Function test preparation of MSC-IMS signalling interaction in simulated environment

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Abstract: **Mobile Switching Center (MSC) is a part of GSTN network and it provides mobile exchange capabilities. In many cases signalling protocol between two MSCs is ISUP. IMS is the Internet Protocol (IP) Multimedia Subsystem defining a generic architecture for offering Voice over IP and multimedia services. SIP is the main signalling protocol used in IMS networks. The paper gives a basic overview of the function test preparation of signalling interaction in simulated environment between control plane using ISUP and call and session control functions using SIP.**

1. INTRODUCTION

A new communication culture is emerging driven by trends for new multimedia services demands based on IP. IMS offers a standardized way to deliver convenient IP-based consumer and enterprise services to fixed, mobile and cable community - enabled by one common core and control. It is the cornerstone of the evolution of current networks to a single, all-IP based network where all types of services (messaging, telephony, etc.) and media (voice, video, pictures, text, etc.) can be integrated into a single user experience[3].

IMS is developed to interaction with existing networks. This paper describes an IMS architecture design with existing GSTN networks translated to simulated environment for the purpose of software testing the signalling interaction between SIP and ISUP protocols.

2. NEXT GENERATION NETWORK ARCHITECTURE

One key benefit of IMS is that it enables true convergence, and interactions in several dimensions – across fixed and mobile access – in the service layer, control layer and connectivity layer [2].

The telecommunication community is migrating towards a new network architecture Fig. 1 based on horizontal layers [1]. The change in the network architecture introduces new logical network nodes and also changes the role of existing nodes in the network. This architecture changes the current vertically specialized network into a horizontal layered architecture.

The traditional network structure – with its service-unique functionality – is very complex and costly to build and maintain as the number of services grows.

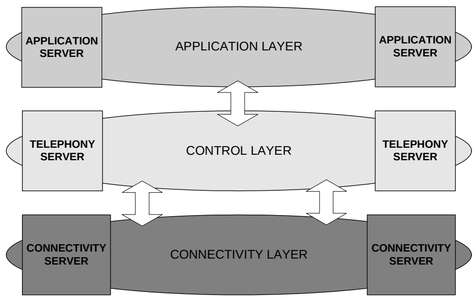

Figure1. Horizontal layered architecture

In practice the layering means that different levels in network hierarchy are separated and communicate over well-specified interfaces; thus different applications share resources in the lower level of the network. Ericsson's solution for the next generation networks adopts the layering principles outlined above [1].

The layered Core Network architecture is derived from the current standards reference model by separating the control plane functions in the SN from their user plane functions, thus turning these nodes into Servers and Media Gateways (Connectivity Servers).

3. IMS STRUCTURE

The role of IMS is to provide a secure and reliable means for terminals and applications to reach, negotiate and communicate with each other. This facilities for the operator to provide multiple services to the user and maximizing equipment re-use through horizontalization. The horizontalization provides common; supervision and control of services in the IMS network, management and routing of sessions, as well as supporting the authorization and manipulation of media in the network. In the IMS specification the "core" of IMS comprises two main nodes: the Call Session Control Function (CSCF) and the Home Subscriber Server (HSS). In the IMS architecture overview (Fig. 2) the General Switched Telephony Network (GSTN) interactioning functions Media Gateway Control Function (MGCF) have been depicted beside the IMS Core [2].

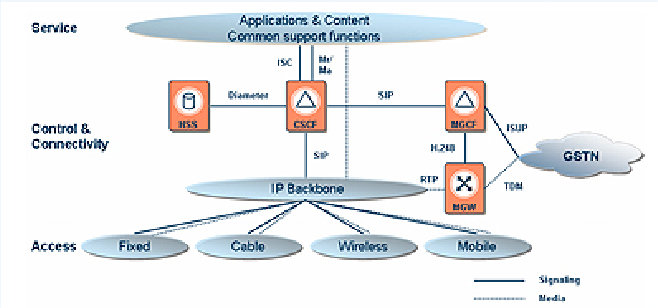

Figure 2. IMS architecture overview

3.1 Call Session Control Function (CSCF)

The Call Session Control Function (CSCF) is the heart of the IMS architecture and is used to process SIP signalling. The main function of the CSCF is to provide session control for terminals and applications using the IMS network. Session control includes the secure routing of the SIP message, subsequent monitoring of the SIP sessions and communicating with the policy architecture to support media authorization. It has also the responsibility for interacting with HSS. The CSCF plays three different roles: Serving Call Session Control Function (S-CSCF), Interrogating Call Session Control Function (I-CSCF) and Proxy Call Session Control Function (P-CSCF) [2].

3.2 Home Subscriber Server (HSS)

The Home Subscriber Server (HSS) is the master database that contains user and subscriber information to support the network entities handling calls and sessions. It provides the following functions: identification handling, access authorization, authentication, mobility management (keeping track of which session control entity is serving the user), session establishment control, service provisional support, and service authorization support. When a user registers in the IMS domain, the user profile (relevant information related to the services to be provided to the user) is downloaded from the HSS to the CSCF. For session establishment, HSS provides information on which CSCF currently serves the user [2].

3.3 Media Gateway Control Function (MGCF)

The Media Gateway Control Function (MGCF) is the central node of the PSTN gateway. The MGCF is responsible for controlling the media resources used when traffic needs to flow between networks using different media, typically between a Time Division Multiplexing (TDM) network and an IP-based network. It interacts with: the call and session control function using SIP; the control plane of the GSTN using ISUP; and with the Media Gateway using H.248 protocol as illustrated in Fig. 3[2].

The Media Gateway (MGW), controlled by the MGCF using H.248, is responsible for providing the interaction of the media flows between different networks. It provides interactioning between the different media transport formats, RTP/UDP/IP and TDM, as well as media transcoding of voice and video, if required.

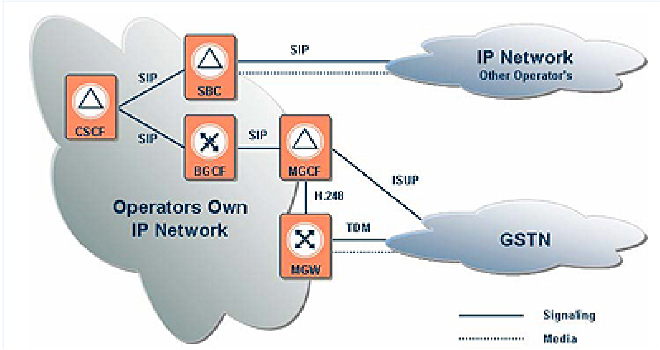

Figure3. The IMS – IP Network/GSTN interface

3.4 Breakout Gateway Control Function (BGCF)

The Breakout Gateway Control Function (BGCF) is responsible for selecting break out operator and/or site for outbound session to the GSTN. It is the logical entity within the IMS network that decides how to route the Telephony session initiated in the IMS network and destined for a circuit switched network (GSTN). The circuit switched networks can be any legacy network, PSTN or other wireless networks. If breakout occurs in the IMS network, then the BGCF routes the session to a Media Gateway Control Function that then allocates a Media Gateway or the BGCF routes the session to a BGCF in another operator network as illustrated in the Fig. 3 [2].

4. SIP OVERVIEW

SIP is an IETF application-layer control protocol that can establish, modify, and terminate multimedia sessions (conferences) such as Internet telephony calls [5]. SIP can also invite participants to already existing sessions, such as multicast conferences. Media can be added and changed to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility - clients can maintain a single externally visible identifier regardless of their network location. SIP is not a vertically integrated communications system. It is structured as a layered protocol, which means that its behaviour is described in terms of a set of fairly independent processing stages with only a loose coupling between each stage. SIP is a component that can be used with other IETF protocols to build complete multimedia architecture. SIP was designed to be a modular component of a larger IP telephony solution and thus functions well with a broad spectrum of existing and future IP telephony protocols.

SIP service technologies can be integrated into WWW technologies that are used widely today. Through the adoption of these technologies the 3G network will be able to provide rapid and inexpensive services and, probably most importantly, there will be a mass of people with the ability to provide the wealth of services desired. SIP is one of the key protocols used to implement VoIP. Although performing telephony call signalling and transporting the associated audio media over IP yields significant advantages over traditional telephony, a VoIP network cannot exist in isolation from traditional telephone networks. It is vital for a SIP telephony network to interaction with the Core Network. An IP network using SIP may serve as a transit network between gateways - a call may originate and terminate in the Core Network, but cross a SIP-based network somewhere in the middle [4].

SIP telephony network is transparently with respect to the Core Network. Traditional telecom services such as call waiting, freephone numbers, etc., implemented in protocols such as SS7 should be offered by a SIP network in a manner that precludes any debilitating difference in client experience while not limiting the flexibility of SIP.

SIP telephony network is routability of SIP requests - a SIP request that sets up a telephone call should contain sufficient information in its headers to enable it to be appropriately routed to its destination by proxy servers in the SIP network [1]. SIP has possibility of querying for the capabilities of a SIP server or client using OPTIONS, or cancelling a pending request using CANCEL. SIP does not provide services. Rather, SIP provides primitives that can be used to implement different services [1]. SIP does not transport media streams. SIP is used for signalling only and it is not a session description protocol. The details of the session, such as the type of media, codec, or sampling rate, are not described using SIP. SIP does not offer conference control services and does not prescribe how a conference is to be managed. It does not have any baseline mechanism to carry any mid-call information along the SIP signalling path during the session. SIP is not an easy protocol to secure. Its use of intermediaries, its multi-faceted trust relationships, its expected usage between elements with no trust at all, and its user-to-user operation make security far from trivial [5].

5. SIMULATED ENVIRONMENT FOR SOFTWARE TESTING THE SIP-ISUP INTERACTION

The simulated environment is chosen for software testing the SIP-ISUP interaction on the functional level because most of the functions can be software tested in this environment as it is a relevant representation of the real environment. The main reason is for that because the costs of these tests are much less than the cost if they were executed in the real environment. The costs for performing the function test in the real environment (with real nodes) are cca. 50.000 EUR/week. The additional problem of the real environment is that since it is used from many different units it needs to be booked at least couple of months in advance. Often a situation is that a booked environment is useless due to the current status because there were delays in software developing process. For the simulated environment it is only necessary to have PC or working station to execute the simulated softwares properly. This gives a big advantage to software testing in the simulated environment.

The simulated environment is divided into three different parts as illustrated on the Fig.4 :

(1) SEA – representing MSC node
(2) SIP protocol generator
(3) ISUP protocol generator

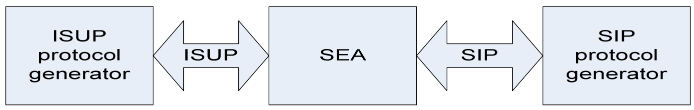

Figure 4. Simulated environment network

SEA (Simulated Environment for AXE) representing MSC node has all the features as the MSC node in the real environment. It has the same version of software loaded. It has a terminal used to give MML commands in order to set an exchange data. An exchange data is a number of MML commands which are ordered to set up a working environment. For this particular case a signalling terminals are used to send/receive ISUP messages and UDP/TCP sockets are used to send/receive SIP messages. They have been defined through MML commands. Once the environment is being set, it is ready to send and receive SIP and ISUP messages. ISUP protocol generator represents ISUP node that has all the ISUP messages built-in with all the mandatory and optional ISUP parameters. SIP protocol generator represents SIP node that has built-in SIP structure. For both protocol generators the TTCN-3 tool is being used.

SEA has built-in MGW feature which is important because it enables SIP-ISUP interaction . Since SIP is the main signalling protocol in IMS and ISUP is the main signalling protocol in GSTN network, interaction between ISUP and SIP in simulated environment represents MSC and IMS interaction in real environment.

6. BASIC CALL SCENARIOS

Basic call is a simple call case where a calling party or A-side calls a called party or B-side. It is always the first traffic test case which is being performed when new functionality is being implemented, in our case a functionality which enables a MSC-IMS signalling interaction. A basic call is considered as a positive test case. Later on in testing process many positive and negative test cases are derived from this test case.

6.1 Basic call initiated by SIP node

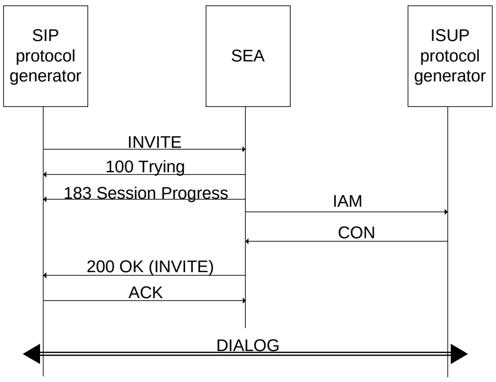

Figure 5. Basic SIP session signalling procedure

Basic call signalling sequence when SIP node initiates a call is shown on Fig.5. It represents one positive test case. In this test case is shown that before the called party is being alerted, a network resources have been reserved. This is typical for session establishment in SIP protocol. After 1xx responses have been received in SIP node, IAM message is received in ISUP node. ISUP node answers with CON message which is further translated to a 200 response to INVITE request. On 200 response ACK request is sent from SIP node and at this moment a dialog between two parties is started. From this test case many negative test cases can be derived. For example, on INVITE request responses 4xx and 5xx can be obtained if there is a failure or mismatch in the INVITE request.

6.2 Basic call initiated by ISUP node

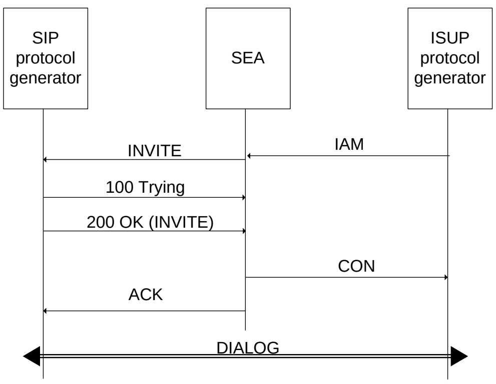

Figure 7. Basic SIP session signalling procedure

Basic call signalling sequence when ISUP node initiates a call is shown on Fig.6. In this test case IAM message is received in SEA and translated to INVITE request towards SIP node. SIP node answers with 100 Trying and 200 OK to INVITE request which is further translated to CON message towards ISUP node. ACK request is received on SIP node and at this moment a dialog starts between two parties. As in previous case many negative test cases can be derived from this test case. For example, when IAM message is sent to SEA, SEA answers with REL message because of the failure or mismatch in IAM message.

6.3 Call release initiated by SIP node

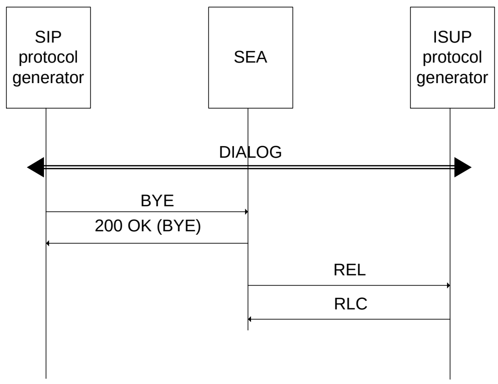

Figure 6. Basic SIP session signalling procedure

Signalling sequence when SIP node initiates a call release on a established dialog is shown on the FIG.7. SIP node sends BYE request and receives 200 response. ISUP node receives REL message and responds with RLC message. From this test case many negative test cases can be derived. For example, when BYE request is being sent from SIP node, a response 4xx or 5xx can be received.

6.4 Call release initiated by ISUP node

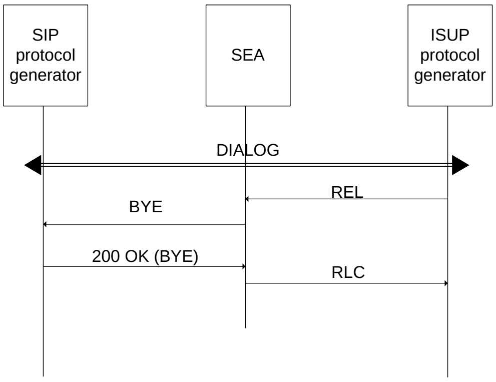

Figure 8. Basic SIP session signalling procedure

Signalling sequence when ISUP node initiates a call release on established dialog is shown on Fig.8. ISUP node sends REL message which is translated to BYE request to SIP node. SIP node responds with 200 response which is further translated to RLC message towards ISUP node.

7. CONCLUSION

IMS using IP as a bearer type and Call Control Protocol such as SIP will become a main force in the global Telecommunication industry in the next few years. Therefore an interaction with existing nodes such as MSC is one of the crucial requirements for both IMS and MSC development. Software testing of this interaction plays a big role in development process of the next generation networks.

Function test in simulated environment covering signalling interaction ISUP-SIP emerges as necessity in testing process since the costs of the testing in a real environment are very high. At this level of testing many faults and errors can be discovered before the software testing goes to system test on real nodes. This shortens the period of system tests which reduces the final cost.

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