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Technical Report

Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); SDP Interworking between Call/Session Control Protocols (SIP/SDP, RTSP/SDP; etc.) and the Gateway Control Protocol (H.248/SDP)

Reference

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

1 Scope

The present document specifically describes the differing SDP usage between SIP [i.2] and H.248 [i.3] together with the implied mapping capability that is performed by the MGC/Call Server.

SDP [i.1] has been widely selected as the protocol of choice within VoIP (or multimedia; MMoIP) to describe the media requirements of a given session/call/connection. However, the different VoIP control protocols that utilise SDP each specify differing requirements in their use of SDP. There is therefore a need for a MGC/Call Server to arbitrate between these variations in the use of SDP and perform the interworking between them.

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Any network element (e.g. a MGCF) which handles both H.248/SDP signalling and SIP/SDP signalling provides any necessary interworking between both signalling protocols (see figure 1). Such interworking comprises in general:

- interworking between SIP and H.248 signalling on message and procedural level (out of scope of the present document); and
- interworking between the two SDP segments (SDP-SDP interworking; the scope of the present document).

The function providing SDP-to-SDP interworking between SIP/SDP and H.248/SDP signalling is, in the present document, termed a "SDP Mapper" (see also clause 3.1).

The SDP Mapper performs SDP-SDP interworking capability to reconcile the different uses of SDP between control protocols H.248 and SIP. In order to perform this role, the SDP Mapper takes into account i) which parts of SDP are required to be sent on an interface, and ii) which parts of SDP are received on an interface. For a given session/call, which use the two different control protocols at each end, some SDP parameters may be transited whilst others may not. The SDP Mapper ensures that the differing requirements with regard to SDP handling at each end are mutually satisfied.

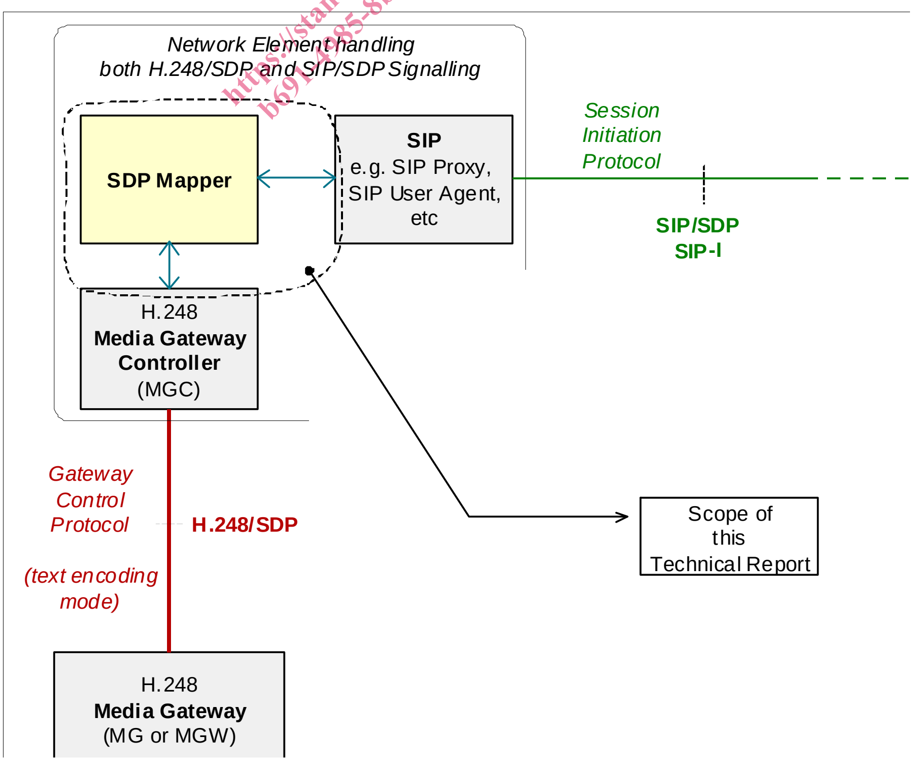

Figure 1: Scope

1.1 Applicability

This paper is applicable to any MGC/Call Server that exhibits both a SIP and H.248 interface. The former includes interfaces to both User Equipments (i.e. SIP User Agents) and peer SIP proxies (like Call Servers). The latter includes interfaces to any H.248-controlled MGW (e.g. RMGW, AMGW, TMGW, BMGW, etc.).

2 References

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.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
  - if it is accepted that it will be possible to use all future changes of the referenced document for the purposes of the referring document;
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NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

2.1 Normative references

Not applicable.

2.2 Informative references

[i.1] IETF RFC 4566 (2006): "SDP: Session Description Protocol".

[i.2] IETF RFC 3261 (2002): "Session Initiation Protocol".

[i.3] ITU-T Recommendation H.248.1 (2005): "Gateway control protocol: Version 3".

[i.4] IETF RFC 3264 (2002): "An Offer/Answer Model with Session Description Protocol (SDP)".

[i.5] IETF RFC 3262 (2002): "Reliability of Provisional Responses in Session Initiation Protocol (SIP)".

[i.6] IETF RFC 4317 (2005): "Session Description Protocol (SDP) Offer/Answer Examples".

[i.7] IETF RFC 2327 (1998): "SDP: Session Description Protocol".

[i.8] ITU-T Recommendation Q.1912.5 (2004): "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".

[i.9] ITU-T Recommendation Q. Supplement 45 (09/2003): Technical Report TRQ.2815: "Requirements for interworking BICC/ISUP network with originating/destination networks based on Session Initiation Protocol and Session Description Protocol".

[i.10] ITU-T Recommendation T.38 (2005) "Procedures for real-time Group 3 facsimile communication over IP networks".

[i.11] ITU-T Recommendation V.152 (2005): "Procedures for supporting voice-band data over IP networks".

[i.12] ITU-T Recommendation H.248.39 (2006): "Gateway control protocol: H.248 SDP parameter identification and wildcarding".

[i.13] ITU-T Recommendation H.248.49 (2007): "Gateway control protocol: Session description protocol RFC and capabilities packages".

[i.14] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

[i.15] IETF RFC 3951: "Internet Low Bit Rate Codec (iLBC)".

[i.16] IETF RFC 3952: "Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech".

[i.17] ETSI ES 283 002: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); H.248 Profile for controlling Access and Residential Gateways".

[i.18] ETSI ES 283 024: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); H.248 Profile for controlling Trunking Media Gateways; Protocol specification".

[i.19] ETSI EN 383 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".

[i.20] ETSI TR 183 014: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN Emulation; Development and Verification of PSTN/ISDN Emulation".

[i.21] IETF RFC 3108: "Conventions for the use of the Session Description Protocol (SDP) for ATM Bearer Connections".

[i.22] IETF RFC 4733: "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals".

[i.23] IETF RFC 2543: "SIP: Session Initiation Protocol".

[i.24] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

[i.25] IETF RFC 3551: "RTP Profile for Audio and Video Conferences with Minimal Control".

[i.26] ITU-T Delayed Contribution COM16-D410-E (01/2004), "Proposal to begin work on H.248.1 version 3", (Clause 2.1.1 "SDP compatibility between H.248 and other SDP-based protocols").

[i.27] IETF RFC 3330: "Special-Use IPv4 Addresses".

[i.28] IETF RFC 5156: "Special-Use IPv6 Addresses".

[i.29] IETF draft-ietf-mmusic-sdp-capability-negotiation: "SDP Capability Negotiation".

[i.30] IETF draft-ietf-mmusic-sdp-media-capabilities: "SDP Media Capability Negotiation".

[i.31] 3GPP TS 29.802: "(G)MSC-S - (G)MSC-S Nc Interface based on the SIP-I protocol".

[i.32] IETF RFC 4291: "IP Version 6 Addressing Architecture".

[i.33] IETF RFC 4293: "Management Information Base for the Internet Protocol (IP)".

[i.34] IETF RFC 3849: "IPv6 Address Prefix Reserved for Documentation".

[i.35] IETF RFC 3056: "Connection of IPv6 Domains via IPv4 Clouds".

[i.36] IETF RFC 4380: "Teredo: Tunneling IPv6 over UDP through Network Address Translations (NATs)".

[i.37] IETF RFC 1897: "IPv6 Testing Address Allocation".

[i.38] IETF RFC 3701: "6bone (IPv6 Testing Address Allocation) Phaseout".

[i.39] IETF RFC 4843: "An IPv6 Prefix for Overlay Routable Cryptographic Hash Identifiers (ORCHID)".

[i.40] IETF RFC 4773: "Administration of the IANA Special Purpose IPv6 Address Block".

[i.41] IETF RFC 3232: "Assigned Numbers: RFC 1700 is Replaced by an On-line Database".

[i.42] IETF RFC 1918: "Address Allocation for Private Internets".

[i.43] IETF RFC 1797: "Class A Subnet Experiment".

[i.44] IETF RFC 3068: "An Anycast Prefix for 6to4 Relay Routers".

[i.45] IETF RFC 3171: "IANA Guidelines for IPv4 Multicast Address Assignments".

3 Definitions and abbreviations

3.1 Definitions

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

SDP Mapper: function for SDP-to-SDP interworking between two different, SDP-using signalling protocols

NOTE: One of these signalling protocols is the Gateway Control Protocol according H.248 in text-encoding mode. The other signalling protocol is SIP in the scope of the present document.

SIP-I: use of SIP with a message body that encapsulates ISUP information

NOTE: Definition according to ITU-T Recommendation Q.1912.5 [i.8] and clause 4.8 in ITU-T Supplement 45 to Q-series Recommendations (TRQ.2815) [i.9].

3.2 Abbreviations

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

ALN	Analog Line
AMGW	Access Media GateWay
B2BIH	Back-to-Back IP Host (mode)
BCF	Bearer Control Function
BGF	Border Gateway Function
BMGW	Border Media GateWay
DNS	Domain Name System
GCP	Gateway Control Protocol
GW	GateWay
IP	Internet Protocol
IPR	IP router (mode)
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
LCD	Local Control Descriptor
LD	Local Descriptor (H.248)
MG, MGW	Media GateWay
MGC	Media Gateway Controller
MGCF	MGC Function
MIME	Multipurpose Internet Mail Extensions
MMoIP	MultiMedia-over-IP
PCMA	Pulse Code Modulation A-law
PSTN	Public Switched Telephone Network
RD	Remote Descriptor (H.248)

RFC	Request For Comments (IETF)
RMGW	Residential Media GateWay
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP-I	SIP with the MIME encoding of ISUP
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TGW	Trunking GateWay
TMGW	Trunking Media GateWay
TMR	Transmission Medium Requiremenrt
UA	User Agent
UDP	User Datagram Protocol
USI	User Service Information
VoIP	Voice-over-IP

4 Differences between SIP/SDP and H.248/SDP Usage

Clause 4.1 describes the SDP usage in SIP. Clause 4.2 describes the SDP usage in H.248. Clause 4.3 summarizes the differences between them.

4.1 SIP usage of SDP

SIP uses SDP for describing multimedia sessions RFC 3261 [i.2].

In terms of bearer control and usage of SDP, SIP has defined a basic Offer/Answer model that is documented in RFC 3264 [i.4] and illustrated in RFC 4317 [i.6]. The Offer will contain zero or more media streams. The basic Offer/Answer model is extended by an enhanced Offer/Answer model according IETF drafts [i.29] and [i.30].

The "offer-answer" mechanism mandates that when a block SDP is sent in one direction ("the Offer"), a corresponding block of SDP should be returned to the originator ("the Answer"). It is not possible to make a new "Offer" until an "Answer" is received. However, within a given session, there is no limit to the number of Offer/Answer exchanges that may occur (i.e. mid-session bearer change).

SIP does not permit the SDP block to contain more than one session description, although multiple media streams may be contained in each session description (with the implication that all streams are required simultaneously), and multiple codecs may be contained within each media stream (with the implication that one of the codecs is selected for use).

When SDP is sent in SIP, the following SDP lines are mandatory:

- **Protocol Version** line:
  Always set to "v=0".

NOTE: This value is recommended by RFCs on SDP, i.e. the "v=" line is not used for discrimination between the "SDP versions" as defined by RFC 4566 [i.1] and its predecessor RFC 2327 [i.7]. Both RFCs defining version 0 of the SDP.

- **Session Name** line:
  This can be defaulted to "s=" or else hold a string as defined in RFC 4566 [i.1].

- **Timing** line:
  This can be defaulted to "t=0 0".

- **Origin** line:
  This will be set to "o=<user name> <session id> <session version number> IN IP4 (or IP6) <IP4 address> (or <IP6 address>)".
  The session number can be zero and the session version initialized to zero. The IP4 (or IP6) address can be the

same as that appearing on the Connection Line.
The <user name> can default to "-".

- **Connection Data** line:
  Holds the network type, address type and connection address. Set to "c=IN IP4 <IP4 address>" or "c=IN IP6 <IP6 address>".

- If there is at least one media stream, the following line is also mandatory:

- **Media Description** line:
  Holds the media type, port number and the "codec types" (defined by transport protocol "proto" and media format "fmt" fields).

4.1.1 Basic O/A Model (RFC 3264 [i.4]): Initial Offer/Answer Exchange

SIP permits the initial Offer/Answer exchange within a SIP session to be realized via a number of SIP message combinations, dependent on when the necessary SDP information becomes available to be passed across the SIP interface. This is illustrated in table 1.

Table 1: Offer/Answer scenarios in SIP

SDP OFFER in:	SDP ANSWER in:	Comments / Additional Information
INVITE	180/183 and 200 OK	The ANSWER is repeated in the 200 OK if 100rel not being used.
INVITE	200 OK	Late terminating SDP.
180 / 183	PRACK	This is late originating SDP. RFC 3262 [i.5] mandates that the ANSWER to a 18X OFFER will be included in the PRACK.
200 OK	ACK	Late SDP at both originating and terminating ends.

RFC 3264 [i.4] mandates that the same SDP Timing (t=) line will appear in both SDP blocks (i.e. the Offer and corresponding Answer) and that there will be identical numbers of Media Description (m=) lines in both SDP blocks (the Offer and corresponding Answer). The implication of the latter is there will be a mechanism by which a given media line can be rejected/disabled. This is achieved by one or more of the following techniques:

- via the use of the Media Attribute line "a=inactive" to indicate that the related SDP is not sending/receiving;

- via the use of a null IP address of 0.0.0.0 (see notes 1 and 2; see also Annex A concerning a different semantic in SIP/SIP-I) in the Connection Data (c=) line;

NOTE 1: The initial specification for SIP version SIP/2.0 defined that placing **media on hold** was accomplished by setting the *connection address* to **0.0.0.0** (see RFC 2543 [i.23], paragraph B.5). Its usage for putting a call or media on hold is **no longer recommended** for SIP/2.0 (see RFC 3261 [i.2]), since it does not allow for RTCP to be used with held streams, does not work with IPv6, and breaks with connection-oriented media (see RFC 3264 [i.4], paragraph 8.4).
But there is one **applicability statement** in the context of Offer/Answer procedures (see RFC 3264 [i.4]).

However, it can be useful in an **initial Offer** when the offerer knows it wants to use a particular set of media streams and formats, but **does not know the addresses and ports** at the time of the Offer.
Of course, when used, the **port number** is NOT zero, which would specify that the stream has been **disabled** (see note 3). An SIP user agent will be capable of receiving SDP with a connection address of 0.0.0.0, in which case it means that **neither RTP nor RTCP** should be sent to the peer.

NOTE 2: IPv6 is different. There is no specification for the correspondent usage of the IPv6 connection address value 0:0:0:0:0:0:0:0 (or the abbreviated form).

- via the use of a null (zero) port number the Media Description (m=) line (see note 3).