A Framework for Traffic Engineering In a SIP-over-MPLS based network

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Abstract: The next generation communication systems will provide multimedia services and will need to support high quality of service in a more flexible and intelligent manner than previous generations. In this paper, a network based on SIP (Session Initiation Protocol) over MPLS (Multi-Protocol Label Switching) is proposed. A network architecture that adopts SIP protocol as signaling to set up a call and MPLS as a core network forwarding mechanism is identified. To combine of SIP and MPLS give this network two major features: it overcomes the weakness of traditional SS7 running in the Internet and provides high quality of service to the user using Traffic Engineering technology. A Traffic Engineering (TE)-SIP server is proposed, as a part of a network architecture that will use a SIP session to provide a TE request for a SIP client. Following this, the TE-SIP signaling mechanism, which includes the TE-SIP message flow and protocol, is discussed.

1 Introduction

Internet telephony is one of the most exciting areas in electronics today. Organizations of all sorts—Internet Service Provider (ISP), telephony companies and ordinary commercial enterprises---stand to benefit from this technology, which provides telephone or multimedia services over data networks based on the Internet Protocol (IP).

There are two problems with using an IP network to carry voice or multimedia traffic [1]. The first is the protocol which can be used to set up a call, the traditional telephone network PSTN(Public Switched Telephone Network) uses SS7 (Signaling System 7) for setting up a call, SS7 can be run over an IP network, but brings with it the assumptions of the centralized PSTN. However, if SS7 were used in an IP based network, we couldn't take advantage of the inherent flexibility of the distributed IP network. The second problem is how to guarantee the voice or multimedia quality we expect from the telephone system, for example, to remove the effects of variable delays.

To overcome these problems, the IETF (Internet Engineering Task Force) has proposed some protocols, one of them is SIP [2]: an emerging protocol, designed to provide basic call control and application signaling for voice and multimedia calls, or session management in a packet-switched network.

MPLS [3] is an advanced forwarding scheme. It extends routing with respect to packet forwarding and path controlling. MPLS was primarily developed for IP networks, one principal use of MPLS is to implement virtual connections, referred to as Label Switched Paths (LSPs). Combining SIP and MPLS together is a new idea to improve the Quality of Service in an IP network. In this paper, we propose a network architecture for SIP supporting an MPLS network to overcome the problems of the telephone system over IP network. Initially, brief overviews of the SIP and MPLS are presented. The proposed architecture is discussed in Section 4, Section 5 describes the TE-SIP server and the TE-SIP signaling mechanismis presented in Section 6. Some conclusions are given in Section 7.

2 SIP and MPLS

SIP is an important and evolving protocol for providing basic call control and application signalling for voice and multimedia calls, or sessions between users. SIP can be used to control Internet multimedia conferences, Internet telephone calls and multimedia distribution, and is likely to be used in both the core and the periphery of the future communications network. SIP is a set of standards that define how devices (computers, telephones, and mobile phones) exchange information with each other, it is also a signaling protocol that can be used to set up and manage any of these types of session, regardless of the media type (phone call, game, living video or instant message)[5]. It is modeled closely on HTTP. Like HTTP, it was designed to work over IP networks. One of the chief strengths of SIP is its ability to interact with other communications protocols, and to tie together multiple features into more advanced services. Members in a SIP session can communicate using multicast or unicast relations, or a combination of these. In addition, SIP is independent of the lower-layer transport protocol, which allows it to take advantage of new transport protocols.

Multi-Protocol Label Switching (MPLS)[3] is a new technology that will be used by many futures cores networks, including converged data and voice networks. MPLS does not replace IP routing, but will work alongside existing and future routing technologies to provide very high-speed data forwarding between Label-Switching Routers (LSRs) together with reservation of bandwidth for traffic flows with differing Quality of Service (QoS) requirements.

MPLS sets up explicit paths through the network [6]. The path is defined by the sequence of IP addresses of the nodes to be traversed. The paths that the data follows are referred to as a label-switched path (LSP). MPLS uses some signalling protocols like RSVP (Resource Reservation Protocol) or LDP (Label Distribution Protocol) to set up LSPs.

Many of today's discussions regarding MPLS revolve around Traffic Engineering (TE). Traffic Engineering involves making the best use of resources across an entire network by distributing traffic over different paths. That is not possible

with raw IP, but it is possible with MPLS LSPs. Traffic engineering offers VoIP (Voice over IP) providers the chance of utilizing network resources more fully and increases the ability to provide the required QoS in busy network.
By using a combination of SIP and MPLS for access network and core network respectively, we propose a method to provide the QoS guarantees required to transport voice or other multimedia traffic.

3 SIP over MPLS network Architecture

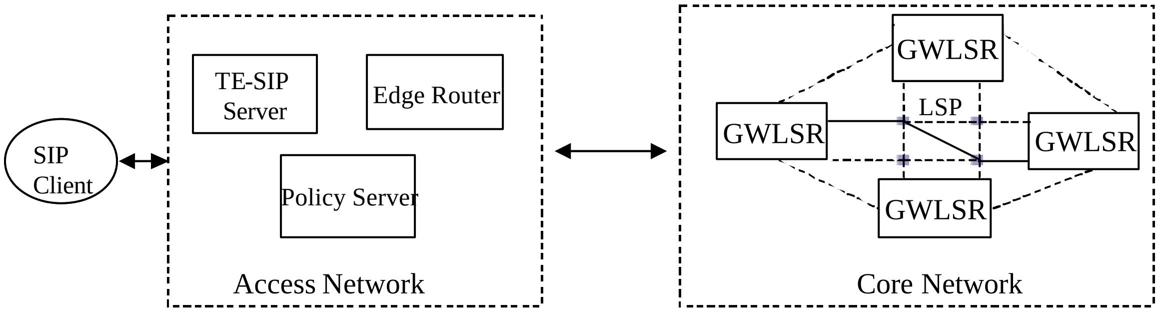

Figure 1 Reference Architecture for SIPoMPLS

Figure1 shows the Reference Architecture for SIP over MPLS (SIPoMPLS). This Architecture is divided into three parts: SIP Client or Terminal Equipment, Access Network and Core Network.
SIP Client is the terminal of the call, SIP client can start a SIP call setup session.
The Access Network is an Internet Service Provider (ISP),which can provide SIP service. When SIP client starts a SIP call-setup session, the Access Network will provide call control functions and signalling gateway functions.
The Core Network is a MPLS based network. There are two node types: Gateway devices and Label Switch Routers (LSR). The Gateway LSR (GW LSR) contains the functionality of a Label Edge Router (LER) as well as many other functions. The Gateway device interfaces the MPLS network with:

- Other media (e.g., IP, ATM etc.)
- Another MPLS network
- Other network (e.g., PSTN, VoIP etc.)
- With Access device.

The core MPLS network must be capable of supporting many different LSPs to convey voice or multimedia payload in an MPLS environment.

4 TE-SIP Server

The basic idea of designing a TE-SIP server is: the client sends a SIP message to its proxy server and receives the reply from its server. The SIP servers are therefore involved in the message exchange between the client and can add /read Traffic Engineering related information in the SIP messages. The enhanced SIP server is TE-SIP server (Traffic Engineering enabled SIP Server), it is in the Access Network. The originating TE-SIP server adds TE information in the SIP messages. This is meant as a hint that the originating side is capable of TE and is willing to exploit it. If the terminating SIP server is able to handle TE in a compatible way and it is willing to exploit it, it will answer positively with proper information in the response SIP messages. A legacy SIP server on the terminating side will not understand the TE information in the SIP message and will ignore it. Obviously, the SIP session would then be setup with no TE. So the setup of a TE session is logically composed of two aspects:

1. The end-to-end signalling mechanism to exchange TE information.
2. The TE negotiation between the SIP agents and MPLS network.

The SIP client is unaware of TE aspects and the local SIP servers do all the TE job, thus relieving the terminals from unneeded complexity.

5 TE-SIP Signaling Mechanism

Figure 2 shows the SIPoMPLS function model. We will describe the detail of the signaling mechanisms of the proposed SIP based MPLS network. When a call set up is initiated, the caller SIP client starts a SIP call setup session through the SIP proxy server [4]. If a TE-SIP server is encountered, this will start a TE session interacting with TE-SIP server. The TE-SIP server will communicate with another TE-SIP server, which belongs to the destination URL Access Network. At the same

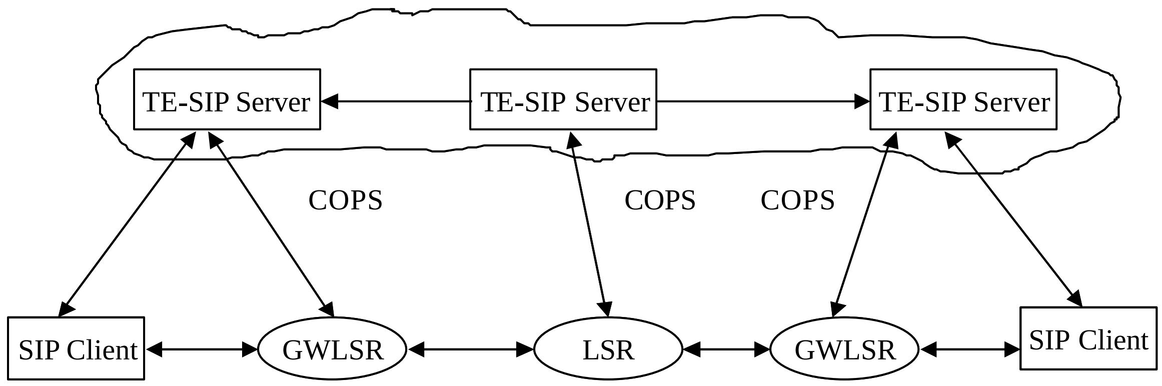

Figure 2 SIPoMPLS function model

time, TE-SIP server will request the TE with GW-LSR via the COPS (Common Open Police Service) protocol. TE requests are handled in the GW-LSR of the core MPLS network. The GW-LSR should implement all the mechanisms needed to perform admission control decisions and policy function.

5.1 TE-SIP message flow

Figure 3 shows the Signalling flow for the SIPoMPLS calls setup procedure. The call setup starts with a standard SIP INVITE message sent by the caller to the local TE-SIP server. The message carries the callee URL in the SIP header and the session specification within the body SDP. The caller sees the TE-SIP server as a standard SIP proxy server. The TE-SIP server decides whether a TE session has to be started or not according to the caller ID and session information. If a TE-server session has been setup, it inserts the required descriptors within the INVITE message and forwards it towards the invite callee. When the callee responses with a 200 OK message, it is passed back to the last TE-SIP server that is the TE-SIP server that controls the callee access network. The TE-SIP server on the callee side has all the information to request a specific TE routing (LSP). The TE-SIP server communicates with GW-LSR for the callee-to-caller traffic flow. When the caller TE-SIP server receives the response for the LSP request, if it is positive, it stores such SLP information and send it within the 200 OK message toward the caller. If the response for the LSP is negative, the TE-SIP server does not store the LSP information, but it still inserts in the 200OK response for normal SIP proxy server in order to deal with normal SIP call setup.
When the caller TE-SIP server receives the 200 OK message with the complete LSP indicators, it completes the LSP setup. If the response for this flow is negative, the caller-to-caller flow will deal with a normal call. When a call is terminated all resource that has been reserved must be released. This action is triggered by the BYE messages. The architecture keeps compatibility with standard SIP client and standard SIP servers.

5.2 TE-SIP Protocol

When the first TE-SIP server (i.e. the caller TE-SIP server) is encountered, it inserts a new field in the SIP header that is:
CallerER: <caller ER address>
The other TE-SIP servers know the IP address of the caller ER; moreover, the caller TE-SIP server add its VIA field (as every SIP proxy), in which it includes some specific information (considered as SIP "comments") that will be not visible to any other SIP or TE-SIP server, since they are within its own VIA field. This information will be used by the same TE-SIP server while processing the 200 OK responses. The VIA field is structured as follows:
VIA: SIP/2.0/UDP <SIP server address>[: <port>]
(FirstTESIP/CallerER: <caller ER address>[/<next comment>])
According to the standard SIP protocol processing rules each SIP proxy that manages the INVITE message adds a new VIA field; while all the other field, such as the CallerER field should be forwarded. Each TE-SIP server that manages an INVITE message containing the CallerER field, will also copy the caller ER address within its VIA field, as follow:
VIA: SIP/2.0/UDP <SIP server address>[:<port>] (CallerER:<caller ER address>[/<next commet>])
When the INVITE message reaches the callee host, the user client processes the call, and generates the 200 OK response (if the call is accepted). If the client is not aware of TE-SIP it simply discards each TE-SIP field (i.e. the CallerER) when forming the new response message.
When the 200 OK reaches the callee TE-SIP server, the corresponding VIA field is read, the TE-SIP session information is extracted (including the caller ER address) and TE requests for the IP flow in the callee-to-caller will communicate with the GW-LSR via COPS. When this LSP request/response phase is concluded, the TE State is stored and the 200 OK message is

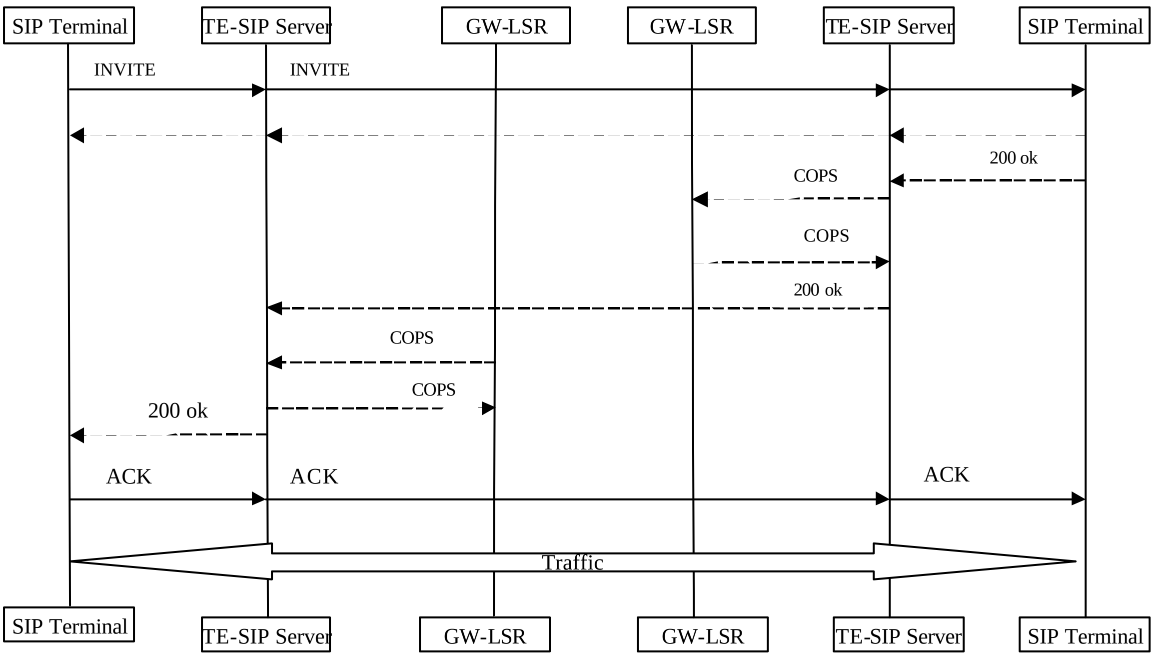

Figure 3 Signalling flow for the SIPoMPLS call setup procedure

relayed toward the caller
Within this new response message, the corresponding VIA field is dropped and a new field specifying the callee ER address is inserted, that is
CalleeER: <callee ER address>
If there are additional SIP servers handling this response in the path between the callee TE-SIP and caller TE-SIP servers, they will only drop their own VIA field according to standard SIP rules. The TE-SIP server recognizes that they are not the callee TE-SIP servers because the CalleeER field is already present in the message. When the first TE-SIP server is encountered (i.e. the caller TE-SIP server), it recognizes the field FirstTESIP within its VIA field and extracts the TE session information (including the callee ER address). Then it communicates with GW-LSR via COPS and starts the TE request for the IP flow in the caller-to-callee direction and stores the "TE State" for this flow.
Definition of VIA field is very important, because the use of a defined VIA field lets each TE-SIP server extract all information needed for the TE request directly from the SIP message that it is processing.
This mechanism allows the TE-SIP not to keep per session information until a TE request call is completely installed and can be used in MPLS network.

6 Conclusion

In this paper, we propose a network architecture to provide Traffic Engineering in a SIP over MPLS based Network. This model adopts the SIP protocol as signaling to set up a call and MPLS as a core network forwarding mechanism. The advantages for the network are that it overcomes the weakness of traditional SS7 running in the Internet and provides high quality of service to the user using Traffic Engineering technology. We propose a TE-SIP server to set up a TE request call, which uses a SIP session to provide TE request for SIP client. A TE-SIP signaling mechanism, which includes the TE-SIP message flow and protocol, has also been discussed.

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