Network Working Group
Internet Draft

H. Schulzrinne
Columbia University
L. Slutsman
AT&T
I. Faynberg
H. Lu
Lucent Technologies

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Interworking between SIP and INAP

Status of this Memo

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This document outlines a proposal to the IETF for a new work item in support of service delivery in converging networks.
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Abstract

The goal of this document is to identify a new IETF work item. The document defines the term "soft switch" as a mechanism by which PSTN Intelligent Network (IN) service control can be accessed by VoIP gateways and associated SIP servers. Specifically, the work item is on the mechanism for interworking of the Session Initiation Protocol (SIP) and Intelligent Network Application Part Protocol (INAP).

1. Introduction

This document first defines the term "soft switch" for the purposes of describing a mechanism by which existing PSTN Intelligent Network (IN) service control can be re-used in IP networks. IP telephony is one application that can benefit from this mechanism, but there are other potential applications of interest (for example, Unified

Messaging), which this draft does not address. Specifically, the document demonstrates the role of the soft switch in IP telephony. The

document then narrows, for practical purposes, the scope of the soft switch so as to identify an architecture and mechanism for interworking of Session Initiation Protocol (SIP) and Intelligent Network Application Part Protocol (INAP). Some (but not all) parts of this work have already been discussed in the IETF. To help delineating the proposed work item, this Internet draft explains how it relates to the efforts of the existing IETF working groups that deal with the protocols for PSTN/Internet interworking (IPTEL, MEGACO, PINT, SPIRITS, and SIGTRAN) and ITU-T.

The remaining sections of this document present, in this order, the problem statement, proposed area of standardization and service examples, relation of the proposal to the existing work in the IETF and other standards bodies, security considerations, acknowledgments, conclusion, references, and an appendix with figures.

2. Problem Statement

We address the porting of services that exist today in the PSTN to IP telephony gateways, provided by the Intelligent Network (IN). Such services include "freephone" (known as "800 number" in the US), Local Number Portability (LNP), and Virtual Private Network (VPN).

We first explain the basics of IN. Figure 1 shows a simplified IN architecture, in which telephone switches, called Service Switching Points (SSPs), are connected via a packet network called Signaling System No. 7 (SS7) to Service Control Points (SCPs), which are general purpose computers. At certain points in a call, a switch can interrupt a call and request instructions from an SCP. The points where a call can be interrupted are standardized within the Basic Call State Model (BCSM) [1]. The BCSM models contains two processes, one each for the originating and terminating part of a call. When the SCP gets an request for instructions, it can reply with a single response, such a simple number translation augmented by criteria like time of day or day of week, or, in turn, get into a complex dialog with the switch. The situation is further complicated by the necessity to engage other specialized devices, which collect digits, play recorded announcement, perform text-to-speech or speech-to-text conversion, etc. (These devices are not discussed here.) The related protocol as well as the BCSM is standardized by the ITU-T and known as the Intelligent Network Application Part Protocol (INAP). Only the protocol, but not an SCP API, have been standardized.

A soft switch is a process that behaves like a PSTN switch to any PSTN entity, such as SCP or SSP, and speaks IP signaling protocols. A soft switch can "talk" SS7 to SCPs and PSTN switches (thus acting as an SSP), and at the same time be a Media Gateway Controller to a Media Gateway, "speak" H.323 to a Gatekeeper and act as a SIP UA.

Here, as depicted in Figure 2, we are only concerned about the interaction of SIP and INAP, not the remaining aspects of a soft switch. In addition to the soft switch scenario, we can also have

a SIP call-stateful proxy server, presumably associated with a soft switch, interact with an SCP. In that configuration, the SCP effectively acts as a SIP location server (as shown in Figure 3).

We do not intend to consider the issue of how the soft switch "finds" the right SCP or how it authenticates itself to the SCP. The transport of INAP messages is outside the scope of this work as well.

SIN can be thought as something similar to the SIP Common Gateway Interface (CGI) [3, 4], specialized for INAP interworking. Unlike sip-cgi, which is transaction-oriented (although cross-transaction state can be maintained with some effort), a SIN mechanism should be call-oriented so as to correspond more closely to the BCSM. Initial work on mapping between the SIP and IN call models has been done in the IPTEL group [5, 6, 7]; it may be appropriate to continue this work in an IN-focused venue such as SIN.

3. Interface between SIP servers and INAP/SCP (SIN)

When interworking between VoIP and IN-PSTN networks the main issue is to translate between the states produced by VoIP signaling and those used in traditional IN environment.

3.1 The concept of state in SIP

In a SIP call, only UAs have to maintain SIP call state. All other servers can be either stateless or, in the case of proxy servers, only maintain transaction state. (Transaction state is maintained for a single SIP request only.) Proxy servers can choose to be call-stateful if necessary. In that case, they generally ensure their presence in all SIP signaling exchanges by adding their name to the SIP Record-Route list. However, the soft switch acts as a SIP UA, so this is of no concern.

3.2 SIN issues

When interworking between VoIP networks and IN, it is useful to have a common understanding of how to translate from the states produced by VoIP signaling to those produced by IN signaling.

In this model, each SIP server is pre-configured to communicate with one logical SCP server, using whatever communication mechanism is appropriate. (In particular, the mechanism may not use an IP network.) Different SIP servers (e.g., those in different administrative domains) may communicate with different SCP servers, so that there is no single SCP server responsible for all SIP servers.

This proposal is applicable only when SIP-controlled Internet telephony devices are to interoperate with PSTN devices. The SIP UAs using this interface would typically appear together with a media gateway. It is *not* applicable within an all-IP network and is not

needed where PSTN media gateways (not speaking SIP) need to communicate with SCPs.

This proposal is not a wire protocol, but rather an abstract interface mechanism that simplifies the construction of SIP servers that want to access IN services. (On initial inspection, it does not seem appropriate to repackage SIP requests and responses.)

Since SIP proxy and redirect servers do not have access to the media data path, special mechanisms need to be deployed to enable IN services such as digit collection or announcement services. Announcement services can use the mechanism in [8] to deliver messages or can proxy the call to a SIP UAS that also terminates the data stream. That UAS may then transfer the call [9] to the final destination.

4. Related IETF efforts

The IETF IPTEL, MEGACO, PINT, SPIRITS, and SIGTRAN WGs deal with related interworking issues. Thus, it may turn out that this work can be accommodated within an existing working group.

IPTEL: The scope of this WG are perhaps closest to the present proposal. In fact, the original proposes for call model mapping have all been presented to IPTEL. Yet IPTEL deals with two specific items, namely Call Processing Language and TRIP, which are unrelated to the problem discussed here.

MEGACO: This WG develops the Media Gateway Control Protocol, which operates between the MG and the MGC and does not directly intersect with INAP and service issues.

PINT and SPIRITS: These WGs address the other side (relative to the soft switch) of Figure 2, namely interworking of service control with Internet-provided services. As such, neither group has anything to do with the soft switch or VoIP-PSTN gateways and, consequently, the proposal.

SIGTRAN: This WG specifies a transport protocol for carrying SS7 messages, which is orthogonal to service interworking.

5. Security Considerations

Since the soft switch has access to services in the PSTN network, it needs to be secured to prevent unauthorized use. However, since no protocol is being specified, this is primarily an operating system access control issue.

6. Conclusion

This document has identified a new work item for the IETF to define a mechanism that simplifies the interworking of SIP-enabled

soft switches with SCPs speaking INAP.

7. References

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[8] S. Donovan, H. Schulzrinne, J. Rosenberg, M. Cannon, and A. Roach, "SIP 183 Session Progress Message", Internet Draft, <draft-ietf-sip-183-00.txt>, October, 1999, work in progress.

[9] R. Sparks, " SIP Call Control Transfer", Internet Draft, <draft-lennox-sip-cgi-04.txt>, July, 2000, work in progress.

8. Acknowledgments

Janusz Dobrowloski, Vijay Gurbani, Frans Haerens, Jack Kozik, Warren Montgomery, and Jonathan Rosenberg contributed to the discussions on the relationship of IN and SIP call models. (Janusz was the first to bring the discussion to the IETF.)

9. Authors' Addresses

Henning Schulzrinne
Department of Computer Science
Columbia University

Phone: (212) 939-7042
Email: hgs@cs.columbia.edu

Lev Slutsman
AT&T Labs
200 Laurel Avenue South
Middletown, NJ 07748
Phone: (732) 420-3752
Email: lslutsman@att.com

Igor Faynberg
Bell Labs/Lucent Technologies
Room 4C-611A
101 Crawfords Corner Road
Holmdel, NJ 08816
Phone: (732) 949-0137
Email: faynberg@bell-labs.com

Hui-Lan Lu
Bell Labs/Lucent Technologies
Room 4C-607A
101 Crawfords Corner Road
Holmdel, NJ 08816
Phone: (732) 949 949-0321
Email: hui-lan.lu@lucent.com

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11. Appendix

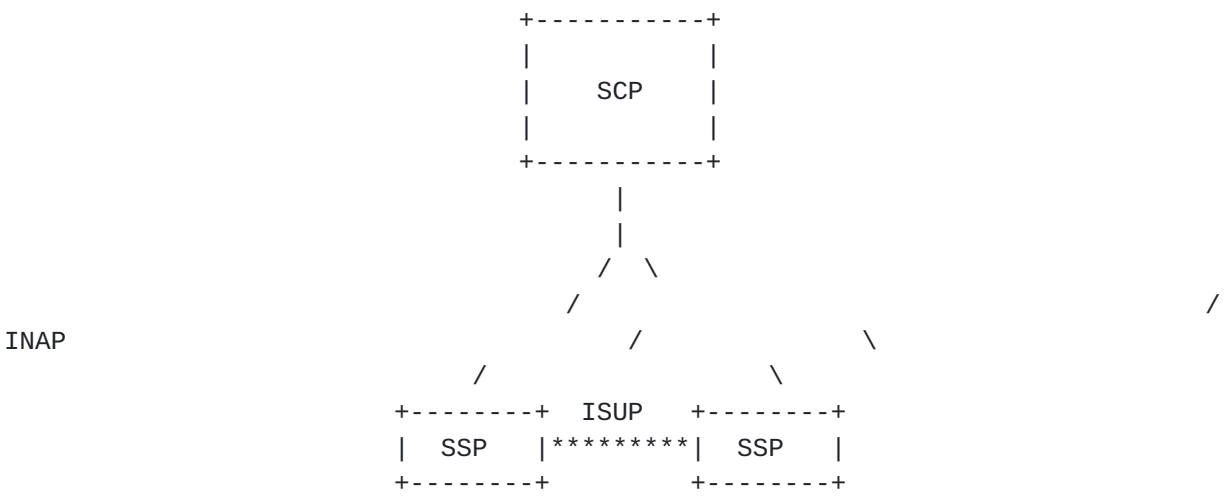

Figure 1. Simplified IN Architecture

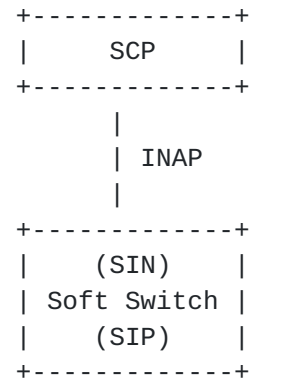

Figure 2. Simplified soft switch architecture

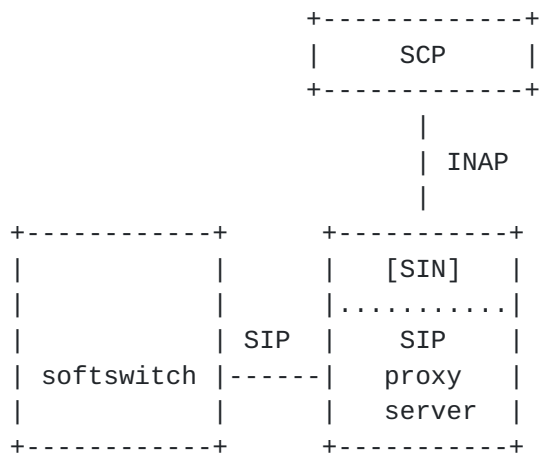

Figure 3. SIP proxy using INAP as a location server