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Services and Protocols for Advanced Networks (SPAN) - Intelligent Networks (IN) - Architectures and signalling requirements for IN-based networks interworking with IP-based networks

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Foreword

This ETSI Guide (EG) has been produced by ETSI Technical Committee Services and Protocols for Advanced Networks (SPAN).

Introduction

The present document is closely aligned with ITU-T Recommendation Q.1244 [19], such that clauses 5 and 7 are similar in technical content with the related parts of the ITU-T Recommendation. Clause 6 whilst aligned with ITU-T Recommendation Q.1244 [19] provides more detail in the figures and tables describing the relationship of the reference scenarios with the lower layer transport protocols. The intention of the present document is to define a set of enhancements for IN CS-3 for interworking with IP-networks, which comprises IN CS-4 in the ITU-T. In ETSI these enhancements will be considered as a revision of ETSI Core INAP.

1 Scope

The present document describes the standardization of functions to allow interworking between Intelligent Networks and IP-networks for IN CS-3. These functions include:

- Signalling Requirements for interworking between functional entities in the IN and IP-networks;
- Signalling Requirements to support benchmark capabilities between functional entities in the IN and IP-networks;
- Architecture supporting the transport of higher layer session multimedia protocols between IP-network and the Circuit Switched Network;
- Interworking and addressing of service control functions and service control gateway functionality across IN and IP-network boundaries;
- Study of the security aspects.

The present document only considers scenarios for interworking IN CS-3 capabilities and IP-based networks. Service and network integration is outside the scope of the present document.

The management functional entity requirements and interfaces are outside the scope of the present document.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

[1] Void.

[2] Void.

[3] Void.

[4] ITU-T Recommendation Q.1224: "Distributed functional plane for intelligent network Capability Set 2".

[5] ITU-T Recommendation Q.1231: "Introduction to Intelligent Network Capability Set 3".

[6] ITU-T Recommendation H.225.0: "Call signalling protocols and media stream packetization for packet-based multimedia communication systems".

[7] ITU-T Recommendation H.245: "Control protocol for multimedia communication".

[8] ITU-T Recommendation H.246: "Interworking of H-Series multimedia terminals with H-Series multimedia terminals and voice/voiceband terminals on GSTN and ISDN".

[9] ITU-T Recommendation H.248: "Gateway control protocol".

[10] ITU-T Recommendation H.323: "Packet-based multimedia communications systems".

[11] IETF RFC 2543 (1999): "SIP: Session Initiation Protocol".

[12] IETF RFC 2458 (1998): "Toward the PSTN/Internet Inter-Networking - Pre-PINT Implementations".

[13] IETF RFC 2848 (2000): "The Pint Service Protocol: Extensions to SIP and SDP for IP Access to Telephone Call Services".

[14] IETF RFC 3136 (2001): "The SPIRITS Architecture".

[15] ETSI TR 123 821: "Technical Specification Group Services and System Aspects; Architecture Principles for Release 2000".

[16] IETF RFC 791 (1981): "Internet Protocol".

[17] IETF RFC 3015 (2000): "Megaco Protocol Version 1.0".

[18] ETSI ES 201 915 (V1.1.1) Parts 1 to 12: "Open Service Access; Application Programming Interface".

[19] ITU-T Recommendation Q.1244: "Distributed functional plane for Intelligent Network Capability Set 4".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

call: point-to-point multimedia communication between two H.323 endpoints

NOTE: The call begins with the call set-up procedure and ends with the call termination procedure. The call consists of the collection of reliable and unreliable channels between the endpoints. A call may be directly between two endpoints, or may include other H.323 entities such as a Gatekeeper or MG. In case of interworking with some CSN endpoints via a Gateway, all the channels terminate at the Gateway where they are converted to the appropriate representation for the CSN end system. Typically, a call is between two users for the purpose of communication, but may include signalling-only calls. An endpoint may be capable of supporting multiple simultaneous calls.

call signalling channel: reliable channel used to convey the call set-up and teardown messages (see ITU-T Recommendation H.225.0) between two H.323 entities

composite gateway: logical entity composed of a single MGC and one or more MGs that may be reside on different machines

NOTE: Together, they preserve the behaviour of a gateway as defined in ITU-T Recommendations H.323 [10] and H.246 [8].

GateKeeper (GK): H.323 entity on the network that provides address translation and controls access to the network for H.323 terminals, Gateways and MCUs

NOTE: The Gatekeeper may also provide other services to the terminals, Gateways and MCUs such as bandwidth management and locating Gateways.

GateWay (GW): an H.323 GateWay (GW) is an endpoint on the network which provides for real-time, two-way communications between H.323 Terminals on the packet based network and other ITU Terminals on a switched circuit network, or to another H.323 Gateway

NOTE: Other ITU-T terminals include those complying with Recommendations H.310 (H.320 on B-ISDN), H.320 (ISDN), H.321 (ATM), H.322 (GQOS-LAN), H.324 (GSTN), H.324M (Mobile), and V.70 (DSVD).

H.248: describes a control model and protocol for an MGC to control an MG

NOTE: An MGC-MG association reserves the behaviour of a H.323 gateway. H.248 has been developed in ITU-T SG16, in co-operation with IETF MEGACO, with the intention of providing a single, international standard for Media Gateway Control.

H.323 Entity: any H.323 component, including terminals, Gateways, Gatekeepers, MGCs and MGs

H.323 Service Control Protocol: specifies protocol for multimedia communications over packet networks to be used between gatekeepers

NOTE: This is work under study in SG16, intended to enhance service related information transfer to and from the gatekeeper, which is currently limited to RAS. This work is expected to be strongly influenced by the IN-IPT interworking model and the joint work of SG16/SG11 in general.

IP-address: 32-bit address defined by the Internet Protocol in IETF RFC 791

NOTE: It is usually represented in dotted decimal notation.

IP-network: general term denoting networks based on the Internet Protocol (IP) suite

NOTE: A network which uses IP as the Layer 3 protocol.

JAVA: software platform trademark of Sun Microsystems

Media Gateway (MG): converts media provided in one type of network to the format required in another type of network

NOTE: For example, an MG could terminate bearer channels from a switched circuit network (i.e. DSOs) and media streams from a packet network (e.g., RTP streams in an IP-network). This gateway may be capable of processing audio, video and T.120 alone or in any combination, and will be capable of full duplex media translations. The MG may also play audio/video messages and perform other IVR functions, or may perform media conferencing.

Media Gateway Controller (MGC): controls the parts of the call state that pertain to connection control for media channels in a MG

proxy, proxy server: intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients

NOTE: Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets, and, if necessary, rewrites a request message before forwarding it, see [11]. This functional element is functionally similar to the user agent server. In essence the proxy server is comprised of both a SIP client and a SIP server.

RAS (Reservation, Admission and Status): the RAS signalling function uses H.225.0 messages to perform registration, admissions, bandwidth changes, status, and disengage procedures between endpoints and Gatekeepers

NOTE: For details refer to ITU-T Recommendations H.323 [10] and H.225.0 [6].

RAS channel: reliable channel used to convey the registration, admissions, bandwidth change, and status messages (following ITU-T Recommendation H.225.0) between two H.323 entities

redirect server: server that accepts a SIP request, maps the address into zero or more new addresses and returns these addresses to the client

NOTE: Unlike a proxy server, it does not initiate its own SIP request. Unlike a user agent server, it does not accept calls, see [11]. The redirect server is not responsible for call control but will simply respond to SIP requests with a new address.

registrar server: simply responds to registrar requests

NOTE: Typically this is co-located with either the proxy or the redirect server, and may be adapted to perform location-based services, see [11].

server: application program that accepts requests in order to service requests and sends back responses to those requests

NOTE: Servers are either proxy, redirect or user agent servers or registrars [11].

Session Initiation Protocol (SIP): text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users

NOTE: Such sessions include voice, video, chat, interactive games, and virtual reality. The IETF SIP working group is chartered to continue the development of SIP.

terminal: an H.323 Terminal is an endpoint on the network which provides for real-time, two-way communications with another H.323 terminal, Gateway, or Multipoint Control Unit

NOTE: This communication consists of control, indications, audio, moving colour video pictures, and/or data between the two terminals. A terminal may provide speech only, speech and data, speech and video, or speech, data and video.

User Agent (UA): application which contains both a user agent client and user agent server

User Agent Client (UAC), calling user agent: the user agent client is the functional entity that may initiate a SIP request

User Agent Server (UAS) called user agent: a user agent server is a server application that contacts the user when a SIP request is received and that returns a response on behalf of the user

NOTE: The response accepts, rejects or redirects the request. The user agent server is the functional entity that may initiate a SIP response.

Zone: collection of all Terminals (Tx), GateWays (GW), and Multipoint Control Units (MCU) managed by a single GateKeeper (GK)

NOTE: A Zone includes at least one terminal, and may or may not include Gateways or MCUs. A Zone has one and only one Gatekeeper. A Zone may be independent of network topology and may be comprised of multiple network segments that are connected using routes (R) or other devices.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

API	Application Programming Interface
ASP	Application Service Provider
BCF	Bearer Control Function
BCSM	Basic Call State Model
BICC	Bearer Independent Call Control
C/B GF	Call/Bearer Gateway Function
CCF	Call Control Function
CM	Call Manager
CORBA	Common Object Request Broker Architecture
CSN	Circuit Switched Network
CTD	Click-To-Dial
CTF	Click-To-Fax
CTFB	Click-To-Fax Back
DA-GF	Dial Access Gateway Function
DFP	Distributed Functional Plane
DN	Directory Number
DRC	Direct Routed Call
DSS1	Digital Subscriber Signalling system No.1
GCI	Global Connection Identifier
GF	Gateway Function
GK	GateKeeper
GRC	Gatekeeper Routed Call
GT	Global Title
GW	GateWay
HTTP	Hyper Text Transfer Protocol
IAP	Internet Access Provider

IDP	Initial DP
IETF	Internet Engineering Task Force
IN	Intelligent Network
IP	Internet Protocol
IPT	IP Telephony
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ISUP	ISDN User Part
ITU-T	International Telecommunications Union
JAIN	Java APIs for Integrated Networks
MCU	Multipoint Control Unit
MEGACO	MEdia GAteway COntrol
MG	Media Gateway
MGC	Media Gateway Controller
MIB	Managed Information Base
MM-GF	Media Manager Gateway Function
MRF	Media Resource Function
MSISDN	Mobile Subscriber ISDN number
MTP	Message Transfer Part
NAI	Network Access Identifier
OAM	Operation and Maintenance
O-BCSM	Originating BCSM
PC	Personal Computer
PDU	Protocol Data Unit
PINT	PSTN Internet Interworking
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RAS	Registration Admission and Subscription
RM	Resource Manager
RTCP	Real-Time Control Protocol
RTP	Real-Time Protocol
SA-GF	Service Application Gateway Function
SCCP	Signalling Connection Control Part
SCF	Service Control Function
SC-GF	Service Control Gateway Function
SCTP	Simple Control Transmission Protocol
SDF	Service Data Function
SDP	Session Description Protocol
S-GF	Signalling Gateway Function
SIP	Session Initiation Protocol
SM	Session Manager
SMF	Service Management Function
SMTP	Simple Mail Transfer Protocol
SPC	Signalling Point Code
SPIRITS	Service in the PSTN/IN Requesting InTernet Service
SRF	Specialized Resource Function
SSCOP	Service Specific Connection-Oriented Protocol
SSF	Service Switching Function
T-BCSM	Terminating BCSM
TDP	Trigger Detection Point
TIPHON	Telecommunication and Internet Protocol Harmonization Over Networks
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
URL	Universal Resource Locator
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network

4 Services

It is intended to support IN CS-3 benchmark services, Internet based service customization and Voice over IP.

4.1 Benchmark services

List of benchmark services:

- Internet Call Waiting (ICW) Service;
- IN-based NAI/URL Address translation and resolution services;
- IN-based service for Dial-up Internet Access;
- Click-to-Dial (CTD) Service;
- Click-to-Fax (CTF) Service;
- IN-IP Telephony Interworking;
- IN-IP personal mobility service.

4.2 IN-Internet service examples

- Request-to-Call-Back CSN
  A user is able to initiate a telephone call by clicking a button during a Web session. The call can be first set up in the direction of the requester of the call, or first be set up in the direction of the party the requester wants to be connected to. E.164 addressing for both A-party and B-party is assumed, and both parties are assumed to be connected to the Circuit Switched Network. Possible reasons for failures are A-party busy, A-party no answer, B-party busy, B-party no answer. No detailed notifications are reported back to the requester. An example of an application of this feature would be on-line shopping: A user is browsing through an on-line catalogue, and clicks a button thus inviting a call from a sales representative. In the IN the request could be handled depending on availability of agent, time of day, etc.

- Request-to-Call-CSN
  A user is able to initiate a telephone call by clicking a button during a Web session. The requested call is to be set up between two parties identified by E.164 addresses, which are connected to the Circuit Switched Network. The requester him/herself may or may not take part in the call to be set up. Possible reasons for failures are A-party busy, A-party no answer, B-party busy, B-party no answer. No detailed notifications are reported back to the requester.

- Request-to-Call-Back IP
  A user is able to initiate a telephone call by clicking a button during a Web session. The call can be first set up in the direction of the requester of the call, or first be set up in the direction of the party the requester wants to be connected to. E.164 addressing for both A-party and B-party is assumed, and one or both parties have a VoIP service. A VoIP user might be a mobile user as well. Possible reasons for failures are A-party busy, A-party no answer, B-party busy, B-party no answer. No detailed notifications are reported back to the requester. An example of an application of this feature would be on-line shopping: A user is browsing through an on-line catalogue, and clicks a button thus inviting a call from a sales representative. In the IN the request could be handled depending on availability of agent, time-of-day etc.

- Request-to-Call IP
  A user is able to initiate a telephone call by clicking a button during a Web session. The requested call is to be set up between two parties identified by E.164 addresses, where one or both parties have a VoIP service. A VoIP user might be a mobile user as well. The requester him/herself may or may not take part in the call to be set up. Possible reasons for failures are A-party busy, A-party no answer, B-party busy, B-party no answer. No detailed notifications are reported back to the requester.

- End User Service Data Customization via an IP-network
  A service end-user can customize his/her service data, service profile via an IP-network.

NOTE: This feature has been implemented for IN CS-3 for access through PSTN. See SMF - SMF [5].

- Request-to-Fax
  A user of an IP-network who does not have access to a fax machine is able to send a fax to a receiver who has access to a fax machine but not to an IP-network, during a Web session. The receiver is identified by an E.164 number. Possible reasons for failure are receiver busy, receiver no answer. No detailed notifications are reported back to the requester. An example of an application of this feature would be a hotel reservation form on the website of a travel agent, where the user fills out the form and then clicks a button to request the form to be sent as a fax to the hotel being reserved.

- Request-to-Fax-Back
  An Internet user who has access to a fax machine is able to request (and subsequently receive) to have certain electronically stored information sent by fax during a Web session. The requester is identified by an E.164 number. Possible reasons for failure are receiver busy, receiver no answer. No detailed notifications are reported back to the requester.

- Request-to-Hear-Content-from-IP
  A user shall have the possibility to have access to Web content by telephone. The user can have a subset of a Web page content delivered in audio form via telephone. The requesting user has access to an IP-network, and can invoke this service via a Web session. The receiving user is identified by an E.164 number. Possible reasons for failure are receiver busy, receiver no answer. No detailed notifications are reported back to the requester.

- Request-to-Hear-Content-from-CSN
  A user shall have the possibility to have access to Web content by telephone. The user can have a subset of a Web page content delivered in audio form via telephone. The requesting user has access to the Circuit Switched Network (e.g. PSTN/ISDN), and can invoke this service via the PSTN/ISDN. The receiving user is identified by an E.164 number. Possible reasons for failure are receiver busy, receiver no answer. No detailed notifications are reported back to the requester.

- Internet Call Waiting
  A user is notified of incoming calls during a Web session and, by clicking a button, is able to instruct the network on how this cal shall be further processed (e.g. rejected, forwarded to a voice mail system, accepted with or without interruption of the Web session (in case of acceptance without Web session interruption VoIP is assumed)). A sub-set of this feature would be just to keep a log of the times a user receives calls during an Internet session.

- Web Controlled PSTN/IP Conferencing Service
  Basic Web controlled PSTN/IP conference call, initiation of conference call, adding parties.

- IP Gateway Selection
  A service involving a CSN connection to a gateway to an IP-network is provided, making use of a network setup with several gateways towards the IP domain. An IN service is used to decide on which physical gateway to use, based among other things on the availability of the gateway, or on its load.
  This scenario is applicable for so-called Internet Access Servers (IAS) as well as Voice over IP (VoIP) gateways.

- Call Logging Service
  The Call Logging Service gives the ability to the Network Service Provider to provide the Internet user with all incoming related information (e.g., Calling Party Identification, Time Called, etc.) while the user is busy, i.e. logged on to a Web session, etc. The Network Service Provider offering such a service (i.e., Call Logging Service) also provides the capability to the user to retrieve such call related information at a later time. The Call Logging Service is associated with the Internet Call Waiting service offered by the Network Service Provider and the user must subscribe to these features to avail the benefits of the Call Logging Service. Additionally, a Network Service Provider may provide the capability to the user to present all incoming call related information to user (e.g., an IP client).

- NAI/URL Address Translation Service
  The following are service features and requirements related to address translation:

  - Registration of previously registered IP-addresses of the communicating end systems within IN infrastructure.

  - Registration of mnemonic addresses (e.g., Names) of the communicating end systems infrastructure.

- Optionally, it should be possible to disseminate the registered information to where it is needed, and collect registered information from other service providers vital for address translation on a global basis.
- E.164 to IP-Address translation result:
    - For the user with dial-up access to Internet through telephone line, the IP-address is dynamically allocated. The translation result is provided from Internet to IN;
    - For the Intranet user, the IP-address is statically allocated and the associated E.164 number is a pre-assigned feature number. The translation result is provided from IN to Internet;
- It should be possible for the network to support the following as part of address translation:
    - Time-of-day translation;
    - 1-to-N address translation;
    - N-to-1 address translation;
- It shall be possible for the network to allow terminals to register the following:
    - Terminal characteristics (e.g., Video/Audio Coder characteristics);
    - QoS related parameters;
    - Different levels of security; and
    - Authentication.

- PINT benchmark services: click-to-dial, click-to-fax-back, click-to-fax, voice-to-access-content.
- Extended PINT services: PC-to-phone case for click-to-dial, phone initiated voice-to-access-to-content.
- Internet telephony (iPtel): phone-to-PC, PC-to-phone, PC-to-PC (dial-up access).
- Internet user incoming call screening.
- Service data customization through Internet.
- IN-based service for dial-up Internet Access.
- Conferencing Services.
- Multimedia control services.

5 Functional architecture

5.1 Introduction

The functional model proposed is an extension of the IN CS-2 functional model (see figure 5.1). It is intended to support IN CS-3 benchmark services, Internet based service customization and termination of Voice over IP to reach users in the telephone domain as well as general IN management capabilities.

The clause describes the functional entities required to support the IN CS-3 benchmark features, which include:

- New functional entity requirements;
- Extensions to existing functional entities as required;
- Lower layer protocol gateway and mapping function requirements.