INTERGRATION OF THE IMS CORE AND THE IP PBX: A CASE STUDY
MỘT GIẢI PHÁP TỔ HỢP PHÂN HỆ ĐA PHƯƠNG TIỆN TRÊN NỀN IP (IMS) VỚI IP PBX

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ABSTRACT

The improvement of communications is not only create new and fantastic services, but also implies the migration and simplification of existing services. In today's public networks, even NGN, there are neither products nor standardized solutions to provide corporate Private Branch Exchange (PBX)-like services. Currently, there are plenty of IP PBXs installed at corporate with rich set of IP Centrex features, e.g. call forwarding upon busy, call barring, etc. The example of such system is Asterisk, the most popular and open source IP PBX today, which is using SIP as session control protocol. While in the public side, IMS is approaching as the unique solution for the core of 3G and LTE networks. There is, therefore, a demand for integration and seamless interoperability between two domains, namely IMS at the core and IP PBX at the access/service layer. This short paper presents our initial experimental works for solving this problem. In our solution we propose to use Asterisk as the feature server on top of the Fokus' OpenIMSCore to provide the IP Centrex services to IMS users. The proposed architecture will also use SER as IMS Gateway to mediate between non-IMS Asterisk and the IMS core components. The article also presents the test-bed that was implemented in order to prove the concepts and presents key considerations for the implementation.

TÓM TẮT

Sự phát triển của các hệ thống viễn thông không chỉ dựa trên việc tạo ra các dịch vụ mới mà còn là quá trình kết hợp và đơn giản hóa các dịch vụ viễn thông có sẵn vào các hệ thống mới. Trong phần lớn các mạng công cộng ngày nay kể cả các mạng NGN đều không có các sản phẩm hoặc các giải pháp đã được chuẩn hóa để cung cấp các dịch vụ tương tự như trong hệ thống PBX. Hiện nay các tổng đài IP PBX đều có thể cung cấp một loạt các dịch vụ phong phú như chuyển cuộc gọi khi bận, chặn các cuộc gọi không mong muốn .v.v. Một trong các hệ thống PBX thông dụng hiện nay là Asterisk. Về phía mạng công cộng, IMS (IP Multimedia Subsystem) là một nền tảng đã được chuẩn hóa cho các mạng 3G và LTE (Long-Term Evolution). Do đó nhu cầu tổ hợp và cung cấp giải pháp tương thích cho IMS ở mạng lõi và các tổng đài IP PBX ở mạng truy cập là một nhu cầu thực tế. Bài báo này đưa ra một giải pháp sử dụng Asterisk như một máy chủ ứng dụng cho hệ thống OpenIMSCore được phát triển bởi Fraunhofer FOKUS (CHLB Đức) để cung cấp các dịch vụ IP Centrex cho người sử dụng IMS. Bài này cũng sẽ trình bày quá trình triển khai một hệ thống thử nghiệm cũng như đề cập đến các vấn đề cần phải giải quyết trong quá trình triển khai

I. INTRODUCTION

Supplementary services for telephony are a fixed part in today's communication world. Everybody uses features like call forwarding, completion of calls on non-reply or holding. These features are provided by a centralized local or private branch exchange. Due to centralization this approach has many dis-advantages. Administration and maintenance of an exchange are very expensive as usually every terminal needs a separate line. Furthermore, in case of a breakdown every connected client is affected and the whole system will be unavailable. In order to save costs and provide resilience, Internet Multimedia Subsystem (IMS) [1] provides a reasonable approach to solve these problems. In this paper we describe how we can realize network-centric supplementary telephone services using an IMS services creation environment. We developed an extensible testbed/framework in which we can add supplementary telephone services to common IMS core networks. Furthermore, thanks to IMS mobility feature, this approach is platform independent and can be used both on stationary and mobile devices. The paper is organized as follows. Section 2 gives a summary introduction to IMS technology and a more detailed explanation of the IMS service creation platform. Afterwards we give an overview over supplementary telephony services in current switched telephone networks which will be realized in our overlay IMS architecture. Section 3 describes how these services can be realized using IMS Core functions and the so called Telephony Application Server (TAS). Conclusions and directions for future work are presented in section 4.

II. IMS SERVICE ARCHITECTURE

2.1 IMS Core Layer

Figure 1 shows a simplified view of the key components in an IMS Network.

The IMS follows the NGN standardized architecture that's inclusive of three layers of transport/access infrastructure, control and signaling functions and services creation and execution environment. In IMS the control and signaling layer is generally called IMS Core. IMS Core contains following key components featuring the session control and user/access authorization.

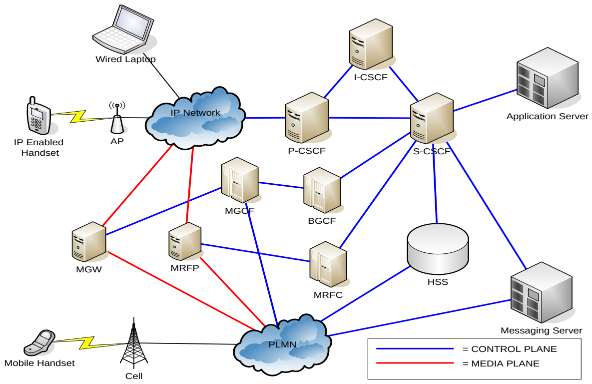

Fig.1 IMS key components

- **HSS:** A key component of the converged IMS solution is the concept of the HSS or Home Subscriber Service. The HSS is a centralized control and management point that controls a subscriber's devices, preferences, and features.

The HSS knows what devices a subscriber has, which ones are registered on the network, and how to contact each of them. This information can be very powerful when combined with an effective presence application. For example, you may specify that between the hours of 8 and 5 PM all calls must be routed to your work VoIP terminal. Evening calls must be routed to your home phone. When you are in transit, calls are routed to your cell phone. If you are in an important meeting, you want to send all calls to voice mail right away. The HSS ties all of your devices together; telephony and presence applications can therefore be consistently applied across multiple devices.

P-CSCF, S-CSCF, I-CSCF

IP enabled devices become active on the IMS network by registering with a SIP proxy that knows each device is valid and what subscriber it belongs to. In IMS, the call control SIP proxies are called Call Session Control Functions (CSCFs). Each IP based device must register in the IMS network, and these registrations, as well as all subsequent requests to initiate or modify communications sessions, must traverse the CSCFs. All communications between the subscriber devices and the CSCFs occur in the MEDIA PLANE.

The IMS model defines three different types of CSCF

- P- CSCF: Proxy CSCF. This is the CSCF that the each device sends all IMS control plane traffic to. Which P-CSCF you connected to depends on what physical IP network your device in.

- S- CSCF: Serving CSCF. This is the CSCF that is providing the call control and applications support to the User. When a device registers, the P-CSCF sends the registration request to the owning S-CSCF. The S-CSCF uses information stored in the HSS to authenticate the device, as well as determine what services and preferences exist for that device. The S-CSCF can include specialized application servers (presence, telephony, messaging, etc) in session control traffic as appropriate.

- I-CSCF: Interrogating CSCF. This is needed when a device first tries to register with a P-CSCF, when the P-CSCF does not know which owning S-CSCF to send control messages to. The IMS has defined procedures for a P-CSCF to query an I-CSCF to determine the correct S-CSCF for a given device/subscriber.

Once a device has been registered with the IMS S-CSCF, all subsequent transactions will go from the device to the P-CSCF to the S-CSCF. The S-CSCF will forward messages to application servers when specific service triggers are invoked, or may simply forward the transactions to the called party or another CSCF. The S-CSCF caches HSS information for the subscribers, and may read more information later in support of specific session requests.

For simple IMS to IMS calls, the CONTROL PLANE may reside in a single S-CSCF. The Media Plane for a simple VoIP IMS call (ex: the IP enabled handset in the diagram calls the wired laptop) may be direct from the calling device to the called device. The use of firewalls and Session Border Controllers increase the complexity of the Control and Media Planes.

BGCF, MGCF, MGW

For IMS sessions that leave the IMS domain, such as PLMN and PSTN calls, there is a need for both control and signaling gateway functions between the networks. The IMS components of this gateway functionality are:

- BGCF: Breakout Control Gateway Function. The S-CSCF communicates with the BGCF using SIP. The BGCF identifies the appropriate MGCF and MGW that will be used to support a specific call instance. This is required when a session passes outside (Breaks Out) of the IMS domain. The routing for the BGCF is based on telephone numbers.

- MGCF: Media Gateway Controlling Function. The MGCF converts the Control Plane information on the IMS side to the specialized signaling used in the PLMN/PSTN network, and vice versa. This allows the information needed to initiate, modify, or terminate a session to be passed between networks that use different signaling systems. The MGCF is the Control Plane Gateway.

- MGW: Media Gateway. The MGW acts as the interface between the IMS device Media stream and the PLMN / PSTN device media stream. The MGW is the Media Plane Gateway.

Using the S-CSCF, BGCF, MGCF, and MGW, sessions can be requested (Control Plane) and established (Media Plane) between the IMS and other networks.

MRFC, MRFP

Another important function in the network requires the use of Media Resource Servers. These media functions are needed for:

- Collection of DTMF digits contained in the audio path.
- Playing of announcements (audio/video)
- Multimedia conferencing (e.g. mixing of audio streams)
- Text-to-speech conversion (TTS) and speech recognition.

In the IMS network, these Media Resource Functions are provided by the:

- MRFC: Media Resource Control Function. The MRFC is a SIP proxy that controls the MRFP. THE MRFC is in the Control Plane.

- MRFP: Media Resource Control Processor. The MRFP performs the required media functions by sending or receiving media. The MRFP is in the Media Plane.

In the converged 3G environment, the IMS, PLMN, and PSTN should use Media Resource Servers that are capable of supporting all three networks.

2.2 IMS Service Layer

The application layer host and execute services, and interfaces with the CSCF and each other using SIP. The user database (HSS) that supports the IMS network entities that are actually handling the calls/sessions is put in this layer. When multiple HSSs are used the *Subscriber Location Function (*SLF) is needed

The AS can operate in SIP proxy mode, SIP US (user agent) mode or SIP B2BUA (back-to-back user agent) mode, depending on the actual service. An AS can be located in the home network or in an external third-party network. If located in the home network, it can query the Home subscriber server (HSS) with the DIAMETER Sh interface (for SIP-AS and OSA-SCS) or the MAP interface (for IM-SSF).

The core component of the service/application layer is the so-called service delivery platform (SDP) which includes of SIP servers (or other application servers through gateway function) and several service enablers.

- SIP application server: The SIP application server acts as the central component of an operator's SDP for its IMS (real-time, IP-based) services. This software-based platform uses Session Initiation Protocol (SIP) [2] [12] as the signaling protocol to enable real-time communication sessions in IP networks. SIP application servers provide a service creation environment and developers' toolkit for operators to create and host new services. They also typically include process logic to orchestrate and mediate transactions. In general, application servers (including both web and SIP) are part of an integration backbone and incorporate process logic that initiates synchronous transactions to other network element and software systems. Upon initiating these transactions, the SIP application servers mediate the cascading commands and responses from the various network elements in real time. SIP application servers provide the foundation infrastructure for operators to create and deliver unique real-time service applications (i.e., instant messaging, ad hoc conferencing, real-time gaming, Wi-Fi/3G seamless mobility). In the today communication evolution, there demand for (public) IMS operators to provide the enterprises with PBX-like services, in that context SIP application server can act as the telephony application server (TAS) to host supplementary services. SIP application servers integrate directly into the IMS components in the network control layer. For the sake of implementation and operational efficiencies, most vendor solutions have this process logic functionality embedded in their application servers. Operators may also centralize some business logic on the SIP application server rather than distribute different pieces of business logic in different SDP components.

- Policy management: Policy management is the heart of an operator's network-facing SDP. An SDP incorporates many different flavors of policy management into its network to deliver new services, many of which will originate with third parties. Policy management rules range from enforcing how third-party developers or content providers will access an operator's network; establishing prepaid billing limitations, bandwidth policy considerations and location service priorities; and enforcing policy requirements that subscribers set (i.e., parents specifying who their child may contact and when they may receive cell phone access).

- Location server: The location server acts as a central gateway integrating information contained in multiple systems (i.e., location-based applications, geographic information system [GIS] information and content applications to determine and send out location positioning for subscribers). The location servers may also contain process logic to determine, for example, prioritization of service for different applications that request access to the location server (i.e., public safety applications).

- Presence server: Presence servers provide information on subscriber availability to other parties; subscribers may elect to alter the presence information available for different parties. Subscriber information outlining contact preferences may be contained in either a home subscriber server (HSS) or preference/profile server module. Subscribers may opt-in to provide presence information to third parties that would market or advertise directly to them. In this instance, presence and location information may combine to provide more targeted offerings to subscribers.

- Service broker: The Service Capability Interaction Manager (SCIM) [13] is a service broker that manages and executes discrete processes within the SDP that are required to deliver a service. For example, the SCIM can contain the business logic to determine how customer service requests may be distributed among various application server or media server resources. If a customer has paid for dedicated support, specific resources may be set aside for that customer; and the SCIM would direct these customer requests to specific rather than general shared resources. The SCIM or service broker may also manage business logic for combining specific application resources. 3GPP has not yet provided a very explicit definition of the SCIM concept. The level of separate transactions required to provide services in the current mobile environment is limited because operators are not yet offering more complex composite services, for example, that combine presence, location, voice or video.

Telephony Supplementary Services

Common communication carriers provide their customers a great variety of supplementary telephony services [3]. These services are realized on centralized exchanges with some disadvantages. These supplementary services are for example:

- Calling Line Identification Presentation/Restriction is a telephony network service that transmits the caller's telephone number to the called party or blocks the transmission. The state for the indication is set by the caller and evaluated in the telephone exchange. The exchange is responsible for proceeding the correct information to the called party.

- Completion of Calls to Busy Subscribers/on No Reply is a telephony network service that allows the calling party to automate the call origination in the telephone exchange for the unavailable subscriber. Because the exchange as a centralized instance knows the state of every participant, it can initiate the callback if the desired destination is available again.

- Call Waiting is a feature that a subscriber gets an audio signal during an active connection signaling that another party wants to establish a connection. The called party doesn't get the congestion signal but the ringing signal. The party informed by the audio signal can decide if the waiting call should be rejected, accepted by terminating the current call or accepted by putting the current call on hold.

- Call forwarding is a feature where an incoming phone call is redirected to another party. The diversion can occur immediately, after a certain time or if the called party is busy.

- Three-Party-Conference is a feature whereby a party can establish a conference between itself and two other parties. Afterwards every party can hear the others at the same time. This feature is normally controlled by the telephone exchange but can also be provided by the end device of the originator. In the last case the originator needs at least two phone lines.

- Hold is a telephony network feature where during a call a party can put the remote party on hold. The party on hold is receiving music on hold from the exchange. The initiated party has the possibility to switch to another phone and resume the connection. This procedure has to be finished within a certain time; otherwise the connection to the party on hold will be terminated by the exchange.

Some of the features shown above are to be realized within our IMS testbed framework. In the next section we will see how the feature of Call forwarding upon busy could be realized using Fraunhofer's IMSOpenCore and open source Asterisk SIP server.

III. IMS-BASED CALL FORWARDING UPON BUSY

In this Section, we will present a proof of concept implementation of the CFB feature in IMS environment.

3.1 Service Concepts and Deployment Scenario

In our research, we consider a scenario in which an IMS user (hereafter is called Alice) initiate a call to another IMS user (hereafter is called Bob) which has registered to the CFB service. Unless otherwise indicated, bob1@ims.hut.edu.vn and bob2@ims.hut.edu.vn are used to represent the other end points of Bob that communicate with Alice; they may or may not be in the same network as Alice but for its simple here we're assuming that they are in the same domain and Alice is at her home domain. Therefore I-CSCF may or may not include in the call flow. The call scenario (Figure 3) would be like that:

Alice calls Bob at bob1@ims.hut.edu.vn, where Bob is registered to the call forwarding service. We have a TAS function that host the call forwarding logics in order to forward the call to bob2@ims.hut.edu.vn in the condition that bob1@ims.hut.edu.vn is in busy state.

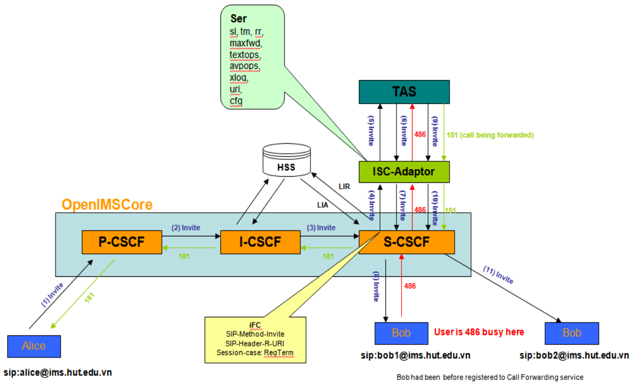

Fig.2 Request routing scenario of IMS-based call forwarding feature

The overview of request flow would be like this (Figure 3):

- Alice (sip:alice@ims.hut.edu.vn) sends SIP-INVITE message to the S-CSCF of home domain via the P-CSCF. The 'To' header is sip:bob1@ims.hut.edu.vn.

- S-CSCF of home domain (or terminating domain in case of inter-domain call) analyses the Initial Filter Criteria of the bob's profile and sees whether there is any such IFC which is met (trigger=Sip Method =INVITE and Session Case=Terminating Registered and Sip Header =R-URI). Since, assuming, bob registered to CFB service then the SIP INVITE message is forwarded to the Asterisk via ISC-Adaptor (SER– for manipulating the SIP headers to be compliant to IMS specifications) for processing.

- TAS, configured as the B2BUA, does some modification to the Invite/Via and sends the modified Invite back to S-CSCF via ISC-Adaptor. S-CSCF, in its turn, forwards the modified Invite to bob1@ims.hut.edu.vn. This subscriber response with 486-busy here then this 486 goes back, via the S-CSCF, to the Asterisk

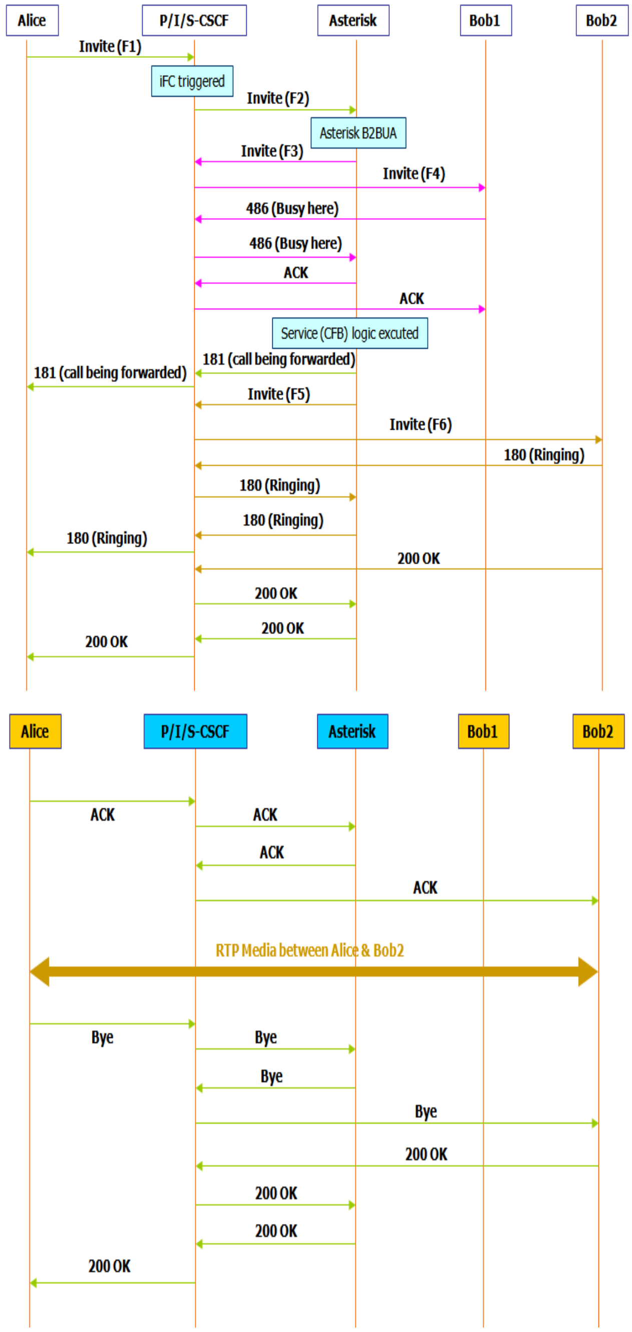

Fig.3 The message flow between entities

- TAS runs its pre-configured call forwarding service logic to create the new Invite and send back to S-CSCF with the R-URI of bob2@ims.hut.edu.vn, and simultaneously send back the *181-call being forwarded* to the originating party (alice@ims.hut.edu.vn) to indicate call forwarding is ongoing.

- After exchanging the provisional messages (ringing, ok, ack) then the call is established between alice and bob2 at the same domain

3.2 Proof of Concept Implementation

Figure 4 shows the structure of the Hanoi University of Technology (HUT) IMS testbed. All IMS network nodes and SER are running Ubuntu Linux 7.10 on x86-based hardware. The Asterisk is on Redhat Linux 4.0. As IMS CSCFs we use the latest OpenIMSCore development snapshot [4]. For ISC-Adaptor, we use SER [5], extended to implement the SIP extensions according to RFC 3455 [6] (3GPP headers), RFC 3327 [7] (Path header), and RFC 3608 [8]. The IMS terminals are UCT IMS client [9]. Alternatively we have modified the SIPp SIP traffic generator [10] that is now capable to generate and terminate IMS signaling traffic. For testing purposes we integrated an Asterisk Server [11] to play the role of an IMS Telephony Application Server within our network. The Asterisk Server is configured and extended to do several tasks for supplementary services (e.g. digit analysis) and to function as the SIP B2BUA.

IV. CONCLUSIONS AND DISCUSSION

In the article, we present an experiment approach to implement the PBX-like telephony supplementary services with IMS technology with example of Call forwarding upon busy. The approach has some advantages as the followings:

- Allowing of the fast and smooth migration from the traditional PBX services in the enterprise domain to the all-IP operator domain.

- Proofing of the possibility that IMS operators can offer the legacy telephony supplementary services to their corporate customers

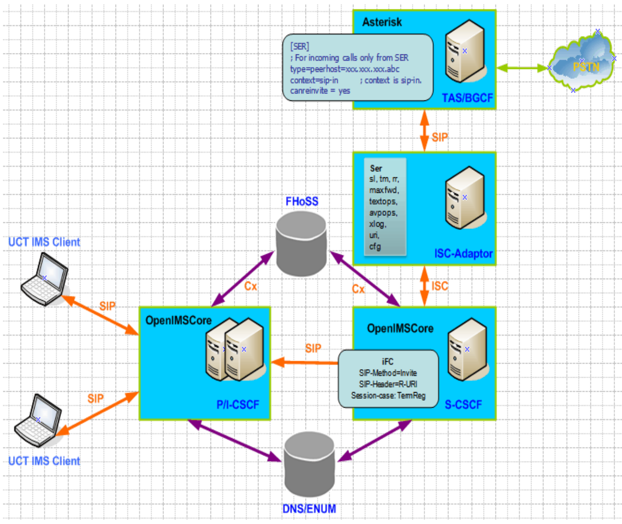

Fig.4 Implementation Testbed

- Utilization of the existing open sources for most of the testbed components with minimum additional workload.

- On the other hand, there are some further issues that will be investigated and implemented in the near future:

- The current implementation is only for call forwarding upon busy as a proof of concept. Therefore, in the near future, we will proceed with the full implementation of PBX-like supplementary services on the TAS.

- We also will investigate more inside of the Asterisk to extend it for fully comply to the 3GPP/3GPP2 IMS ISC interface with the Core. Besides that we intend to implement the ENUM services for the Asterisk and OpenIMSCore in order to make the call to Tel URI and ultimately to make Asterisk to functions as the IMS BGCF.

- Finally we would also study to develop the TAS using other service creation technologies like SIP servlet and JSLEE.

REFERENCES

1. TS 23.328; IP Multimedia Subsystem; 3GPP, Release 6.
2. *J. Rosenberg, et al.* ; SIP: Session Initiation Protocol ; RFC 3261, IETF, June 2002
3. Siemens Communication Lexicon: Siemens Enterprise Communications GmbH &Co. KG, http://networks.siemens.com/communications/lexicon en/index.htm

4. Open IMS Core Playfround, see http://www.openimscore.org/

5. SIP Express Router, see http://www.iptel.org/ser

6. IETF RFC 3455: "Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)", January 2003

7. IETF RFC 3327: "SIP Extension Header Field for Registering Non-Adjacent Contacts", December 2002

8. IETF RFC 3608: "SIP Extension Header Field for Service Route Discovery During Registration", October 2003

9. UCT IMS Client, see http://uctimsclient.berlios.de

10. OpenSource SIP Protocol Testtool, see http://sipp.sourceforge.net/

11. Asterisk, the open source PBX & Telephony platform, see http://www.asterisk.org

12. *M. Handley, V. Jacobson*; SDP: Session Description Protocol; RFC 2327, IETF, April 1998

13. Service Capability Interaction Manager, see http://en.wikipedia.org/wiki/Service_Capability_Interaction_Manager

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