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ECN Based Multi-path Mechanism for VoIP Transmission

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Abstract: By the remarkable improvement of physical performance of networks (such as speed, capacity, etc) which consist of Internet and the diffusion of high-end terminals such as smart-phones, voice over IP (VoIP) is becoming one of the major communication technologies. However current packet networks still can not provide a stable transmission quality and it can cause VoIP call quality degradation. Especially, the main factor affecting VoIP quality is congestion across a network. Multi-homing can be a good solution to avoid points of congestion in the network. We therefore propose Real Time & Multi-path Transmission Protocol (RTMTP), a transport layer protocol in which multiple path real-time transport and effective congestion control are available. Our protocol uses the explicit congestion notification (ECN) mechanism to detect network congestion level and to control transmission rate of VoIP traffic at the sender side. To maximize network bandwidth efficiency of multiple paths, the RTMTP shifts traffic among those paths. Our simulation results show that RTMTP improves VoIP data transmission throughput by effectively utilizing multi-path without voice call quality degradation, and helps users to use more inexpensive access network as much as possible while maintaining an acceptable voice quality.
Keywords: VoIP, Multi-homing, SCTP, RTP, ECN, E-model

1. Introduction

With improved terminal performance and demand of users for more sophisticated service, it has even become possible that one terminal supports several interfaces, that is, one terminal can connect to a number of access networks simultaneously. The demand and necessity for the multiple paths use of an end-terminal are increasing, as it can give benefits of advanced services and efficient management of network resources and traffic. Especially, using multiple paths can help to guarantee a certain level of quality for time-sensitive communications service such as VoIP.

VoIP quality is affected by latency, jitter, or packet loss. Especially, bursty packet loss has a severe impact on Voice over IP call quality. Because bursty packet loss tends to occur during network congestion, in order to guarantee voice quality, congestion should be under control. To avoid network congestion, sender needs to control sending rate. However, Real-Time Transport Protocol (RTP)/ RTP Control Protocol (RTCP)[1] which are widely used for VoIP can not support an effective method to control it.

For the purpose of providing a reliable end-to-end message transportation service, the Stream Control Transmission Protocol (SCTP)[2] and its various extended versions[3-5] were proposed. They support TCP-like congestion control and well-designed multi-homing solution. However, they are not suitable for real time application.

We therefore propose a new protocol named Real Time & Multi-path Transmission Protocol (RTMTP), a transport layer protocol that is capable of supporting multiple path real-time transport and effective congestion control. RTMTP data transfer is designed to be suitable for transporting data with real-time character. It does not support reliable transmission.

RTMTP supports multi-homing. Especially, it can utilize multiple paths simultaneously to transfer data. This enhances the throughput by bandwidth aggregation over multiple paths. RTMTP adopts congestion avoidance scheme. The congestion level of each path is monitored continuously. The level of network congestion is measured by ECN-marked packet probability. With the information, RTMTP shares traffic among paths for per path congestion control.

In order to maximize the use of the user preferred path, each path has a user preference value set up by a user.

RTMTP adopts the ITU-T E-model[6] as a measurement method to estimate voice call quality.

The rest of this paper is organized as follows. In section 2, the related works are introduced briefly. In section 3, we describe in detail the proposed Real Time & Multi-path Transmission Protocol. Section 4 presents the performance evaluation with a simulator. Finally, in section 5, we make some conclusions.

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2. Related Works

2.1 RTP/RTCP

The Real-Time Transport Protocol (RTP)[1] is an Internet protocol standard that 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 is commonly used in VoIP applications.

RTP runs on the top of the User Datagram Protocol (UDP). It does not in itself address resource reservation and does not guarantee quality of service (QoS) for real-time services. (since it is dependent on network characteristics) RTP is used in conjunction with the RTP Control Protocol (RTCP) [1] to monitor data delivery statistics and QoS.

RTP provides facility for jitter compensation and detection of out of sequence arrival in data. However, RTP/RTCP can not quickly respond to dynamically changing networks.

2.2 SCTP

The Stream Control Transmission Protocol (SCTP)[2] is a transport layer protocol that provides a connection oriented, full duplex, reliable data communication path and in-sequence transport of messages with congestion control (TCP like). It was defined by the IETF Signaling Transport (SIGTRAN) working group.

SCTP serves in a similar role to the Transmission Control Protocol (TCP). However, unlike TCP, SCTP provides message-based multi-streaming.

SCTP supports a good solution for multi-homing. During an SCTP association initialization, the two end-hosts exchange their multiple IP addresses. However, the SCTP multi-homing supports only communication reliability. Only one primary path is used for data unit transmission. Secondary paths are used to deal with all kinds of retransmissions or as a backup path. It is not used for load balancing.

In order to enhance the multi-homing ability of the original SCTP, extended versions of SCTP such as LS-SCTP (Lord-Sharing SCTP)[3], cmpSCTP (concurrent multi-path SCTP)[5] and CMT-SCTP (Concurrent Multi-path Transfer - SCTP)[4] were proposed. These protocols support effective load-balancing and multi-path transfer.

2.3 ECN

Explicit Congestion Notification (ECN)[7-9] is an extension to TCP/IP and supports the binary congestion information of the networks. ECN allows end-to-end notification of network congestion occurrence in the network without dropping packets.

TCP/IP networks originally recognize the network congestion by checking packets dropped by a router. Instead, an ECN-aware router may set a bit in the IP header as a mark indicating congestion without dropping a packet. The congestion indication is echoed by the receiver to the sender, which reduces its transmission rate as if it detected a dropped packet.

The purposes of the ECN are to avoid unnecessary packet losses and to support quick congestion control

3. Real Time & Multi-path Transmission Protocol

Our main goal is to design a new multi-path real-time transport protocol, and it has to satisfy the following features:

- Suitable for the delivery of time-sensitive data such as VoIP
- Support effective congestion control mechanism
- Maximize the effect of path diversity
- Considering user's path preference

In this section, we introduce RTMTP in which multi-path real-time transport and effective congestion control are available.

3.1 RTMTP Design

The RTMTP is a transport layer protocol, serving in a similar role to the cmpSCTP and RTP/RTCP.

RTMTP borrows a multi-homing concept from cmpSCTP and a data transmission feature from RTP, and adds an ECN based congestion control concept.

RTMTP doesn't support end-to-end flow control, but congestion control is performed per each path. Especially, congestion avoidance is achieved by traffic shifting mechanism. RTMTP also applies play-out buffer mechanism to minimize the unexpected effect due to transmitting traffic through multiple paths which have

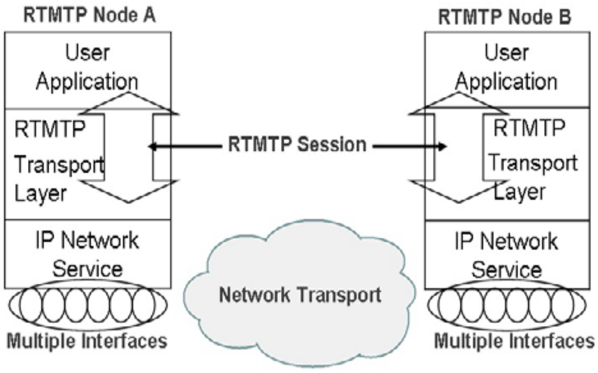


Fig. 1 RTMTP protocol stack.

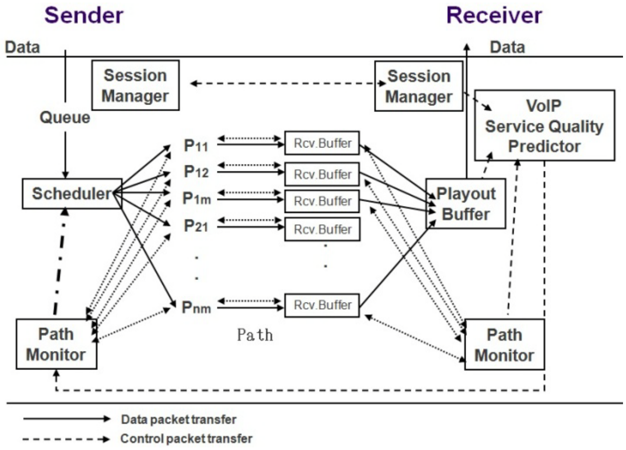


Fig. 2 RTMTP architecture.

different attributes.

Figure 1 shows the protocol stack of RTMTP.

RTMTP is between Application layer and Internet layer. Each RTMTP node can utilize more than one IP address in a RTMTP session and uses multiple paths to deliver VoIP data. The RTMTP connection between two RTMTP nodes is called a "RTMTP session" and RTMTP does not allow a half-open connection. A 4-way handshake is used for establishing a RTMTP session. (We borrowed this concept from SCTP)

3.2 User Preferred Path

RTMTP can maximize the use of user preferred interface. In case of using multiple-interfaces, a user may want to send data traffic through the cheapest interface. In order to reflect user's demands, each interface has a priority in RTMTP. User can set up the priority of each interface.

During session initialization, all the paths priorities are decided according to the interface priority.

RTMTP tries to allocate data traffic into more preferred path.

3.3 Architectural Overview

Figure 2 provides an architecture model overview of RTMTP.

The session manager is responsible for managing session information such as paths list. During initiation of the RTMTP session, two end-hosts exchange active interfaces lists consisting of their priority information and make out all available paths list. Session information can be dynamically negotiated between two end-hosts during communication.

After session initiation, VoIP traffic is transferred to the peer host through paths.

The scheduler is responsible for allocating RTMTP data packets to paths. It is a weighted round-robin scheduler. The scheduler assigns each packet, with a path allocation ratio Par_i, to one of the paths. Par_i denotes the packet allocation ratio of path i. The sum of all Par_i from $i = 1$ to N equals 1, where N is the number of paths. In the non-congested case, the scheduler tries to maximize the use of a more preferred path. When network congestion is predicted, the scheduler shifts some traffic from the more congested paths to less congested ones. This is achieved by adjusting Par_i values.

The Path monitor checks each path continuously. (See detailed explanation in section 3.5)

The receive buffers minimize the effect of out-of-order packet deliveries. Due to the different path characteristics such as RTT, data packets can arrive out-of-order. It intensifies packet loss, because RTMTP doesn't support session flow control. The role of the receiver buffer is to adjust delay difference among the paths. Each receive buffer calculate the delay difference from the RTT of the slowest path and the buffer holds the received packet during the difference time before it is forwarded to the upper layer. If the play-out buffer size is enough, the receive buffer is not necessary.

The play-out buffer is used at the receiver-end to compensate for variable network delays (jitter) and to maintain packet order.

The VoIP Service Quality Predictor measures current VoIP call quality periodically or calculates the expected call quality when congestion occurs. When congestion is predicted, if the expected call quality is worse than a certain level of quality, the scheduler starts re-allocation of data packets among paths. The VoIP Service Quality Predictor uses the E-model[6] to calculate the voice call quality. (See detailed explanation in section 3.6)

3.4 RTMTP Packet Format

RTMTP packet is uses two types of packet, data packet and control packet.

Figure 3 shows RTMTP packet structure.

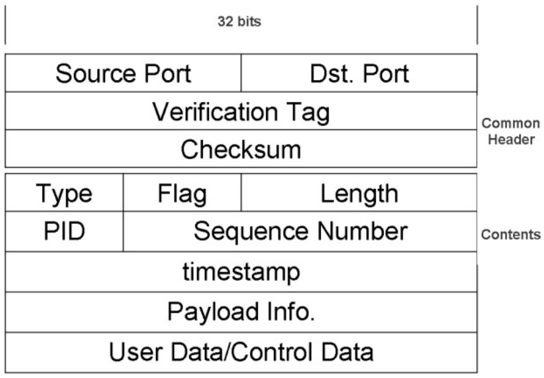


Fig. 3 RTMTP packet structure.

An RTMTP packet is composed of a common header and contents. Contents contain data or control information. RTMTP packet contains a common 12-byte header containing source/destination port numbers, the verification tag and checksum. The tag is a session identifier. Data content contains a user data. Path ID (PID) indicates transmission path. Each path in a session should have a unique ID. The sequence number increases by one for each RTMTP data packet sent, and is be used by the receiver to detect packet loss and to restore packet sequence. The timestamp reflects the sampling instant of the first octet in the RTMTP packet by which RTT and jitter can be estimated. The role of Payload information is same as RTP's payload type (PT). The following control packet types are defined in RTMTP

- Session Initiation / Session Initiation ACK :
- Heartbeat / Heartbeat ACK
- Session Update Request
  / Session Update Request ACK
- Path Status Request / Path Status Request ACK
- Path Status Inform / Path Status Inform ACK
- Path re-assignment Request
  / Path re-assignment Request ACK

3.5 Path Monitoring

Because RTMTP dynamically utilizes multi-paths to transfer VoIP traffic, path attributes should be monitored continuously. 'Path monitor' is a component to check path status.

The path monitor monitors network congestion degree, Round Trip Time (RTT) delay and delay jitter of every active path.

3.5.1 Congestion Monitoring

As network congestion causes packet loss, it is very important to detect congestion before it occurs. The path monitor uses ECN marking capability to predict network congestion.

Every packet arrived at receiver are monitored by the receiver-side path monitor. The path monitor collects information including total number of packets received and the number of ECN-marked packets received in a certain time interval. With the information, the path monitor calculates ECN-marked packet probability *Pepl.*

$$Pepl = \frac{\text{The number of ECN-marked packets}}{\text{Total number of received packets}} \quad (1)$$

Pepl is collected per each path.

By calculating ECN-marked packet probability, the path monitor can predict network congestion of the path and estimate expected packet loss probability.[10] By reducing the sending rate of the path based on the *Pepl* value, congestion can be avoided.

3.5.2 Path Attributes Monitoring

The path monitor checks path attributes such as RTT delay and delay jitter, and updates the active path list by sending Heartbeat control packets periodically

On receiving a Heartbeat packet, the receiver responds to the packet with a Heartbeat ACK packet. With the timestamp recorded in the Heartbeat ACK, RTT and jitter for the path can be estimated.

3.6 Assessing VoIP Call Quality

The ITU-T E-model[6)] is the most widely used subjective voice call quality measurement method. E-model takes into account various network parameter-based factors that affect the speech quality. The transmission rating factor(R-factor) range from 0 to 100 and is converted into a MOS rating to give the MOS score. [11)]

$$MOS = 1+0.035R + 7 \times 10-6R(R - 60)(100 - R) \quad (2)$$

Where the rating factor R is given by:

$$R = (R0 - Is) - Id - Ie\text{-}eff + A \quad (3)$$

Where R_0 represents the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise, I_s is a combination of all impairments related to voice recording, Id represents the impairments caused by delay of audio signal, $Ie\text{-}eff$ impairment factor presents all degradations caused by packet network transmission path, including end-to-end delay, packet loss and codec PLC(Packet Loss Concealment) masking capabilities and A is an advantage factor of particular technology (0 for wireline and 5 for wireless networks). Based on recommended values in ITU-T G.107[6)], Eq. 3 can be written as

$$R = 93.2 - Id - Ie\text{-}eff + A \quad (4)$$

$$Id = Idte + Idle + Idd \quad (5)$$

Id is an accumulation of impairments from talker echo ($Idte$), listener echo ($Idle$) and long delays (Idd). Reference 12) supports a practical and repeatable method of estimating VoIP network delay and Id is computed through it with the measured RTT information.

$$Ie\text{-}eff = Ie + (95 - Ie) \cdot \frac{Ppl}{Ppl + Bpl} \quad (6)$$

where the Ppl is packet-loss probability, and Ie is an equipment impairment factor(caused by PCM..etc,.) and Bpl is packet-loss robustness Factor. Ie and Bpl is defined as a codec-specific value in ITU-T Recommendation P.800[13)].

Ppl can be measured at the play-out buffer because RTMTP data packets consist of sequence number.

3.6.1 VoIP Quality Prediction

When congestion is predicted, the expected packet-loss probability can be estimated from ECN-marked packet probability.

$$Ppl = \sum_i^N (Par_i \times Pepl_i) \quad (7)$$

Where, Par_i is the path allocation ratio of path i, $Pepl_i$ is the ECN-marked packet probability and N is the number of all paths.

Consequently, using E-model and monitored network information, the actual voice speech quality or the expected voice speech quality can be estimated.

3. 7 Congestion Control

RTMTP doesn't support session level flow control, and it is difficult to control the overall sending rate. RTMTP supports traffic shifting method to avoid congestion.

When congestion is predicted, RTMTP calculates the expected voice call quality (Eq. 3). If the expected quality is worse than a certain quality (this quality base can be negotiated between end-users), RTMTP shifts some of its traffic from the path in which congestion is predicted to less preferred path. The sending rate of shifted traffic is decided by ECN-marked packet probability.

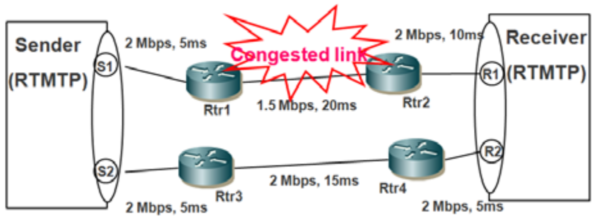
4. Performance Evaluations of RTMTP

We implemented RTMTP in NS-2.31[14)] to analyze the performance of the proposed architecture.

Figure 4 shows the network topology used for simulation. We implemented two network models with RTP and RTMTP respectively for a performance comparison. We tested our implementation under a variety of traffic mixes. To investigate the impact of network congestion, we generate several VoIP traffic (G.711, GSM-AMR) and UDP traffic and concentrate it into the congested link. Especially, to simulate realistic voice traffic, we adopted a talkspurt-silence model for VoIP traffic.

We assumed that voice data packets only are sent

(a) Network model with RTP



(b) Network model with RTMTP

Fig. 4 Network model for simulation.

from the sender to the receiver and smaller numbered interface has higher priority. The sender generates VoIP traffic using the G.711 codec which has a fixed bit rate of 64 Kbps. The payload size is 80 bytes.

It is assumed that the path (S1, R1) is preferred than the path (S2, R2) and data traffic tends to be assigned to the path (S1, R1).

Figure 5 shows the voice flow generated by the sender.

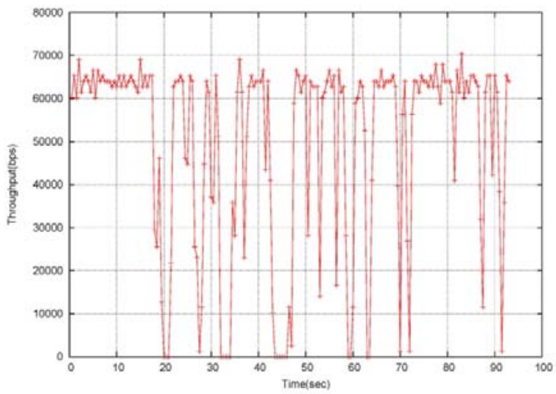


Fig. 5 G.711 VoIP flow generated by the sender.

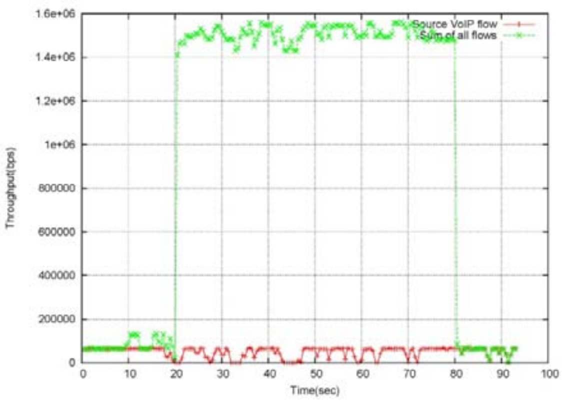


Fig. 6 Traffic flowed into the link between Rtr1 and Rtr2.

Figure 6 shows all the traffic flowed into the link between Rtr1 and Rtr2.

4. 1 RTP Performance Evaluation

In case of RTP, after about 20 seconds, Rtr1-Rtr2 link starts to drop packets. The red solid line if **Fig. 7** shows the dropped VoIP traffic during the congested period. The green dotted line shows the traffic arrived at the receiver without being dropped. Total packet loss ratio is depicted in **Fig.8**. Congested link causes packet loss and it affects VoIP call quality.

We calculated "R" value as the measure of voice quality. (**Fig.9**) During the congested period, voice quality is severely degraded.

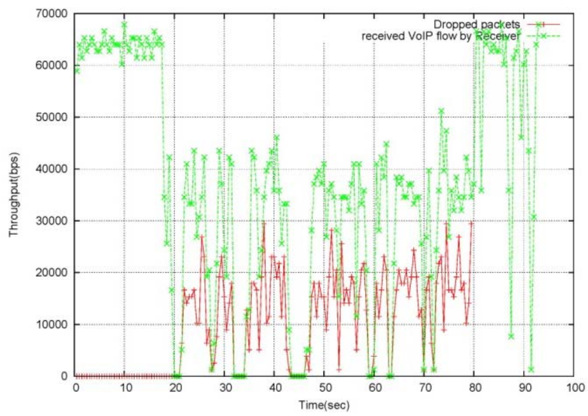


Fig. 7 Dropping VoIP traffic.

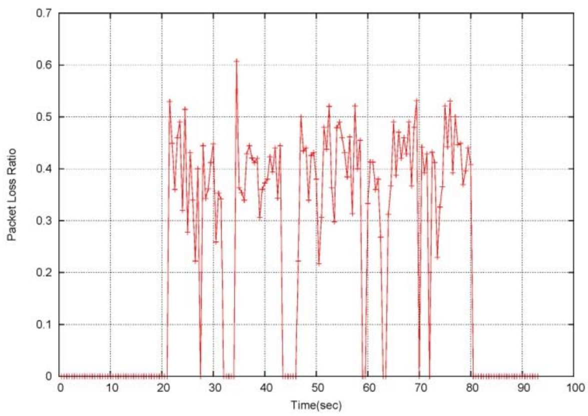


Fig. 8 Packet loss ratio.

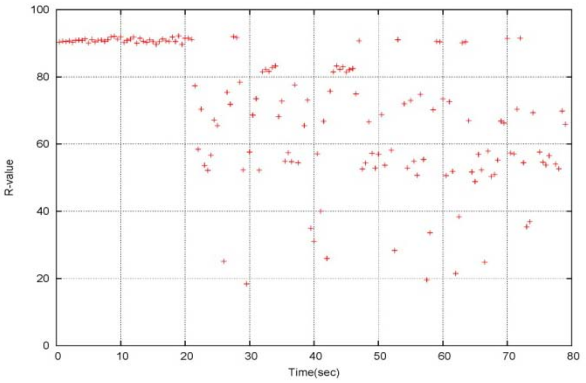


Fig. 9 R-value (RTP).

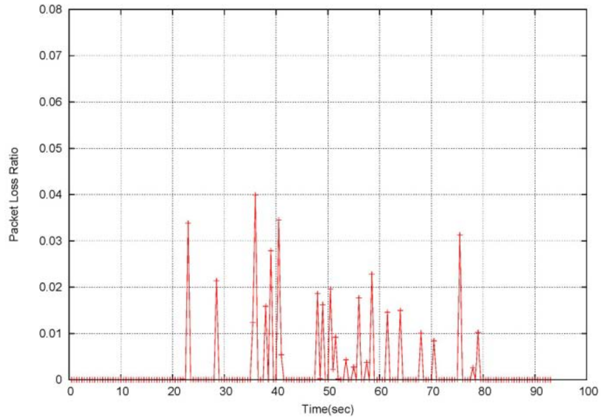


Fig. 12 Total packet loss ratio.

4. 2 RTMTP Performance Evaluation

Now, we show the RTMTP performance evaluation result. The traffic patterns used for the RTMTP system are same as the RTP's.

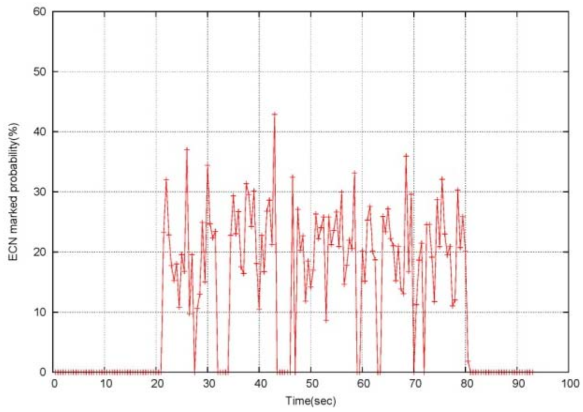


Fig. 10 ECN marked probability.

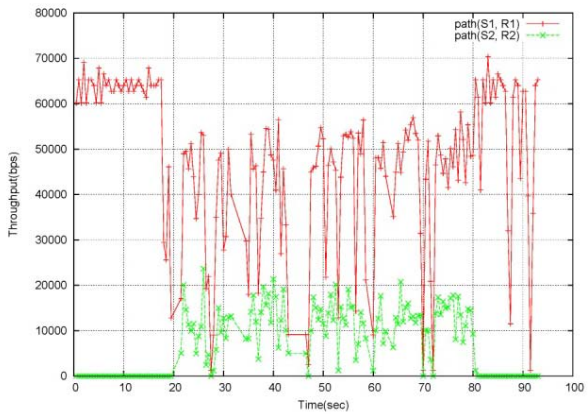


Fig. 11 Traffic shift between paths.

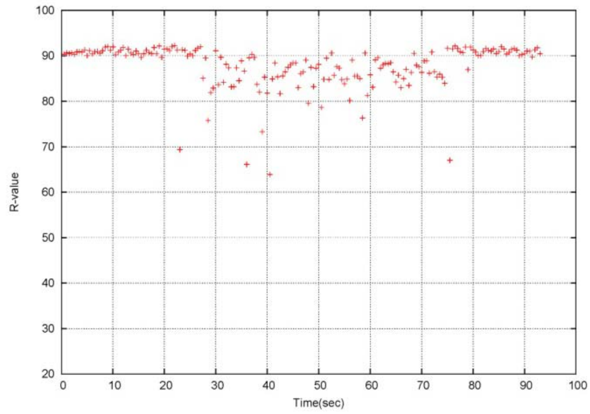


Fig. 13 R-value(RTMTP).

Figure 10 shows ECN marked probability calculated by receiver (R1). The RTMTP scheduler controls the transmission rate of each interface according to the ECN marked probability value. Traffic shifting is triggered by checking ECN marked probability.

In **Fig.11**, the red solid line and the green dotted line indicate the traffic flowed into the path (S1, R1) and the path (S2, R2) respectively.

Figure 12 shows the total packet loss ratio detected by the playout buffer of receiver. It is caused by delay jitter and packet reordering rather than network congestion.

In **Fig.13**, we also calculated voice call quality of RTMTP system. As compared to the RTP case, RTMTP assures reasonable voice call quality by avoiding network congestion and by perform traffic shifting.

5. Conclusions and Future Works

In this paper we suggested a new protocol named Real Time & Multi-path Transmission Protocol (RTMTP) that is capable of supporting multiple path real-time transport and effective congestion control.

We optimized the data transmission scheme of RTMTP for real time transmission. Furthermore, in order to guarantee service quality, RTMTP supports multi-homing and path diversity utilizing.

To minimize or eliminate the harmful effect of network congestion, a predictive method for congestion avoidance is proposed. With the ECN support, network congestion is predictable before actual congestion occurs. ECN-marked packet probability is a reasonable yardstick to decide network congestion degree. Especially, actual packet loss rate can be predicted. This value helps to support fine-grained congestion control.
To enhance the usage of user preferred paths, a simple priority policy is supported. RTMTP has two kinds monitoring scheme. For an effective management of network resources, the status of every path is monitored periodically. Voice speech quality is estimated periodically as well. E-model supports a reasonable method to calculate speech quality with network status information.

To analyze the performance of the proposed architecture, we implemented RTMTP in NS-2. The result showed that RTMTP is suitable for real-time service and achieves the good performance for data transmission.

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