Technical University of Denmark

DTU

Intelligent Services in Converged Networks - Evolution steps in the signalling arena

Soler, José; Fosgerau, Anders; Grabner, Boris

DTU Library
Technical Information Center of Denmark

Intelligent Services in Converged Networks - Evolution steps in the signalling arena.

José Soler Lucas_C
Anders Fosgerau_C
Boris Grabner_T

COM Center.
Networks Competence Area
Technical University of Denmark.
DTU-Building 345v-2800 Lyngby. Denmark.
Tel: +45 45256352   Fax: +45 45936581
www.com.dtu.dk
jsoler@com.dtu.dk ; af@com.dtu.dk

Telekom Austria AG.
Value Added Services
Arsenal Objekt 22 -1030 Wien. Austria.
Tel: +43590591-43117   Fax: +43 590591-43190
www.telekom.at
boris.grabner@telekom.at

While voice transport over IP is not any longer a dream but a reality, the capacity to offer IN-like services, as value added services within VoIP environments has still been rarely treated and implemented. In this paper we present an overview on the subject and the work currently on development, within the IST Project GEMINI, towards the implementation of IN IP-based services and its interoperabiliy with traditional PSTN-SS7-IN networks.

1. Introduction.

This paper aims to present the authors' view of the telephony to come.

Firstly an evolutionary description is shown in Section 2, presenting the transition from plain voice services towards intelligent services (IS) in traditional public switched telephone networks (PSTN). Section 3 presents the way in which the "packet network" trend affects PSTN and how the migration towards this technology realm may be translated in telephony environments.

Section 4 deals with the description of current proposals and standard protocols that allow this network convergence. Finally, section 5 describes the IST Gemini Project in which the authors are currently involved.

2. From plain services to the IN concept.

Subscribers of early telephony systems were used to deal with an operator that manually switched their connection requests towards destination in order to establish a successful phone call.

The increase in the number or subscribers triggered the use of electromechanical exchanges and the service logic was in this way automated. At this stage, the role of the manual operator shifted to provide assistance for more advanced services like conference calls or charge information. It can be said that the operator provided the intelligence required for value added services: those not related to the connection set-up but to its content.

When stored program controlled (SPC) technology generalized, some of these supplementary services were integrated into local exchanges, what allowed users to access services in the quick and transparent way we are already used to, and at the same time, allowed an increase

in the operator-company revenues from a wider span of fees.

The proliferation of SPC exchanges in telecommunication-operator (telco) networks forced the centralization of supplementary services in special network nodes in order to facilitate the update and introduction of such these services. This required databases for service and subscriber data as well as a supporting robust signalling system. This separation of the service logic from the switching logic was branded as network intelligence and gave birth to the concept of intelligent network (IN) [1].

2.1. IN & SS7.

It is not the aim of this article to provide a complete description of the SS7 and IN related standards. The interested reader should check [2] and [3] for a comprehensive description, as well as to the related documents published by ITU.

PSTN can be viewed as two networks working together: a circuit-switched, TDM-based, devoted to voice transmission and a packet-switched devoted to signalling, supporting the first one.

ITU-T Study Group 11 is responsible for the Switching and Signalling (Q.xxx) series of recomendations that define the signalling mechanisms in today telecommunication networks. In the mid 60's ITU-T, then known as CCITT, developed a digital signalling standard called Common Channel Interoffice Signalling System #6 (CCIS6). CCIS6 later evolved into C7, known as SS7, which has become the signalling standard for the telephony world [4]. ITU-T produced also the so-called Intelligent Network (IN) specification (Q.12xx) as a model for modern telephony operations allowing the management and creation of telephony services in a quick and standardized manner. The

IN Conceptual Model consists of four planes, each providing an abstract view of the IN capabilities, taking a top-down view in regard to services and a bottom-up view with respect to the bearer capabilities. It is the lowest plane, the physical plane, that defines physical entities (PEs) and the interfaces between the PEs. It is at this plane where IN bases on SS7. [2] [4].

The set of ITU-T recommendations Q.7xx defines the architecture, protocols and mechanisms to provide an internationally standardized general purpose common channel signalling (CCS) system optimised for operation in digital telecommunication networks in conjunction with stored program controlled exchanges, that can meet present and future requirements of information transfer for call control, remote control, and management and maintenance and provides a reliable means for transfer of information in correct sequence and without loss or duplication [3].

The so-called signalling points, the nodes for the signalling network, are the following:

- Service Switching Points (SSPs): These comprise the exchanges in the telephone network, providing the functionality to communicate within the voice switch via the use of primitives and creating the packets needed for transmission in the SS7 network. They house the Service Switching Function (SSF) of the IN.

- Service Transfer Points (STPs): These are the points that interconnect different SSPs, serving as routers in the SS7 network.

- Service Control Points (SCPs): These network elements provide interface to telephone company databases and so, are the enablers of IN services. They implement the Service Control Function (SCF) of the IN.

- Operations Support Systems (OSSs): These are the remote maintenance centres for monitoring and management of the SS7 and voice networks. To update the SCP databases and monitor the overall performance a Service Management System (SMS) is used. [3]. A diagram of a basic SS7/IN network is shown in figure 1.

SS7 functionality was spread in a layered protocol stack [5] as shown in figure 2 and explained as follows: The Message Transfer Part Layer 1 (MTP-1) is the equivalent to OSI Physical Layer. SS7 does not specify any interface for use. The Message Transfer Part Layer 2 (MTP-2) is an OSI Data-Link Network Layer equivalent, providing for signal unit delimitation, alignment and error-detection and correction. It also monitors and reports up to MTP-3, error rates and congestion parameters. MTP Layer 3 (MTP-3) is a network-layer process allowing for functionalities for message discrimination and distribution as well as for traffic management, link management and routing. The Signalling Connection Control Part (SCCP) layer provides addressing mechanisms to route messages to databases and other network entities, acting as a layer-3 protocol for them. The Transaction Capabilities Application Protocol (TCAP) layer provides a way for end-users applications within network entities, in the SS7 network to access other
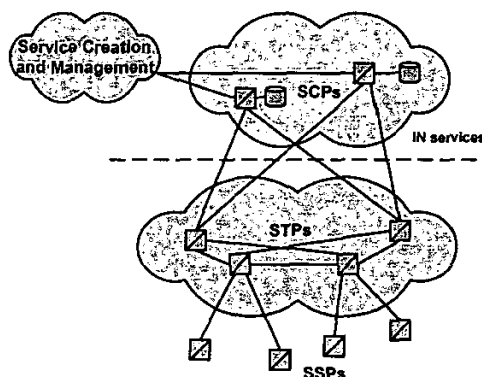

Figure 1 SS7-IN

end-users in a peer-to-peer way. The ISDN User Part (ISUP) is the ISDN-oriented augmentation of the Telephone User Part. It provides the functionalities for setting-up and tearing-down the circuits involved in telephone calls in the PSTN.

3. Convergence: the next step.

Over the past decade Internet has become an ubiquitous communication tool and IP-based applications are common. The total amount of packet based network traffic has surpassed traditional circuit-switched network traffic. Telecommunication equipment vendors have carried out many investment analyses on network integration, offering positive business cases. On the other side, incumbent operators remain rather conservative in their investments towards network integration due to shareholder value considerations as well as a preventive position against a wrong step that would let them in a weak situation comparing to competing operators: Operators' investigations of expected costs after migration scenarios showed only little savings in operational costs due to optimised organisational structures. The foreseen benefits are open standards, multivendor interoperability and the integration of voice and data network and its management as a whole.

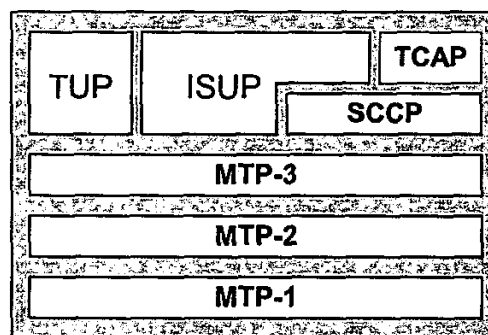The signalling within this new, packet based telephony

Figure 2 SS7 Stack

531

network can be studied from a "distributed switch" concept. As shown in figure 3 the functionality in traditional PSTN exchanges can be split into several independent modules [6]. The line module, deals with the detection-transmission of event-signals from/to the terminal side, the tones towards it and the collection of the dialled number. The call processing module analyses the collected number and asks the signalling module to start signalling accordingly. The trunk module deals with the circuit switched network that forms the media backbone. If this modular system is decomposed in its independent blocks and distributed within a packet network building a distributed switch as shown in figure 4, the basic architecture of the IP-Telephony-Signalling Network is established. These new elements perform the same functionality as the associated modular blocks in figure 3. As shown in figure 4, the interoperability between this packet-based telephony network with the legacy PSTN/IN systems should be a must in order to assure a proper return of investment (ROI) to telcos so that the transition towards a complete packet-based telephony network can be done in a smooth non radical way. This transparent operation between legacy and new (packet based) telephony systems forms what the authors understand as "converged network".

4. Enabling Protocols and Architectures.

In this section we try to, briefly, introduce the current standard protocols enabling the presented network connectivity as well as those still upon discussion.

4.1. ITU H.323

Initially targeted to multimedia conferences over LANs that do not provide guaranteed quality of service QoS, ITU-T H.323 has evolved towards the MAN and WAN environments [7]. A typical H.323 network is composed of a number of zones interconnected via a WAN. Each zone consists of a gatekeeper (GK), a number of terminal endpoints (TE), and a number of multipoint control units (MCU) interconnected by a LAN or MAN. The GK is an H.323 entity providing address translation and control access to the network for the rest of elements. The MCU is an endpoint providing the capabilities for multipoint conferences. H.323 is not a single protocol but an umbrella covering a wide span of them as shown in figure 5.
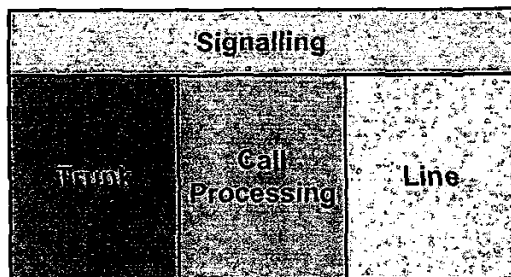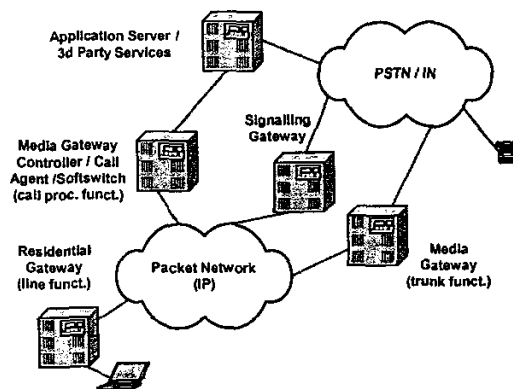

Figure 3 PSTN exchange functionality.

Figure 4 Distributed Switch Architecture

4.2. ITU H.248

The Gateway Control Protocol [8], H.248, defines the communication between elements of a physically decomposed gateway, i.e. a Gateway (MG for Media Gateways) and a Gateway Controller (MGC). This protocol was defined in collaboration with the IETF MEGACO working group.

4.3. IETF SIGTRAN's SCTP and ALs.

As a working group within the IETF Transport Area, the Signaling Transport (SIGTRAN) group addresses the transport of packet-based PSTN signaling over IP Networks, taking into account functional and performance requirements of the PSTN signaling.

SIGTRAN defines the Stream Control Transport Protocol (SCTP) [9] and adaptation layer specifications for the transport of PSTN signalling protocols over SCTP.

SCTP arises from the need to transport a variety of PSTN protocols over IP networks, solving the problems that existing transport protocols (TCP, UDP) face on doing so: while UDP provides a not tolerable unreliability for the transport of a kind of information such as telephony signalling, TCP presents a head of line blocking and security vulnerabilities that might affect the resilience-overall operation of a signalling network based on it. SCTP, as a transport alternative as shown in figure 6, provides unicast communication, session oriented, with reliable transmission mechanisms (error, sequence, flow and congestion controls), rate adaptive, message oriented, multi-streaming, multi-homing, with a continuous
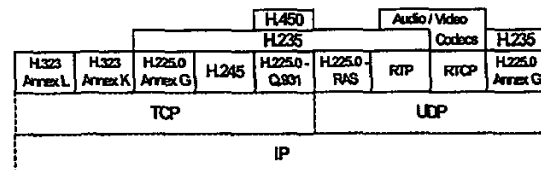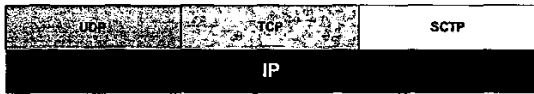

Figure 5 H.323 stack.

Figure 6. SCTP as a transport alternative.

monitoring of reachability and a graceful termination mechanism. Sigtran defines an adaptation layer, User Adaptation, for each of the SS7 protocols that could be transported over SCTP to support the services expected by the signalling protocols from its underlying layer: M2UA for MTP2, M3UA for MTP3, SUA for SCCP and so on [10] [11][12] [13].

4.4. IETF SIP

The Session Initiation Protocol [16] is an application-layer control signaling protocol for creating, modifying and terminating sessions with one or more participants. SIP has attracted a lot of attention because of its simplicity and ability to support rapid introduction of new services, mainly due to its addressing scheme, URL based, and the support of MIME type of data. There are two major architectural elements to SIP: the Server, and the User Agent (UA). There exist three different server types, a redirect server, a proxy server, and a registrar server. The UA resides at the SIP end station, and contains two components: a User Agent Client (UAC), which is responsible of issuing SIP requests, and a User Agent Server (UAS), which responds to such requests. The functional architecture of an SIP system is depicted in figure 7.

It is in the realm of SIP where most of the discussions on IN services in VoIP networks and interoperability with legacy SS7-IN is happening. In the following paragraphs we briefly comment them.

SIP-T

SIP for Telephones (SIP-T) focuses on how SIP should be used to provide ISUP transparency across PSTN-IP interconnections. This is achieved using both translation and encapsulation of ISUP messages into SIP messages. At SIP-ISUP gateways, SS7 ISUP messages are encapsulated within SIP. Intermediaries like proxy servers that make routing decisions for SIP requests cannot be expected to understand ISUP, so simultaneously, some critical information is translated from an ISUP message into the corresponding SIP headers in order to determine how the SIP request will be routed [17].

PINT & SPIRITS

The Services in the PSTN/IN Requesting Internet Services (spirits) Working Group of the IETF Transport Area addresses how services supported by IP network entities can be started from IN (Intelligent Network) requests, as well as the protocol arrangements through which PSTN can request actions to be carried out in the IP network in response to events (IN Triggers) occurring within the PSTN/IN. The SPIRITS Architecture [18] was born as a response to a previous work called PSTN/Internet
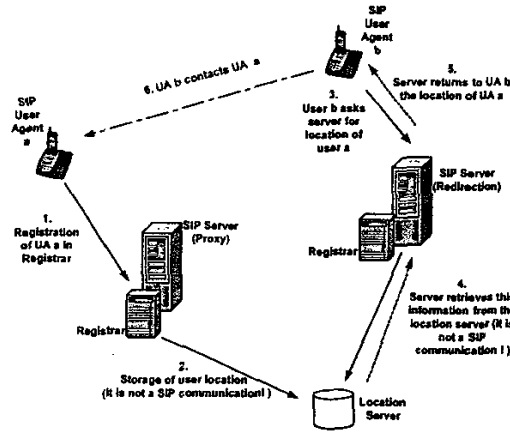

Figure 7 SIP functional architecture and example of operation

Internetworking (PINT) in support of the services initiated in the reverse direction - from the Internet to PSTN [19].

SIN

Some work has also been done recently [20] on interoperation between INAP and SIP, based on mapping the states of the IN Capability Set 2 Basic Call Model into SIP messages. The idea is depicted in figure 8, showing the easiness of interoperation with existing-legacy PSTN IN infrastructure and services.

5. The GEMINI Project.

GEMINI stands for "Generic Architecture for Customised IP-based IN services over Hybrid VoIP and SS7", an EU Information Societies Technology (IST) Programme started in the spring of 2002, with Telekom Austria AG, Alcatel Sel AG, Solinet GmbH, Intracom S.A. Otenet S.A. and the COM Center of the Technical University of Denmark as participants. The main objective of the project is to offer existing and next-generation customized and personalized IN services in an IP based environment to meet the demands of multi-party, multi-connection and multimedia calls. Additionally, GEMINI focuses on the Interworking between the IN-IP based and IN-SS7 based architectures in order to deliver value-added services in a converged environment to both PSTN and IP-based clients. To achieve these aims, GEMINI will base on a modular
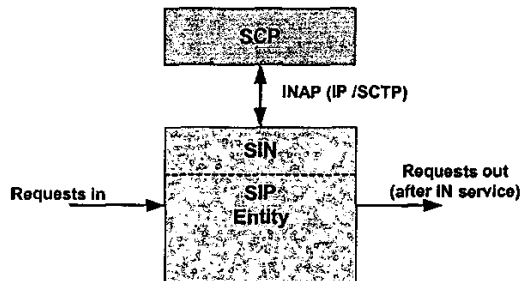

Figure 8. SIN model of service.

533

and scalable architecture, oversimplified in figure 9, with the following innovative features:

● GEMINI architecture bases on SIP, in belief it is the technological framework for future deployment of VoIP services-signalling for both fixed and mobile future telephony networks.

● VoIP endpoints (H.323 or SIP end-users) request access to IN services that are provided by the SS7 world.

● VoIP endpoints request access to value-added services that are handled by a central server within the IP domain, and may act either as a proxy or a redirect SIP server.

● VoIP endpoints request access to services that may be managed in different administrative domains.

● PSTN/ISDN endpoints gain access to hybrid IN and VoIP services.

The functionality of the services available on the PSTN/ISDN side and provided by the IP realm, will be a subset of the functionality used on the IP side due to evident reasons (i.e. bandwidth, terminal and network capabilities, etc). The determination of the service information to be carried out at each side (IP or PSTN) will be performed by the GEMINI architecture.

The end result will be value-added telephony services with a high degree of personalization, and extensions to the traditional IN services, expressed as hybrid, innovative solutions to cater for end users' needs involving PSTN and IP-based endpoints. It must be noted that the services that GEMINI will design and implement do not aim to take over telco services, but merely to augment them. Furthermore, GEMINI does not aim at introducing new communication protocols. Instead, current state-of-the-art approaches will be used and, if needed, extended [21].

Currently closing the specification stage of the project, the trials and demonstrations are expected to be carried in the autum of 2003, after the development stage, to conclude the project in the spring of 2004.

Updated information and contacts to the project partnerts may be found in the world wide web link:
http://gemini.otenet.gr/

6. Conclusion.

A brief description of the evolution of signalling has been given as well as a somere description of some of the enabler protocols and architectures for telephony signalling in IP and IP-SS7 converged networks. Finally the GEMINI project has been introduced and its main aims and innovative architectural features presented.
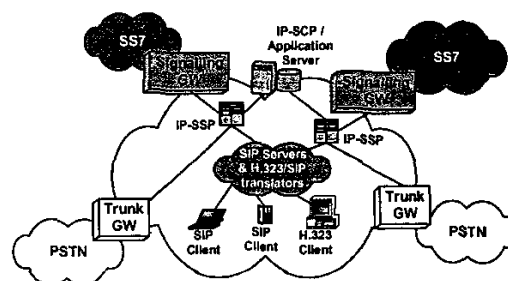

Figure 9. GEMINI Architecture.

7. References.

[1] Ericsson, Telia Studentlitteratur. "Understanding Telecommunications", 1998. ISBN: 91-44-00212-2

[2] Uyless Black, "The Intelligent Network. Customizing Telecommunication Networks and Services", 1998. ISBN 0-13-793019-4

[3] Travis Russell, "Signaling ystem #7", 2000. ISBN: 0-07-136119-7

[4] ITU-T Rec 1201, " Principles of Intelligent Network Architecture", October 1992.

[5] ITU-T Rec. Q.700, "Introduction to CCITT Signaling System No.7.", March 1993

[6] Modarressi & Mohan, "Control and Management in Next Generation Networks: Challenges and Oportunities.", IEEE Communications Magazine, October 2000.

[7] ITU-T Rec H.323, "Packet-based multimedia communication systems", October 2000.

[8] ITU-T Rec. H.248, "Gateway Control Protocol", June 2000.

[9] IETF RFC 2960, "Stream Control Transmission Protocol", October 2000.

[10] IETF RFC 3057, "ISDN Q.921-User Adaptation Layer", February 2001.

[11] IETF draft-ietf-sigtran-sua-10, "SS7 SCCP-User Adaptation Layer (SUA)", June 2002.

[12] IETF draft-ietf-sigtran-m3ua-11, "SS7 MTP3 User Adaptation Layer (M3UA)", January 2002.

[13] IETF draft-ietf-sigtran-m2ua-12, "Signaling System 7 Message Transfer Part 2 User Adpatation Layer", December 2001.

[14] IETF draft-ietf-iptel-trip-09, "Telephony Routing over IP (TRIP)". August 2001.

[15] IETF draft-ietf-enum-rfc2916bis-00. "The E.164 to URI DDDS Application". February 2002.

[16] IETF RFC 3261. "SIP: Session Initiation Protocol". June 2002.

[17] IETF draft-ietf-sip-isup-03 "ISUP to SIP mapping.". August 2001.

[18] IETF RFC 3136. "The SPIRITS Architecture". June 2001.

[19] IETF RFC 2848 "The PINT Service Protocol: extensions to SIP and SDP for IP Access to telephone call services". June 2000.

[20] IETF draft-gurbani-sin-02 "Interworking SIP and IN Applications". December 2002.

[21] GEMINI Consortium. IST-2001-33464-D-2.1."Requirements Specification and GEMINI Architecture".July 2002.