, assuming that each RTP packet size is 1400 Bytes, then RTP packets sending rate per second is about 2048k/(1.4k*8)=183. In the session, the media sender is generating the SRwith a SDES, the SR packet size is L_{SR}=56 Bytes; FT forms a packet into RSI packet, in accordance with the application to set its only contains RTCP Group and Average Packet Size Sub-Report Block, indicating the current average RTCP packet size and the number of receivers in a multicast session, RSI packet length is L_{FT} = 28 Bytes; if 200 receives in the session, given the definition of the relevant parameters are as follows:

TABLE I. MEANING OF PARAMETERS

parameter	meaning
T	RTCP report interval（Sec.）
K(i)	RTP packet number in the interval
V(i)	RTP sendingrate
L_{RTCP}	average RTCP packet size（Byte）
$L_{RTCP,R}$	CompoundRTCPpacketsize（Byte）
$L_{LossRLE}$	Loss RLE report size（Byte）
$L_{Receipt}$	Packet Receipt Times Reportsize（Byte）
M	Session bandwidth(Bps)

N	Number of receivers
f=5%	RTCP bandwidth
T_r/T_l	Thinning factor for XR,（0，15）

If there was no packet loss, considering $L_{LossRLE}$= 0, the RTP packet sending rate was regarding as the perfect rate, that is V(i)=183. Every receiver accepts 183 RTP packets per second, when Tr=0, 2^{Tr}=1, then the application reports upon each RTP packet about the receipt times, so as one receiver generates 183 XR packet receipt times report packets. According tothe expression (2) described in part III, we know the packets size is (12 +4*183) = 744 Bytes; when Tr = 3, 2^{Tr} = 8, then for a group of every 8 RTP packets to be reported, generating about 183/8=23 receipt times report packets per second. The report packets size is (12 + 4*23) = 104 Bytes; So we see when Tr<4, XR packet is too large, affecting the RTCP packet transmission rate; considering the Tr value as 4,5 ... 14,15, after checking, the effective value includes 4,5,6,7,8. The relationship between RTCP report interval and the packet receipt times report thinning factor Tr shown in Figure 2.

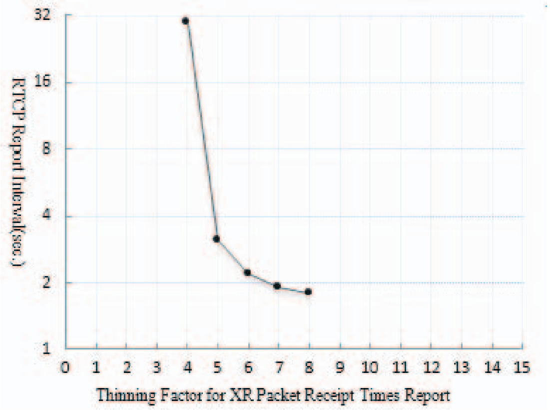

Figure 2. RTCP report interval variations with Tr

Some packets may be loss when congestion occurs in the network. The RTP packet sending rate is lower than the perfect rate, assuming that RTP rate V(i) = 170. Setting the Loss RLE report thinning factor Tl value as 0,1,2 respectively, the receipt times report thinning factor Tr were set as 5,6 ... 14,15. Through simulation experiments, the RTCP report interval changes with thinning factor shown in Figure 3.

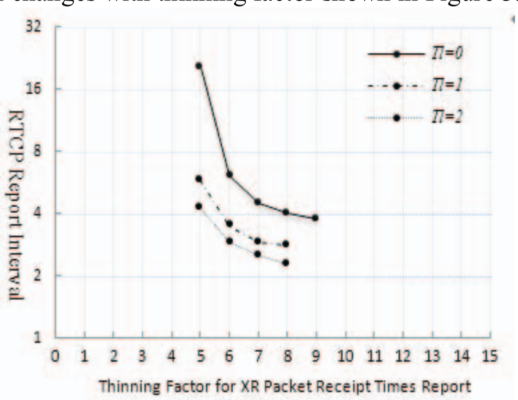

Figure 3. RTCP report interval variations with Tr and Tl

In traditional transfer mode, assuming the average RTCP packet size of 1000 Bytes, while the number of the receiver gradually increased from 200 to 2000, the RTCP report interval grows linearly as the group size increasing. Assuming RTP sending rate V(i) = 150, in the RTCP unicast feedback mode with reasonable compression to XR packets, RTCP report interval was significantly shorter. Figure 4 shows the compared results.

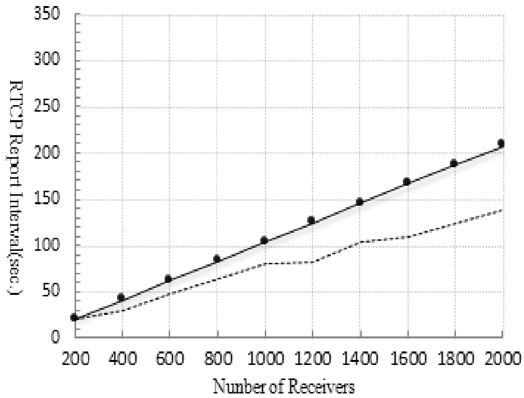

Figure 4. RTCP report interval variations

V. PERFORMANCE ANALYSIS OF DATA

Encoding XR packets using the compression method can effectively reduce the network bandwidth occupied by the feedback information, and improve the efficiency of the feedback. For a 2Mbps video stream, assuming that each RTP packet is 1400 Bytes, then sending RTP packets per second is about 183. When the network is best, if the feedback reported on the XR packet in the arrival time of each packet and reception status of the report, the XR packets length is: (12 + 4*183) + (12 + 2*183/15) = 780 Bytes. When encoding, if 183 packets per second on each of 4 packets to report, the XR packets length (12 +4*183/4) + (12 + 2*183/(15*4)) = 213 Bytes. Through the analysis of the data, we see that the encoding method greatly reduces the feedback message XR packets size and the consumption of network bandwidth. Next step of work, we will combine with the quality control for streaming compression method of monitoring information [7] to further reducing the share of the feedback packet network bandwidth and improving feedback efficiency.

VI. CONCLUSION

XR packets can improve the efficiency of the feedback, but increased the overhead of RTCP packets. Through processingthe reasonable encoding to the XR packets, using appropriate thinning factor can control the size of XR packets. From the experimental simulation results, we know in the RTCP unicast feedback mode, the XR packet compression can effectively reduce the monitoring packet overhead, lower the feedback reporting interval. So combination of using RLE encoding and thinning to control the size of XR packets is well feasible method. Choice of the beginning and ending RTP packet sequence numbers for the trace is left to the application. These values are reported in the block. By default, to ensure good network monitor, XR packets should not be encrypted or filtered.

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