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Abstract—VoIP application allows us to transmit the voice data over Internet. In this paper, a VoIP architecture with fault recovery and resource allocation by adopting Multi-homed Stream Control Transmission Protocol (SCTP) and Multi-Protocol Label Switching Traffic Engineering (MPLS-TE) mechanism was proposed. In the proposed architecture, SCTP was employed to transmit SIP messages while MPLS-TE mechanism was applied to set up the voice data transmitting path. With the multi-homing capability of SCTP, the voice call failure rate could be reduced and network resources could be utilized efficiently. Through MPLS-TE mechanism, traffic engineering functions such as network resources optimization, strict Quality of Service (QoS) voice data delivery, and fast recovery upon link or node failures could be ensured. We simulate three architectures in the Network Simulator (NS2) software and compare them under different network conditions. The simulation results reveal that the applicability of the proposed architecture.

I. INTRODUCTION

With the increasing of network bandwidth and the proliferation of networks, more and more network applications are developed. One of most promising applications is Voice over IP (VoIP). It allows us to transmit voice data over Internet. In VoIP architecture, a session consists of control messages exchange and voice data transmission. In the part of control messages exchanging, Session Initial Protocol (SIP) [1] is used for the voice call setup and the SIP messages are transmitted by User Datagram Protocol (UDP) [2] or Transmission Control Protocol [3]. Since UDP has no congestion control mechanism, the quality of voice call transmission could not be guaranteed and SIP application has to employ its own retransmission mechanism for reliability. This could lead to an unacceptable voice call setup time. On the other hand, using TCP as SIP signaling messages transporting protocol could incur the Head of the Line (HOL) blocking problem. The voice call setup time could be also unacceptable. In the part of transmitting voice data, most of VoIP applications use UDP to transmit voice data. However, UDP does not reserve resources and does not guarantee the quality of voice data transmission. Hence, how to provide a reliable VoIP architecture with the guarantee of quality of voice data is an urgent research topic.

Stream Control Transmission Protocol (SCTP) [4], developed by IETF Signaling Transport (SIGTRAN) Working Group, is a new signaling messages transport protocol that is featured multi-streaming and multi-homing. With the feature of multi-homing, when one transporting path fails, another interface can be used for signaling messages delivery without interruption. The multi-streaming feature separates and transmits messages on multiple SCTP streams. These streams are capable of independent and sequenced delivery. Due to the division of signaling messages into streams, single packet losses among messages of one stream do not block other streams. This could avoid the problem of HOL. SCTP uses a cookies-based 4-way handshake mechanism to ward off SYN flood attacks. Hence, using SCTP transporting SIP signaling messages seems to could archive a much better performance than TCP and UDP.

On the other hand, the support of an adequate QoS provisioning mechanism for VoIP application could satisfy the requirements of voice traffic QoS such as end-to-end delay, jitter, and packet loss [5]. Integrated services (IntServ) [6], differen-tiered services (DiffServ) [7], and Multi-Protocol Label Switch (MPLS) [8] are three QoS models that can be used to provide quality service for VoIP application. Among these protocols, MPLS gains a lot of attentions. MPLS, developed by IETF in 2001, is an advanced packet-forwarding technique. It uses labels to make forwarding decisions at the network node level, in contrast to the traditional destination-based hop-by-hop forwarding in IP networks. With MPLS, the processing overhead required for routing at the intermediate nodes could be reduced, thereby improving their packet forwarding performance. In addition, the MPLS-Traffic Engineering (MPLS-TE) approach [9] allows setting up explicitly routed Traffic Engineering-Label Switched Paths (TE-LSPs) whose paths satisfy a set of traffic engineering constraints, including bandwidth. Through MPLS-TE mechanism, Traffic Engineering functions such as network resources optimization, strict Quality of Service (QoS) delivery, and fast recovery upon link or node failures could be ensured.
In this paper, a VoIP architecture with fault recovery and resource allocation by adopting multi-homed SCTP and MPLS-TE mechanism was proposed. In the proposed architecture, SCTP was employed to transmit the SIP signaling messages. A multi-homed SIP Proxy server with two interfaces is developed to manage the transmission of SIP signaling messages. One interface connects to the primary path and the other connects the backup path. To increase the utilization of network resources, SIP INVITE messages are transmitted by primary path while SIP reply messages are transmitted by backup path. The retransmission of SIP INVITE signaling messages is also transmitted by backup path. This could lead to the better performances in the transmission of SIP signaling messages. Moreover, to reduce the voice call failure rate and archive the purpose of fault tolerance, SIP INVITE messages are transmitted by the backup path when the primary path is inactive. In the part of voice data transmission, due to UDP protocol has the smaller packet header and the advantage of fast transmission, we adopt the UDP protocol as voice data transporting protocol. To guarantee the quality of voice data transmission, MPLS-TE mechanism was applied to set up the transporting path of voice data. With this mechanism, the transporting paths of both SIP signaling messages and voice data are separated. The fast reroute mechanism supported by MPLS-TE could reduce the loss rate of voice data transmission upon network congestion or network link/node failures.

The rest of this paper is organized as follows. Section II describes the proposed architecture and system operations. System simulation and performance evaluation are discussed in Section III. Finally, Section IV we conclude the study.

II. SYSTEM ARCHITECTURE AND OPERATIONS

A. System Architecture

The proposed architecture includes SIP clients, multi-homed SIP proxy servers, and MPLS-enabled networks. SCTP is used to transport the SIP signaling messages while UDP is applied to transport the voice data. SIP clients play the dual role (caller and callee). It could send and receive the SIP signaling messages. Communication between SIP clients is carried out by MPLS-enabled network with TE mechanism support. Each multi-homed SIP proxy server owns two network interfaces. One connects to the primary path and the other connects the backup path. The proposed architecture and the structure of multi-homed SIP proxy server are shown in Fig.1 and Fig.2, respectively.

B. Multi-homed SIP Proxy Server Operations

The operations of multi-homed SIP proxy server are illustrated in Fig.3. Both SIP proxy1 and SIP proxy2 own two network interfaces, one connects the network X.X.X/24 and the other connects the network Y.Y.Y/24. Once SIP proxy1 communicates with SIP proxy2, a SCTP connection, called an association, is established through a four-way handshake mechanism as opposed to the three-way handshake implemented by TCP. After receiving the SIP INVITE message from caller, SIP proxy1 forwards this message from INT1 with IP address X.X.X.1 to SIP proxy2. SIP proxy2, after receiving the SIP INVITE message from INT1 with IP address X.X.X.2, forwards it to the callee. Callee sends 200 OK message to SIP proxy2 if it accepts the invitation. SIP proxy2 transmits this 200 OK message to SIP proxy1 through the INT2 with IP address Y.Y.Y.2. This 200 OK message will be received by SIP proxy1 from INT2 with IP address Y.Y.Y.1 and be forwarded to the caller.

Once an association is successfully established between SIP proxy1 and SIP proxy2, HEARTBEAT and HEARTBEAT-ACK messages are sent periodically to monitor the state of association. When the primary path was detected to be inactive, the transmission of SIP INVITE message will be redirected from the primary path to backup path. This association does not need to reestablish and the transmission of SIP signaling messages would not be interrupted. In addition, the retransmitting SIP INVITE messages are transmitted by backup path until the primary path is active to avoid the retransmitting SIP INVITE messages losing again. The operations of multi-
homed SIP proxy server as primary path inactive are shown in Fig. 4.

C. Operations of MPLS-TE Path Setup and Fast Reroute Path Setup

As shown in Fig.1, the best path is LSR1-LSR2-LSR3-LSR4 and the MPLS-TE fast reroute mechanism is initiated. In general case, voice data are transmitted through this path from user1 to user2. In the proposed architecture, an explicit routing path (LSR1-LSR5-LSR6-LSR7-LSR4) is created using PATH signal of RSVP-TE protocol [10]. After the MPLS-TE path is set up, the voice data will be transmitted along with MPLS-TE path. Although this path is not the best path, the quality of voice data transmission could be guaranteed by MPLS-TE mechanism. On the other hand, once LSR5 detected the link between LSR5 and LSR6 is inactive, LSR5 uses RSVP-TE protocol to switch the routing path from LSR5-LSR6 to LSR5-LSR8-LSR6. The voice data will be directed to this path. With separating the transmission of voice call and voice data, the network resources utilization could be increased. The operations of MPLS-TE path setup and MPLS-TE fast reroute path setup are depicted in Fig.5 and 6, respectively.

III. SYSTEM SIMULATION AND PERFORMANCE EVALUATION

To verify the applicability of the proposed architecture, conventional, multi-homed, multi-homed with MPLS support network topologies, shown in Fig.7, 8, and 9 respectively, are simulated using the Network Simulator (NS2) software [11]. In conventional topology, both the SIP messages and voice data are transmitted by UDP. In multi-homed network topology, SCTP is applied to transmit the SIP messages and UDP is used to transmit the voice data. The proposed architecture is simulated in the multi-homed network with MPLS support topology. For the sake of convenience, we named these three architectures as UDP-SIP, mSCTP-SIP, and mSCTP-SIP-MPLS respectively. We comparatively evaluate the performance of three architectures in terms of voice call setup time and voice call failure rate under different conditions. We also investigate the end-to-end delay of voice data transmission under different conditions in both UDP-SIP and mSCTP-SIP-MPLS architectures.

A. Voice Call Failure Rate

In the simulation, we set that one SIP voice call per second is generated and the packet size of each SIP signaling message
is 577 bytes. After the voice call is setup successfully, voice data is generated using constant bit rate (CBR) mechanism. The packet size of voice data is 160 bytes. Due to there is no congestion control mechanism in UDP-SIP architectures, the retransmission mechanism of SIP application is enabled. To avoid retransmitting the same lost packets, SCTP recovery mechanism is enabled while SIP application retransmission mechanism is disabled in both mSCTP-SIP and mSCTP-SIP-MPLS architectures. To make SIP proxy server having enough time to handle the signaling messages loss, we set that the simulation time is 150 seconds. We also set that each SIP proxy server could handle at most 100 voice calls at the same time for calculating conveniently. Voice call failure rate, \( \text{SIP}_{\text{call-blocking}} \), is calculated by the following equation:

\[
\text{SIP}_{\text{call-blocking}} = \frac{(\text{InviteCalls} - 200\text{OKreply})}{\text{InviteCalls}}
\]

where \( \text{InviteCalls} \) means the number of SIP invite messages sent by SIP proxy server and \( 200\text{OKreply} \) means the number of SIP reply messages received by SIP proxy server.

To simulate the congested network, we generate 100 voice calls in each architecture. Under such simulation environment, we obtain the SIP call failure rate is 0.18, 0, and 0 in UDP-SIP, mSCTP-SIP, and mSCTP-SIP-MPLS architecture respectively. On the other hand, to simulate the network link failure condition, we set that the path between R2 and R3 will be failed from 20 seconds to 40 seconds after the simulation beginning in both UDP-SIP and mSCTP-SIP architecture. In mSCTP-SIP-MPLS architecture, the path between LSR2 and LSR3 will be failed from 20 seconds to 40 seconds after the simulation beginning. Under this simulation environment, we obtain the SIP call failure rate is 0.21, 0, and 0 in UDP-SIP, mSCTP-SIP, and mSCTP-SIP-MPLS architecture respectively. We can see that that voice call failure rate in UDP-SIP architecture is greater than other architectures under different network conditions. The is because that there is no the multi-homing feature and appropriate recovery mechanism in UDP-SIP architecture.

B. Voice Call Setup Time

We also investigate the voice call setup time of three architectures under network congestion and network link failure. The simulation environment and parameters are the same as previous simulation. The simulation results are shown in Fig.10 and 11, respectively.

![Fig. 10. Voice Call Setup Time as Network Congestion](image)

![Fig. 11. Voice Call Setup Time as Network Link Failure](image)

The better SIP call setup time is smaller than 1 second and it is unacceptable by user that SIP call setup time is greater than 5 seconds [12]. From Fig.10 and 11, we can see that the SIP call setup time in mSCTP-SIP architecture is smaller than in UDP-SIP architecture and is greater than in mSCTP-SIP-MPLS architecture. The setup time of most SIP calls in UDP-SIP architecture are close or greater than 5 seconds. This is because that both SIP signaling messages and voice data are transmitted on the same path and the retransmission mechanism is handled by SIP application. With the feature of multi-homing, both mSCTP-SIP and mSCTP-SIP-MPLS architectures could archive the purpose of fault tolerance and obtain the smaller SIP call setup time. Besides, mSCTP-SIP-MPLS architecture uses MPLS-TE mechanism to separate the transmission of SIP signaling messages and voice data. Consequently, the SIP call setup time in mSCTP-SIP-MPLS architecture is smaller than in mSCTP-SIP architecture.

C. End-to-End Delay of Voice Data Transmission

If the end-to-end delay of voice data transmission is smaller than 150ms, the quality of voice is excellent. If the end-to-end
delay is between 150ms and 250ms, part of user could feel to be influenced during communication [13]. In this simulation, we evaluate the end-to-end delay of voice data transmission upon network congestion and network link failure conditions in both UDP-SIP and mSCTP-SIP-MPLS architectures. In mSCTP-SIP-MPLS architecture, we set that the transmitting path of SIP INVITE messages is LSR1-LSR2-LSR3-LSR4 and the reservation bandwidth of MPLS-TE path to 500K bytes. After 2 seconds from simulation beginning, a MPLS-TE path (LSR1-LSR5-LSR6-LSR7-LSR4) is setup to transmit voice data. To avoid too many voice data traffics are generated and exceed the reservation bandwidth of MPLS-TE path, we set that SIP proxy server could accept at most 11 voice calls with 64k bits rate. In UDP-SIP architecture, the transmission of SIP signaling messages and voice data is along with the R1-R2-R3-R4 path. The simulation time is 49 seconds.

To simulate the congested network, the disturbing traffics, which the packet size is 1000 bytes and the transmission rate is 1M bits per second, are generated after 10 seconds from the simulation beginning. This disturbing traffics are generated between the path R2 and R3 in UDP-SIP architecture and are generated between the path LSR6 and LSR7 in mSCTP-SIP-MPLS architecture. We generated 11 voice calls in each architecture. In this simulation, the average end-to-end delay is 101 milliseconds in mSCTP-MPLS architecture and the average end-to-end delay is 168 milliseconds in UDP-SIP architecture. The end-to-end delay of voice data transmission of single voice call as network congestion is shown in Fig.12.

![Fig. 12. The End-to-End Delay of Data Transmission of Single Voice Call as Network Congestion](image)

On the other hand, to simulate the network link failure, we assume that after 20 seconds from the simulation beginning, the path between R2 and R3 will be failure and recover at 30 seconds in the UDP-SIP architecture and the path between LSR6 and LSR7 will be failure and recover at 30 seconds in the mSCTP-SIP-MPLS architecture. In the mSCTP-SIP-MPLS architecture, the voice traffic will be directed to the path LSR5-LSR8-LSR7 using MPLS-TE mechanism. The same as previous simulation, we generated 11 voice calls. In this simulation, the average end-to-end delay is 54 milliseconds in mSCTP-MPLS architecture and the average end-to-end delay is 68 milliseconds in UDP-SIP architecture. The end-to-end delay of voice data transmission of single voice call as network link failure is depicted in Fig.13.

![Fig. 13. The End-to-End Delay of Data Transmission of Single Voice Call as Network Link Failure](image)

**IV. CONCLUSION**

In this paper, we have proposed a VoIP architecture with fault recovery and resource allocation by adopting SCTP and MPLS-TE mechanism. By separating the transmission of voice call and voice data, the utilization of network resources could be increased and voice call failure rate could be reduced. Through efficient path selection and fast recovery mechanism supported by MPLS-TE mechanism, the quality of voice data transmission could be guaranteed. Simulation has been conducted to verify the applicability, and the simulation results illustrate the proposed architecture is suitable for VoIP application.

**REFERENCES**