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QoS-aware path switching for VoIP traffic using SCTP

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ABSTRACT

Voice over Internet protocol (VoIP) has been a prevalent multimedia service nowadays. It allows us to transmit voice data over IP networks. However, quality of service (QoS) is a major challenge to VoIP services. It must provide similar quality to traditional public switched telephone network or cellular phone services. Therefore, QoS related protocols have become important for real-time applications. Multi-protocol label switch (MPLS) is one of the important techniques to improve the network performance from QoS point of view. It employs label swapping to speed up packet forwarding. However, when a large number of users utilize VoIP services, the network congestion issue still exists. It causes delay, jitter and packet loss that affect VoIP QoS. In this paper, we propose a QoS-aware path switching strategy by using stream control transmission protocol (SCTP) in MPLS network to improve the VoIP traffic. This was done by employing SCTP selective acknowledgment mechanism to report the transmission parameters of primary path and to determine the criteria to switch to backup path. Simulation results show significant improvement in VoIP QoS.

1. Introduction

Voice over Internet protocol (VoIP) allows people to make telephone calls through any IP-based network. The voice signal is converted into digital packets and transmitted over the network. This type of multimedia communication service has increased users' demand for data transmission resources. The VoIP service should offer not only the reliability but also the quality of service (QoS) [1–4]. Offering high quality conversation services with limited resources is still a hot research topic.

It has been popular for Internet and network researchers to correlate the QoS issues with VoIP services. A lot of Internet standards on VoIP networking and QoS services can be found in [5–9]. In order to improve the VoIP QoS, the network must assure a certain level of connection. Unfortunately, IP networks are best-effort services that cannot guarantee certain QoS levels. Especially, the massive VoIP traffic entering the IP network makes the network performance worse.

Multi-Protocol Label Switch (MPLS) is a traffic engineering (TE) introduced by the Internet Engineering Task Force (IETF) [10] which combines the label swapping of the core network and the IP routing of the edge network. Based on its small and fixed length label, MPLS routers are able to identify and classify packets as well as to determine the next hop with less processing time. Either using hardware or software, the MPLS label can be implemented easily by arrays. MPLS with differentiated type of services (DiffServ) [11] provides better solution in solving the high-speed issues of router forwarding effectively [12]. This DiffServ/MPLS technology is believed to enhance QoS provisioning ability for conventional IP-based networks. Due to the capability of providing preferential treatments to users by supporting differentiated traffic classes, such as expedited forwarding (EF) [13], assured forwarding (AF) [14], and best effort (BE) classes, MPLS network is also believed to be a suitable platform for VoIP services. Because the conventional VoIP service providers support only one of three service classes, therefore the QoS-mapping for VoIP services in MPLS networks is an issue needed to be resolved.

On the other hand, when transmitted packets keep increasing, the network congestion still exists in MPLS network. The congested network will cause delay, jitter and packet loss for packet transmission, and consequently affect the QoS [15]. Furthermore, if the performance of the pre-defined label switching path (LSP) in MPLS network deteriorates or fails, it must be re-established [16] or re-routed [17]. This costs additional processing resources and increases the transmission delay. So, it is essential for the transport layer protocol to control and monitor network conditions. The traditional transport layer protocol, such as transmission control protocol (TCP) [18], is not suitable for conveying real time multimedia data. User datagram protocol (UDP) [19] has replaced TCP for VoIP services, however, when the network is congested, UDP must rely on the application layer protocol to deal with the packet loss or reliability problems. This results in reduction in transmission efficiency and VoIP QoS.

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The stream control transmission protocol (SCTP) is a potential transport layer protocol proposed by the IETF Signaling Transport (SIGTRAN) working group in October 2000 [20] and updated in September 2007 [21]. Originally, SCTP was designed to transmit public switched telephone network (PSTN) signals. SCTP adopts the idea of TCP flow control as well as congestion control schemes and utilizes the partial ordering and selective acknowledgment (SACK) mechanisms to satisfy the high efficient transmission with reliability. Therefore serious delay, jitter, and packet loss could be avoided by using SCTP as transport layer protocol [22]. Furthermore, the prominent feature of multi-homing, which allows two end-points to set up an association with multiple IP addresses on each host, is capable of supporting the switchovers of different IP addresses at one end-point without interrupting any ongoing data transfer [23]. It is believed that SCTP can efficiently convey transmission of real-time applications, such as VoIP, with high QoS [24] by using its multi-homing [25] feature and/or current multipath transfer (CMT) mechanism [22].

In this study, the multi-homing and heartbeat request/acknowledgment (HEARTBEAT) schemes of SCTP are used to solve the problem of path failure or performance decrease of primary path and then to provide the session continuity and path switching over MPLS network. The SACK and chunk bundling schemes are modified to report the VoIP QoS parameters. Therefore, we propose a QoS-aware path switching strategy for VoIP traffic by using SCTP in MPLS network.

The rest of this paper is organized as follows. Section 2 describes the research related to the QoS improvement for VoIP network. The detailed design of the proposed QoS-aware path switching scheme is illustrated in Section 3. Section 4 provides the performance analyses and evaluations as well as discussions based on the simulation results. We summarize this paper and address future work in Section 5.

2. Background

In VoIP we have to encode analog audio signal and encapsulate the digital data in packets to transmit over IP networks. For the Voice Codec proposed by the International Telecommunication Union (ITU) Recommendation, the 64 K bit rate codec is called Toll Quality. One of the codec standards with 64 K bit rate used in the PSTN network and VoIP is called the G.711 standard [26].

Generally speaking, the network performance, which can be evaluated from the measurable parameters such as delay, jitter, and packet loss, directly reflects the QoS of VoIP [27]. For example, the G.711 standard suggests the criteria of delay parameters [26]. As shown in Fig. 1, the acceptable level of VoIP QoS in delay parameter is 200 ms. The criteria of jitter and packet loss parameters of G.711 are shown in Figs. 2 and 3, respectively. To provide acceptable level of QoS, the jitter for VoIP transmission must not exceed 200 ms. The average packet loss on the other hand should be controlled within 5% to achieve an acceptable VoIP QoS. In this study, we utilized these three important parameters of VoIP QoS as the estimated criteria to design a QoS-aware path switching strategy for VoIP traffic using SCTP in MPLS network.

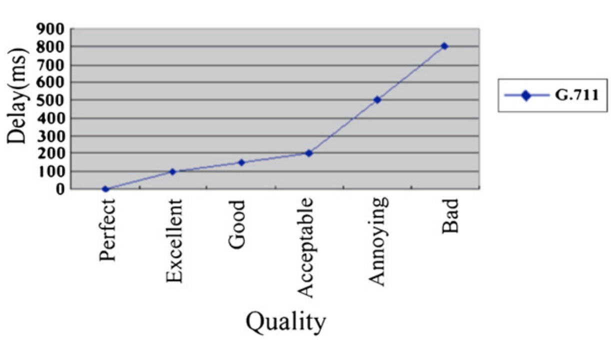

Fig. 1. The criteria of delay parameter in G.711.

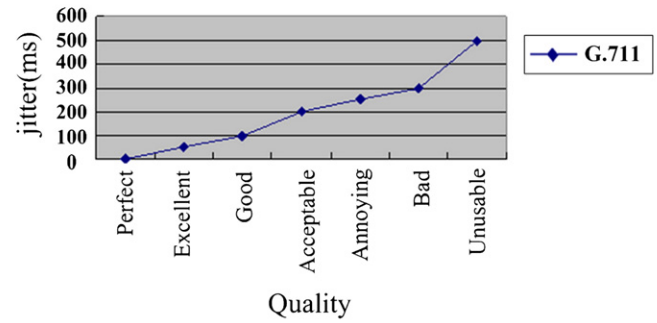

Fig. 2. The criteria of jitter parameter in G.711.

Some of the related researches on QoS using MPLS focused on multi-path issues. For instance, they integrated multi-path [25] with TE or applied concurrent multi-path transmission (CMT) at the transport layer [22] to enable load balancing [28,29], or to provide fault tolerance [30]. Some researches used SIP protocol to monitor the QoS [31,32]. The other groups [33,34] employed active measurement in multi-homed wireless network to achieve optimal path selection. They made the path switching dynamic in the transmitted procedure so that the effective usage of paths could be reached. The research in [35] proposed the pricing approach to provide different QoS levels in various networks, such as MPLS network. The study in [36] introduced a new MAC protocol with multi-beam directional antennas to provide a QoS-aware scheme which allows QoS guarantees for real-time multimedia services.

Researchers in previous study [37] presented an extension to the MPLS networks by using the predefined threshold to calculate the cost of the traffic distribution and the effectiveness of LSPs to minimize packet loss and to control jitter. In [38], authors conducted simulation to model the available bit rate (ABR), constant bit rate (CBR) and variable bit rate (VBR) services with QoS supports for multimedia applications. These papers applied the long-range dependency in traffic of ATM characteristics for real-time applications using MPLS, however, they did not provide any QoS-aware and dynamic path switching mechanisms of LSP in MPLS networks for real-time applications, especially focusing on VoIP services which are extremely sensitive to delay and jitter issues.

In a study of Fowler et al. [39], they proposed a scheme to transport data over the established MPLS LSPs. Their scheme used the probing packet to calculate the minimum delay time of the LSP path. According to the load of each LSP path, evaluated by delay time, they decided the optimal path to transmit packets. The minimum delay time of the path could stand for the shortest path in packet transmission, however, it might not represent the path loading condition and/or variation. Simply based on the statistics of delay time, the source host could not even detect whether the destination host has received the packets or not. The extra cost, due to the probing packets being sent out periodically, also needs to be taken into account when using their scheme. The other group [12] proposed two objective-oriented models for the VoIP services

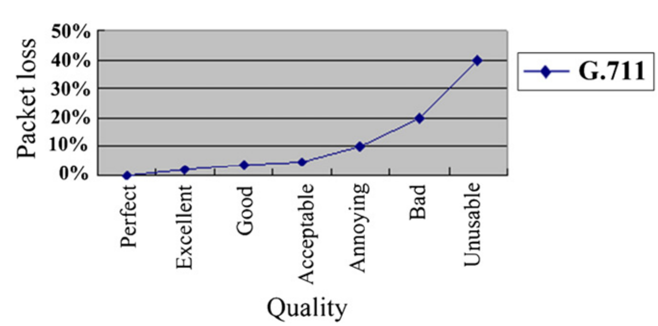

Fig. 3. The criteria of packet loss parameter in G.711.