

Traffic Quality Monitoring System between Different Network Providers

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Abstract—In this paper, we propose a network performance monitoring method which can measure delay of each network using the RTCP(Real-time Transport Control Protocol) timestamp fields to detect a trouble section in interworking environment. According to the increment of real-time multimedia service provision, network performance monitoring method is necessarily needed for QoS management of real-time services in the interworking environment between different networks. Multimedia services such as video conference, VoIP (Voice over IP) service and IPTV service are sensitive to the network performance(e.g. delay, delay variance and packet loss etc). In this case, it is very important to detect a trouble section when service quality degradation occurs in the interworking environment. So, we propose the network monitoring method for multimedia services through delay measurement using RTCP in each network. Also, with this method, we can detect and define which network has generated a poor network performance.

Keywords— NGN, RTCP, QoS, monitoring, delay, multimedia service

I. INTRODUCTION

The various access network technologies are integrated through NGN(Next Generation Network). Also various multimedia services(e.g. IPTV, VoD(Video on Demand), VoIP(Voice over IP), Image Conference, and etc) are provided by network service providers in the NGN converged environment. Particularly, QoS provision has become an important issue because multimedia services such as video conference, VoIP service and IPTV service are sensitive to the network performance(e.g. delay, delay variance and packet loss etc). The method to detect which network has generated poor network performance is needed when service quality occurs because various services is provided through the converged network.

Nowadays studies about network performance management have been going on several standard organizations (e.g. ITU-T, IETF, ETSI etc) and many projects(e.g. EuQoS, MESCAL etc). Also, the studies about QoS provision and interworking are carried out with active in convergence network environment. Especially, interworking

architecture and performance measurement study is being progressed in NGN [1] - [4].

When ISPs (Internet Service Providers) provide end-to-end service, network performance management is required for QoS management in interworking environment. Also, network monitoring method is important to detect trouble section between different networks when the problem occurs in QoS. Until now, many studies have been carried out by using active probe and passive probe method in order to estimate RTT (Round Trip Time) of ICMP or TCP. However, active probe method needs time synchronization between end-users, and the existing passive probe method is not suitable for real-time service because method of filtering ICMP and TCP is insufficient in real-time due to these protocol characteristic.

The most common tools for measuring network delay between end-hosts employ the Internet Control Message Protocol (ICMP) using either time exceeded or echo request/reply messages. Tools using ICMP may be useful in network trouble-shooting, but they have well-known limitations for precise delay measurements including the fact that Internet Service Providers often block or rate-limit ICMP traffic, and that ICMP traffic is often given lower priority in routers. Other tools are to use TCP SYN and SYN-ACK connection setup handshaking mechanism to measure delay. But this type of delay measurement is insufficient to estimate network performance to the real-time service.

So we propose the new network performance monitoring method using RTCP timestamp to measure each network delay in interworking point. The network monitoring system has to filter the RTCP packets using passive probe to calculate each network delay. Our network monitoring method has advantages that do not need accurate time synchronization between end points like active probe method, and it can estimate network delay using the RTCP timestamp in real-time service. Also it is easily realized by software function on interworking routers.

The rest of the paper is organized as follows. In section 2, we introduce the related work for network performance monitoring method and indicate its problem. In section 3, we show the proposed method for real-time service and the analysis process of RTCP data in order to estimate network delay. In section 4, we present an analysis of test results.

Finally, we present a conclusion of the paper and future works in section 5.

II. RELATED WORKS

A. General reference network model

Figure 1 shows a general reference network model for performance measurement in an NGN environment. Along the end-to-end path, there are two CPNs (Customer Premises Networks), two access networks, one or multiple core networks, zero or multiple transit networks, and one or multiple service provider networks. The access networks, core networks, transit networks, and service provider networks may belong to the same or different network or service providers. One example service provider is an IPTV service provider of a central head-end or regional head-end.

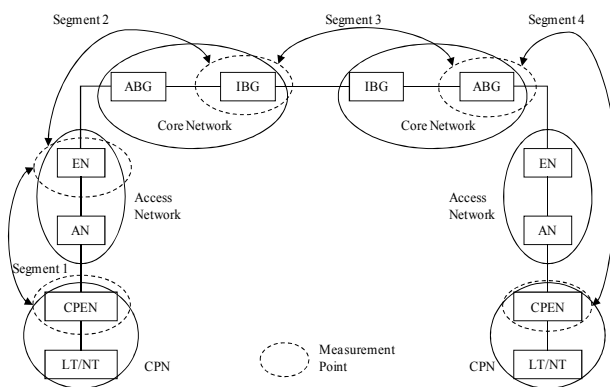


Figure 1. General reference network model and quality measurement point [1]

NGN converged network is defined as environment that is interworked by many network service providers. In this environment, if a problem occurs in using a real-time service (e.g. a video conference), it is important to detect the section of the problem in where network.

The structure of NGN interworking is commonly organized by IX (Inter eXchange) scheme using layer-3 router between different networks. Edge router of each NSP is connected with IX router in interworking point by using a peering transit method. As figure 2, user_1 and user_2 individually are connected with Network_A and Network_B in order to use real-time service (e.g. a video conference). In this case, network performance can be monitored by extraction data from RTCP filtering because typically real-time service uses RTP and RTCP for real-time streaming.

RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee QoS (Quality of Service) for real-time services. The data transport is augmented by a control

protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers. Here, the network quality management system is placed in interworking point and can be constituted by software function on edge router or interworking gateway.

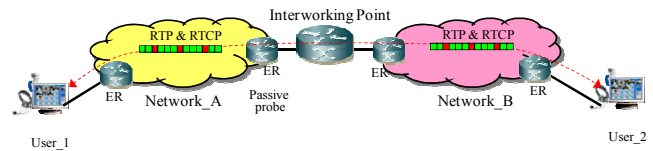


Figure 2. Interworking network model using IX (Internet eXchange)

B. Pre-existing network performance measurement method

Some recent studies describe measurement methods that attempt to infer path characteristics from RTT (Round Trip Times) measurements based on the use of internet Control Message Protocol (ICMP) time-to-live (TTL) and ICMP timestamp options. But Internet Service Providers often block or rate-limit ICMP traffic and that ICMP traffics are often given lower priority in routers [7].

The methods for passive estimation of TCP round-trip times use passive estimation of round-trip times for bulk TCP transfers. The method uses TCP timestamps to locate segments from a bulk data sender that arrive one RTT apart, while the other detects patterns caused by self-clocking that repeat every RTT and can be used throughout the lifetime of a TCP session. However TCP protocol using this method is incongruent for measuring performance to real-time service [7]. So we estimate delay between networks using RTCP.

A previous work [6] introduces a performance measurement method called CoMPACT Monitor, which can estimate user performance in a scalable and lightweight manner. This method only requires counting of the traffic volume (passive monitoring) and simple measurement of thus more feasible and tractable than conventional methods. However, active probe method needs time synchronization between end users to estimate delay. It is inaccuracy that time synchronization is provided by NTP (Network Time Protocol).

Active probe method and Active agent/Passive probe method have been proposed to monitoring network performance [5]. Active probe method generates additive test packets to network delay. But, Active probe method is not suitable to real-time network monitoring because these test packets may impact on network congestion.

C. RTCP (RTP Control Protocol)

The RTP control protocol (RTCP) is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets. The underlying protocol must provide multiplexing of the data and

control packets, for example using separate port numbers with UDP.

The primary function of RTCP is to provide feedback on the quality of the data distribution. This is an integral part of the RTP's role as a transport protocol and is related to the flow and congestion control functions of other transport protocols. The feedback may be directly useful for control of adaptive encodings, but experiments with IP multicasting have show that it is also critical to get feedback from the receivers to diagnose faults in the distribution.

Sending reception feedback reports to all participants allows one who is observing problems to evaluate whether those problems are local or global. With a distribution mechanism like IP multicast, it is also possible for an entity such as a network service provider who is not otherwise involved in the session to receive the feedback information and act as a third-party monitor to diagnose network problems.

RTP is designed to allow an application to scale automatically over session sizes ranging from a few participants to thousands. For example, in an audio conference the data traffic is inherently self-limiting because only one or two people will speak at a time, so with multicast distribution the data rate on any given link remains relatively constant independent of the number of participants.

However, the control traffic is not self-limiting. If the reception reports from each participant were sent at a constant rate, the control traffic would grow linearly with the number of participants. Therefore, the rate must be scaled down by dynamically calculating the interval between RTCP packet transmissions.

The RTCP transmission interval between packets from a session participant should scale with the group size. This interval is called the calculated interval. The calculated interval T is then determined as follows and is based on [6].

1. If the number of senders is less than or equal to 25% of the membership (members), the interval depends on whether the participant is a sender or not (based on the value of RTCP reports). If the participant is a sender, the constant C is set to the average RTCP packet size divided by 25% of the RTCP bandwidth, and the constant n is set to the number of senders. If the number of senders is greater than 25%, senders and receivers are treated together.
2. If the participant has not yet sent an RTCP packet, the constant Tmin is set to 2.5 seconds, else it is set to 5 seconds.
3. The deterministic calculated interval Td is set to $\max(T_{\min}, n \cdot C)$.
4. The calculated interval T is set to a number uniformly distributed between 0.5 and 1.5 times the deterministic calculated interval.
5. The resulting value of is divided by $e - 3/2 = 1.21828$ to compensate for the fact that the timer reconsideration algorithm converges to a value of the RTCP bandwidth below the intended average.

When an RTP or RTCP packet is received from a participant whose SSRC(Synchronization Source) is not in the member table, the SSRC is added to the table. The same processing occurs for each CSRC(Contributing Source) in a validated RTP packet.

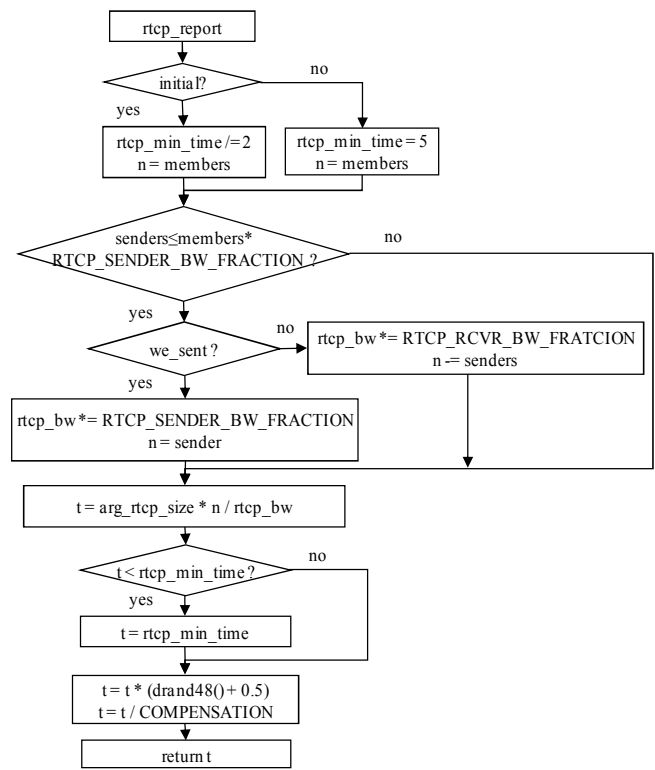


Figure 3. Decision algorithm of RTCP transmission interval[6]

Figure 4 shows example of round-robin time computation using RTCP timestamp. So, we can estimate network delay this property because RTCP is periodically transported according to network conditions.

Let SSRC_r denote the receiver issuing this receiver report. Source SSRC_n can compute the round-trip propagation delay to SSRC_r by recording the time A when this reception report block is received. It calculates the total round-trip time A-LSR(Last Sender Report timestamp) using the LSR field, and then subtracting this field to leave the round-trip propagation delay as formula (1);

$$RTT = A - LSR - DLSR \quad (1)$$

This is illustrated in figure 4. Times are shown in both a hexadecimal representation of the 32-bit fields and the equivalent floating point decimal representation. Colons indicate a 32-bit field divided into a 16-bit integer part and 16-bit fraction part. This may be used as an approximate measure of distance to cluster receivers, although some links have very asymmetric delays.

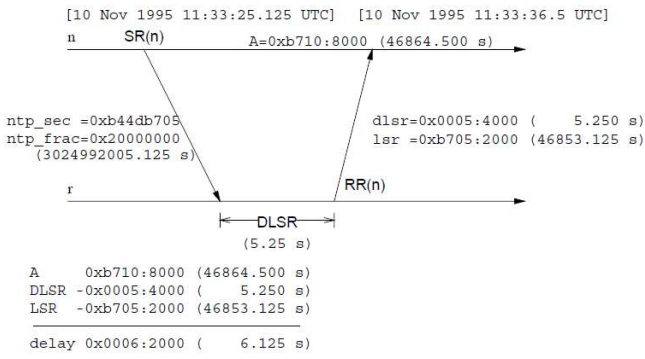


Figure 4. Example for round-trip time computation[6]

III. PROPOSED METHOD TO MEASURE DELAY OF EACH NETWORK

In this paper, we propose the network performance monitoring method which can measure delay of each network using the aforementioned RTCP(Real-time Transport Control Protocol) timestamp fields to detect a trouble section in an interworking environment.

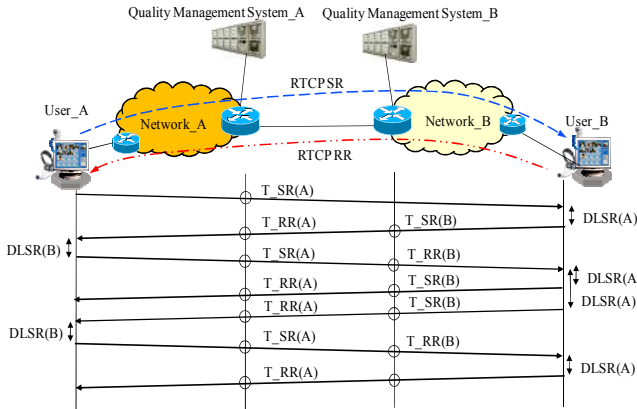


Figure 5. Network environment for delay measurement using RTCP

Figure 5 shows network environment for delay measurement using RTCP. Like figure, each network provider can establish quality management system in interworking point. It can collect RTCP packets and estimate network delay generated in their own network using information of RTCP timestamps.

Network quality management system stores filtering time when RTCP packets used for multimedia service are filtered. Then, stored packets are classified according IP addresses or SSRC Network quality management system can calculate network delay of each network using formula (2). Here, T_{SR} is RTCP SR packet forwarding time and T_{RR} is RTCP RR packet receiving time.

$$\text{Network delay} = \text{RTT}/2 = T_{RR} - T_{SR} - \text{DLSR} \quad (2)$$

The calculated network delay is managed by each network provider and is adopted as decision basis for detection of trouble section when service quality is poor.

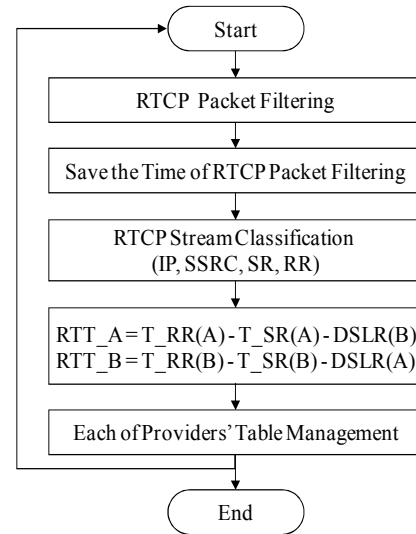


Figure 6. Proposed algorithm for delay measurement of each network using RTCP

IV. TEST RESULTS AND ANALYSIS

In order to measure network delay of each network using the proposed method, we configure test environment between ETRI(Electronics and Telecommunications Research Institute) and CBNU(Chungbuk National University) We use SIP(Session Initiation Protocol) phone which is video telephony terminal for test. SIP phone uses RTCP to monitor and control network condition on calling

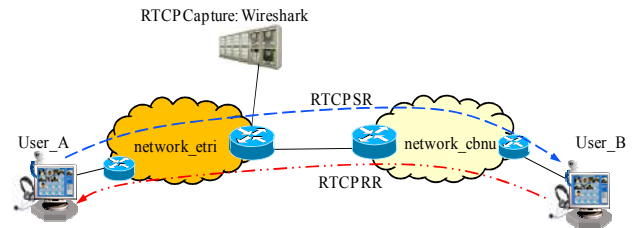


Figure 7. Test environment between ETRI and CBNU

We filter RTCP packets in ETRI access network using Wireshark. Wireshark is one of those programs that many network managers would love to be able to use. Figure 8 shows RTCP capture results. Like figure 8, we know that RTCP packets are separated by SRCC to voice and video streams and are forwarded between end-users.

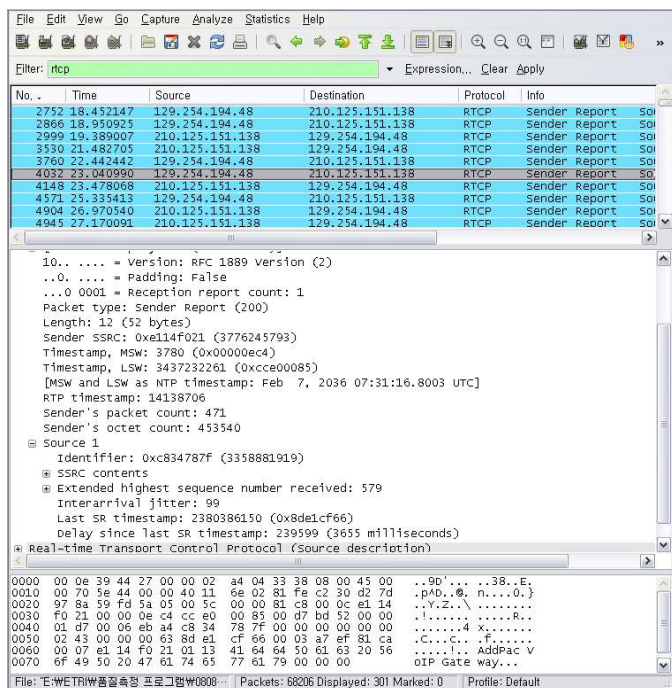
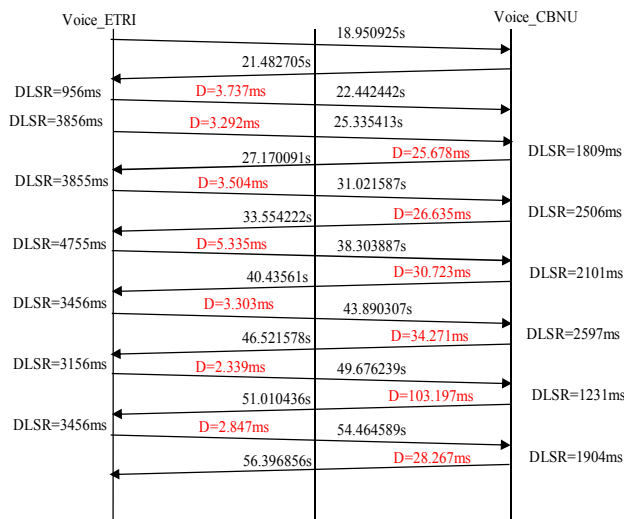


Figure 8. RTCP packet capture results using Wireshark

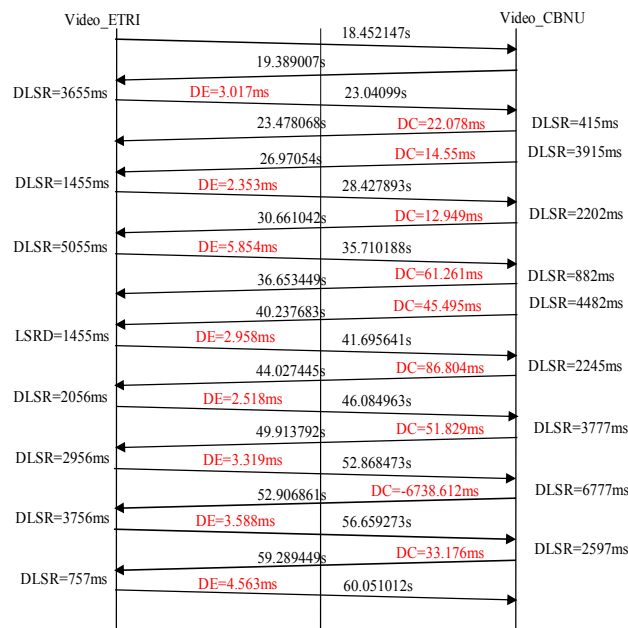
Figure 9 shows delay measurement results of each network using RTCP. Network delay generated in ETRI network section is considerably low because RTCP filtering point is located in ETRI access network.

Therefore, we can detect and define which network has generated a poor network performance and effected service quality degradation.

Video telephony service packets is well forwarded in ETRI section, on the other hand longer delay(6738.612ms) occurs in CBNU section and/or interworking point. This implies that a cause of end-to-end service quality degradation is brought in CBNU section and/or interworking point.



(a) Results of voice stream packets



(b) Result of video stream packets

Figure 9. Delay measurement results of each network using RTCP

Also, we know that delay dispersion of video stream is larger than voice stream's through measurement results analysis of delay about voice and video streams. This implies that video stream is more sensitive than voice stream in network delay. So, buffer size of video steam in router/terminal is assigned larger than voice stream's in order to minimize impact on network delay variation.

V. CONCLUSIONS

The networking monitoring method is needed in NGN environment and plays an important role in quality of service management between different network providers. In order to detect which network is trouble in converged network, we propose the network performance monitoring method using RTCP timestamp to measure network delay in each network. Through this method, we can detect and define the trouble serviced network in time when network problem has been arisen, and easily practice network monitoring with graphically display of estimate results.

However, this method only analyses network performance about real-time service. In NGN (Next Generation Network) environment, various internet services are provided so that it is difficult to measure performance only using RTCP. Now interworking of different networks is constituted by one or more interworking routers in converged network. If NGN network environment is converged by many interworking routers, resource reservation function for RTCP path is needed because it is difficult that expect RTCP path in this network environment. Also if NGN is constituted by the interworking not L3 router but L2 MPLS, needed function or system in

MPLS will add to interworking point. These issues are for further study.

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