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Technical Report

Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); SDP Interworking between SIP/SDP and 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 [2] and H.248 [3] together with the implied mapping capability that is performed by the MGC/Call Server.

SDP [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.

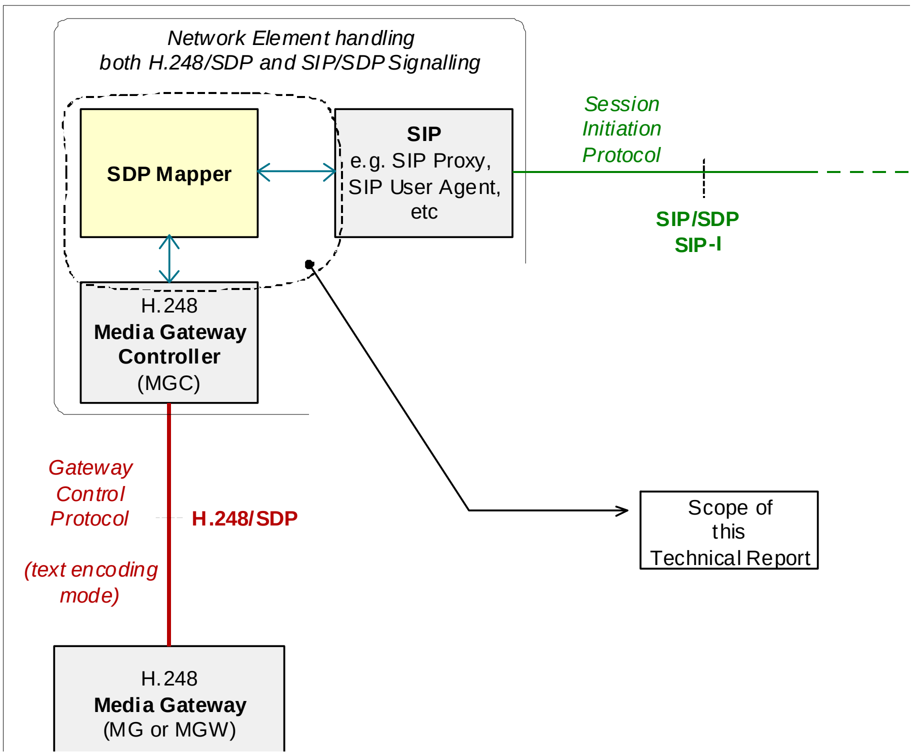

Figure 1: Scope

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;
  - for informative references.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

For online referenced documents, information sufficient to identify and locate the source shall be provided. Preferably, the primary source of the referenced document should be cited, in order to ensure traceability. Furthermore, the reference should, as far as possible, remain valid for the expected life of the document. The reference shall include the method of access to the referenced document and the full network address, with the same punctuation and use of upper case and lower case letters.

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

Void.

2.2 Informative references

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

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

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

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

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

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

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

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

[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".

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

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

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

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

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

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

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

[17] ETSI ES 283 002: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Release 1 H.248 Profile for controlling Access and Residential Gateways".

[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".

[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]".

[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".

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

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

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

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

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

[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").

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 this report.

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

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

3.2 Abbreviations

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

ALN	Analog Line
AMGW	Access Media GateWay
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
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
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)
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 [2].

In terms of bearer control and usage of SDP, SIP has defined an Offer/Answer model that is documented in RFC 3264 [4] and illustrated in RFC 4317 [6]. 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 [1] and it's predecessor RFC 2327 [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 [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>".

- **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 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 4.1.1.

Table 4.1.1: 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 [5] mandates that the ANSWER to a 18X OFFER must be included in the PRACK.
200 OK	ACK	Late SDP at both originating and terminating ends.

RFC 3264 [4] mandates that the same SDP Timing (t=) line must appear in both SDP blocks (i.e. the Offer and corresponding Answer) and that there must be identical numbers of Media Description (m=) lines in both SDP blocks (the Offer and corresponding Answer). The implication of the latter is there must 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) 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 [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 [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 [4], paragraph 8.4).
But there is one **applicability statement** in the context of Offer/Answer procedures (see RFC 3264 [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** MUST NOT be zero, which would specify that the stream has been **disabled** (see note 3). An SIP user agent MUST 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).

NOTE 3: The usage of a null port number within SDP was not yet standardized in the past (before RFC 3264 [4]). There does not exist any normative or informative text, neither from ETSI nor IETF. It is recognized that this mechanism has been already implemented, but the usage of the null port is not recommended for future implementations, although they still have to accept the null port from legacy implementations.
It has also to be noted that the "null port" relates to the well-known port category in case of UDP and TCP, which is still reserved by IANA (www.iana.org/assignments/port-numbers), i.e. not allowed to be used for these transport protocols.

4.1.2 Subsequent Offer/Answer Exchange(s)

It is possible to perform mid-session bearer modifications via subsequent Offer/Answer exchanges.

The new SDP Offer is conveyed either in an UPDATE message or else via a re-INVITE. A re-INVITE may only be used in the post Answer (200 OK to INVITE) phase of the session. An UPDATE may be used once a dialogue has been established. The resulting SDP Answer is returned in the 200 OK (either to UPDATE or re-INVITE).

RFC 3264 [4] applies a number of rules regarding the subsequent Offer/Answer exchange:

- the same Timing (t=) line must be used as previously;
- the same Session Name (s=) line must be used as previously;
- the Origin (o=) line is unchanged apart from the session version being incremented. Note that the session version is incremented any time that the sent SDP (be it an Offer or Answer) has been altered (or to put it another way, if the version has not changed, then the SDP must be identical to that previously sent);
- the same number of Media Description (m=) lines must be present as previously.

A media flow (which could related to an H.248 Stream or Termination) may be disabled via the "a=inactive" mechanism and/or null IP/port addresses.

A new media flow (e.g. bearer redirection) may be enabled via exchanging a new address and port in the Connection Data (c=) and Media Description (m=) lines respectively or via the Media Attribute (a=active) line. The contents of Media Description lines, Connection lines and Media Attribute lines can be altered as desired (e.g. to change address / port / media format / codec list etc.).

4.1.3 Bearer Termination

At SIP session termination, there is no explicit tear down of the bearer, i.e. the SIP BYE terminates the SIP session and the underlying bearer (e.g. RTP session) is also destroyed, e.g.:

- Bearer endpoint located in a SIP user equipment: the bearer is implicitly destroyed as a result of the SIP BYE.
- Bearer endpoint located in an H.248 MG: the SIP BYE will lead to an H.248 Subtract request command from MGC to MG, which then releases the underlying bearer in the MG.

4.2 H.248 Usage of SDP

4.2.1 Local and Remote Descriptor

The H.248 protocol [3] mandates the use of SDP in the H.248 LocalDescriptor (LD) and RemoteDescriptor (RD), when text encoding the H.248 protocol messages.

For the LD sent from the MGC to the MGW, a number of exceptions from RFC 4566 [1] are permitted:

- the "s=", "t=" and "o=" lines are optional;
- the use of the CHOOSE wildcard is allowed in place of a single parameter value;
- the use of alternatives is allowed in place of a single parameter value.

The LD returned from the MGW shall contain the "s=", "t=" and "o=" lines. Furthermore, if the RD is returned from the MGW, the RD shall contain the "s=", "t=" and "o=" lines as well.

In H.248, separate LD/RD are provided per media stream (i.e. within a H.248 StreamDescriptor) within a termination. Therefore, for multimedia (e.g. audio and video), separate StreamDescriptors must be used (see figure 4.2.1). H.248 does not permit multiple Media Description ("m=") lines to be present in a single session (= single H.248 Stream) description. Within a single Media Description line, multiple codecs may be specified and they are interpreted as a request to select one or more of the list options, with the list being in descending order of preference (see clause 7.1.8/H.248.1 [3]). However, H.248 does allow multiple session descriptions to be included as alternatives within a single LD/RD and each of these session descriptions containing a single Media Description line.

To enable interpretation of multiple session descriptions and/or multiple codecs within a Media Description line, H.248 has defined two additional flags, namely *ReserveGroup* and *ReserveValue*. The former indicates whether resource reservation is required to support all or one of the (multiple) session descriptions whilst the latter indicates whether resource reservation is required for all or one of the cited codecs in the Media Description line. If there is only one session description present, then ReserveGroup is redundant.

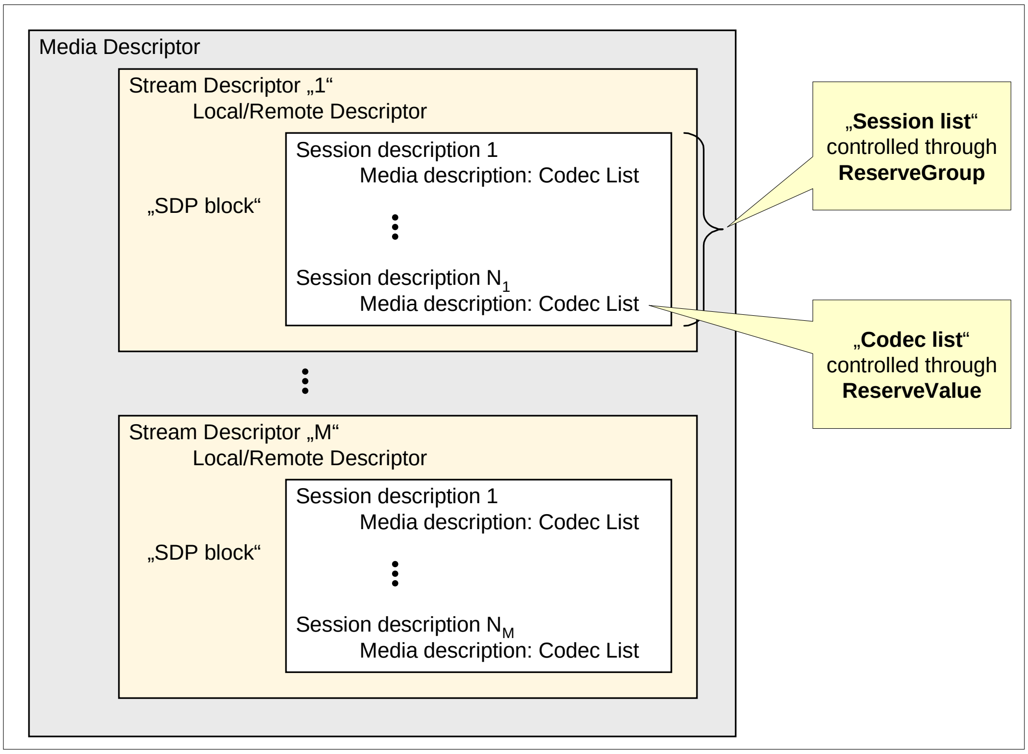

Figure 4.2.1: "SDP Blocks" embedded in H.248 Media Descriptor (here: *M* H.248 Streams per Termination)

The use of multiple session descriptions as opposed to multiple codecs on a Media Description line is somewhat "confusing" in H.248. Multiple session descriptions do enable separate Media Attribute lines to be specified for the audio codec(s) in a given session description, e.g. consider the following two examples of a H.248 LD.

NOTE 1: The reason is of historical nature: "codec negotiation" procedures for H.248 were defined before the SDP Offer/Answer model was published. See ITU-T T01-SG16-040120-D-0410 [26]: "H.248's use of SDP *limits the session descriptions to a single m-line per v-line*. This was done to solve the problem of determining which m-lines were to be concurrent sessions as opposed to alternative sessions. Several years later, the SDP community introduced the offer/answer model described in IETF RFC 3264 [4] which requires all m-lines to exist in a single session (v-line). The two solutions are mutually exclusive: Entities that use RFC 3264 will reject H.248 compliant SDP as invalid, and H.248 entities will reject RFC 3264 constructs as invalid."

NOTE 2: Only the Local Descriptor is shown, other information elements of the H.248 Message are omitted.

Example 1: Local Descriptor (H.248/SDP)

```
…
     Local {
             v=0
             c=IN IP4 $
             m=audio $ RTP/AVP 4 8
             a=ptime:30
} …
```

Example 2: Local Descriptor (H.248/SDP)

```
…
     Local {
             v=0
             c=IN IP4 $
             m=audio $ RTP/AVP 4
             a=ptime:30
             v=0
             c=IN IP4 $
             m=audio $ RTP/AVP 8
             a=ptime:20
} …
```

The first block contains a single session description with multiple codecs and a Media Attribute line (*ptime*) that is compatible with both codecs (i.e. G.723.1 and G.711 A-law (PCMA)). In the latter block, different packetization times have been specified (and G.723.1 requires 30 ms as a default packetization time and cannot use 20 ms due to the inherent codec frame size of 30 ms). The use of multiple session descriptions is confusing and could be avoided by regarding the *ptime* as a preference rather than a mandate and letting the MG override the preference where there is a mismatch with the codec requirements. Alternately, the *ptime* may be omitted and the MG can apply a default *ptime* appropriate to the codec(s), i.e.

Example 3: Local Descriptor (H.248/SDP)

```
…
     Local {
             v=0
             c=IN IP4 $
             m=audio $ RTP/AVP 4 8
} …
```

4.2.2 Wildcarding of SDP fields

The H.248 protocol supports the two wildcard type CHOOSE and ALL, which may be applied also on SDP information elements carried with H.248. ITU-T Recommendation H.248.39 [12] describes all the principles used to identify a single SDP sub-field and how to apply wildcarding to that sub-field.

5 Summary of SDP Usage Differences and Mapping Rules

The differences of SDP usage between SIP and H.248 are listed in table 5a.

Table 5a: SDP usage differences between H.248/SDP and SIP/SDP

No.	Issue	Differences
1	Number of Session Descriptions	H.248 permits multiple Session Descriptions per SDP block whilst SIP permits only one.
2	Number of Media Descriptions lines	H.248 permits only one Media Description ("m= ") line per Session Description whilst SIP permits multiple Media Description lines. In practice, SIP uses a Media Description line per media type (e.g. audio, video) but in theory could also specify multiple Media Description lines of a given media type in order to explicitly define different media attributes.
3	Specific SDP lines	Lines "s=", "t=" and "o=" are • optional in MGC-to-MG direction and • mandatory in MG-to-MGC direction in H.248/SDP. T.38: reference [10] clause V.3.4 highlights case differences for "udptl (UDPTL) " and „T38MaxBitRate (T38maxBitRate) " for SIP/SDP and H.248/SDP, and proposes a solution. The present document follows the guidelines of clause V.3.4/T.38.
4	Control of media source/sink	H.248 is using the StreamMode property (for Inactive, SendOnly, RecvOnly, SendRecv and LoopBack configurations) whilst SIP is using a dedicated "a=" line (for *sendonly*, *recvonly*, *inactive* and *sendrecv* attributes)
5	Impact of Offer/Answer rules	Aspects of the SIP Offer/Answer rules mean that certain SDP lines cannot simply be transited through a Call Server. The implication of this is that a Call Server must break down SDP blocks to ensure correct interworking between SIP and H.248.

There are two directions of SDP interworking (see figure 5).

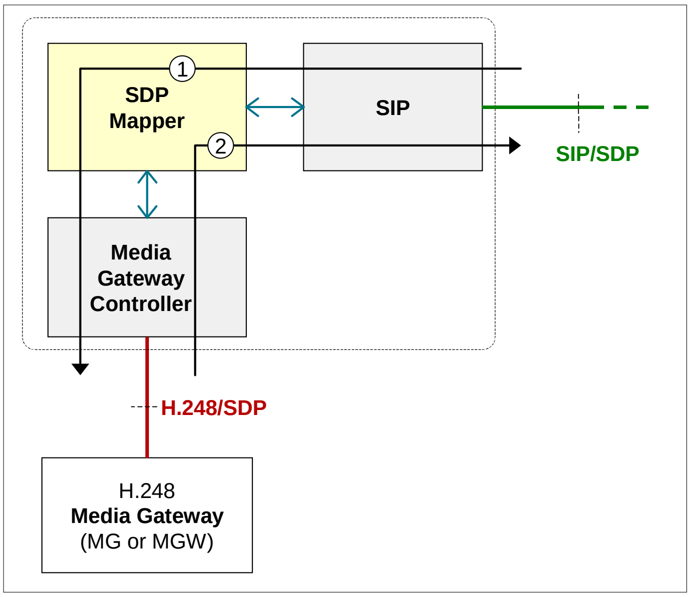

Figure 5: SDP Mapping Directions - SIP-to-H.248 (1) and H.248-to-SIP (2)

The SDP mapping rules are summarized in table 5b.

Table 5b: SDP Mapping Rules H.248/SDP and SIP/SDP

No.	Issue	Mapping Rule	Direction	
			SIP-to-H.248	H.248-to-SIP
1	Number of Session Descriptions	Issues 1 and 2 are related in how they are handled by the SDP Mapper: Each SIP Media Description line is mapped to a separate H.248 session descriptor within a H.248 Stream containing a single Media Description line.	X	
2	Number of Media Description lines			
		Each H.248 Stream Descriptor Media Description line is mapped to a separate SIP/SDP Media Description line.		X
3	Specific SDP lines	The "o= ", "s= ", "t= " lines may be created by the SDP Mapper towards H.248/SDP.	X	
		The "o= ", "s= ", "t= " lines must be included and adapted by the SDP Mapper to the required SIP/SDP usage.		X
		T.38: The SDP mapper shall allow to receive case differences, but shall sent SIP/SDP and H.248/SDP according clause V.3.4/T.38: • Transport protocol: "udptl " and • Attribute field: "T38MaxBitRate " (see note).	X	X
4	Control of media source/sink	The H.248 Stream Mode must be reflected in an Media Attribute line of the SIP/SDP. Note that according to [3] an omitted H.248 stream mode property has a default value ='inactive'.		X
		Conversely, the SIP/SDP Media Attribute line must be reflected in the corresponding H.248 Stream Mode. Note that according to [4] an omitted directionality attribute has a default value = "sendrecv ".	X	
5	Impact of SIP Offer/Answer rules	Changes in the H.248/SDP must be reflected in SIP/SDP such that: • The "s= " line is unchanged. • The "o= " line "session version " number must be incremented. • The number of "m= " lines cannot be reduced. • The same "t= " line is used in a given Offer/Answer exchange.		X
		Changes in SIP/SDP are validated in being in line with Offer/Answer rules and are then reflected as a new Remote Descriptor and/or specific settings of *ReserveValue* and/or *ReserveGroup* in H.248/SDP.	X	
NOTE: These SDP parameters are consistent with the IANA registered values.				

These mappings are applied by the SDP Mapper to provide interworking between SIP/SDP and H.248/SDP. Example mappings are described in clause 6.

5.1 ITU-T Recommendation V.152 mapping rules

According to ITU-T Recommendation V.152 [11] sections 7.1, 7.1.1 and 7.1.2, the same SDP elements, like:

- `a=gpmd:<format> <parameter list>`, whereas specifically the "vbd=yes(no)" parameter/value pair is of interest;
- `a=maxmptime:<list of packet times separated by space>.`

are to be used for SDP based session description for SIP and H.248.

As an option, the SDP mapper may transform an incoming SIP offer, carrying V.152 "vbd=yes" parameter/value pairs but using individual "ptime" attributes, into a H.248/SDP request using maxmptime attribute:

`a=maxmptime:<list of packet times separated by space>.`

EXAMPLE:

Table 5.1: Re-Formatting SDP elements: Example of VBD using V.152

<table>
<tr><td colspan="2">Example of media descriptions for audio, RFC 4733 [22] packet types for telephony events and VBD</td></tr>
<tr><td>Reference:</td><td>-</td></tr>
<tr><td>Description:</td><td>In this example, "A" wishes to establish an audio stream, with telephone-events and dedicated codec declarations for potential VoiceBand data transmission.
"A" offers one stream, one audio for PCMA, RFC 4733 [22] tones (for DTMF) and 2 VBD codecs with individual ptimes.
SDP mapper applies maxmptime attribute in the H.248 request message.</td></tr>
<tr><td>SIP/SDP ('A' side):</td><td>H.248/SDP ('B' side):</td></tr>
<tr><td>(1) – Incoming "Offer":

…
v=0
o=- 1234 0 IN IP4 172.17.2.31
s=SIP Call
c=IN IP4 172.17.2.31
t=0 0
m=audio 6000 RTP/AVP 8 101 102 103
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=rtpmap: 102 PCMA/8000
a=gpmd:102 vbd=yes
a=ptime:30
a=rtpmap: 103 G726-32/8000
a=gpmd:102 vbd=yes
a=ptime:30</td><td>(2) – H.248 ADD request from MGC towards MG:

MEGACO/3 [11.9.19.65]:12345
Transaction = 31205 {
Context = $ {
Add = $ {
Media{
Stream=1{
LocalControl{
Mode=SendReceive,
ReservedGroup = OFF,
ReservedValue = ON},
Local{
v=0
c=IN IP4 $
m=audio $ RTP/AVP 8 101 102 103
a=maxmptime: 20 - 30 30 NOTE1, NOTE2
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=rtpmap: 102 PCMA/8000
a=gpmd:102 vbd=yes
a=rtpmap: 103 G726-32/8000
a=gpmd:102 vbd=yes},
Remote{
v=0
o=- 1234 0 IN IP4 172.17.2.31 ;
s=SIP Call
c=IN IP4 172.17.2.31
t=0 0
m=audio 6000 RTP/AVP 8 101 102 103
/* from SIP m=audio line and attributes */
a=maxmptime: 20 - 30 30
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=rtpmap: 102 PCMA/8000
a=gpmd:102 vbd=yes
a=rtpmap: 103 G726-32/8000
a=gpmd:102 vbd=yes }
},
/* of Stream 1 */

}}}
…}</td></tr>
<tr><td colspan="2">NOTE 1: Default packetization period defined in RFC3550 is applied for audio with PCMA.
NOTE 2: In case the Remote Descriptor is not present the maxmptime attribute line may be underspecified.</td></tr>
</table>

5.2 ITU-T Recommendation T.38 mapping rules

According to [10] there is no specific SDP usage defined for SIP/SDP and H.248/SDP. T.38 attribute definitions as defined in [10], chapter D.2.3 are applicable for SDP usage within SIP and H.248.

If "T.38 Autonomous Transitioning method" is not supported, this shall be signalled in accordance to clause 4.1.1 ("null (zero) port number"). Furthermore the example in table 6.2.3 applies regarding the way a not supported stream is communicated on the H.248 interface.

5.3 Packetization times in SDP

The specification of packetization times with SDP is so far either "codec-independent", e.g. the RFC 4566 [1] attributes "*ptime*" and "*maxptime*" modify the whole media description line as such, or in case of the "*vsel*" attribute according to RFC 3108 [21], linked to a codec type.

NOTE: The '*vsel*' attribute indicates a prioritized list of one or more 3-tuples. Each 3-tuple indicates a *codec type*, an optional *packet length* and an optional *packetization period*.

The codec-independent specification of packetization times might be a particular issue in case of a codec list (see also problem statement as discussed by IETF working group MMUSIC). There might be future SDP versions with support of packetization time indications on a per codec basis.

Such a SDP extension would not necessarily lead to additional mapping rules for the present document, but may affect existing mapping rules which are using non-IETF SDP extension (e.g. like ITU-T Recommendation V.152 [11], see clause 5.1).

6 SDP Mapping Examples

This clause describes some example mappings between SIP/SDP and H.248/SDP.

6.1 SIP/SDP to H.248/SDP Example

The SIP/SDP block will contain a single session description with (in general) multiple Media Description lines. Each Media Description line in SIP typically relates to a different media type (e.g. audio, video) but it is also (theoretically) possible to have multiple Media Description lines relating to the same media type.

Basically SIP/SDP Media Description lines can be mapped to H.248 in different ways:

- SIP/SDP Media Description lines are mapped to separate session descriptions, belonging to one H.248 Stream.
- SIP/SDP Media Description lines are mapped to separate H.248 Streams.

In the following the approach "SIP/SDP Media Description lines are mapped to separate H.248 Streams" is outlined. Within each StreamDescriptor, there will be a single session description containing the LocalControl Descriptor with the appropriate StreamMode and the appropriate Media Description line (i.e. the codec list in a given SIP Media Description line would be copied across to the corresponding Media Description line in the corresponding session description in the appropriate StreamDescriptor).

Consider the following SIP/SDP block advertising two Media Description lines (audio and video) with PCMA (Pulse Code Modulation in A-law encoding according ITU-T Recommendation G.711 [14]) and RFC 4733 [22] as audio codecs and H.261 (video codec for audiovisual services at *p* x 64 kbit/s) as a video codec (see step 1 in table 6.1).

This would be mapped two RDs (i.e. one per "m=" line), each within a separate StreamDescriptor as follows (see step 2 in table 6.1).

It should be noted that in some cases, the H.248 end (= H.248 IP Stream/Termination in H.248 MG) would not be able to support all of the offered media types (e.g. an AMGW would be audio only) and thus the SDP Mapper would be able to map to a single stream to such a MGW. However, in general, the above mapping is valid/possible, - e.g. a SIP call encountering a H.248 controlled BGF.

The response from the H.248 MGW can return a fully specified LDs, e.g. see step 3 in table 6.1.

The returned H.248/SDP is now mapped back into the SIP/SDP, as described in clause 6.2 and illustrated as step 4 in table 6.1.

Table 6.1: Re-Formatting SDP elements: Example of an Audio/Video Description

Example of media descriptions for audio, RFC 4733 [22] packet types for telephony events and video	
Reference:	-
Description:	In this example, "A" wishes to establish separate audio/video streams, one for normal audio, one for telephone-events and one for video. "A" offers separate streams, one audio for PCMA and RFC 4733 [22] tones (for DTMF), another one for video. "B" accepts all media types.
SIP/SDP ('A' side):	**H.248/SDP** ('B' side):

(1) – Incoming "**Offer**":
 	
…
v=0
o=- 1234 0 IN IP4 172.17.2.31
s=SIP Call
c=IN IP4 172.17.2.31
t=0 0
m=audio 6000 RTP/AVP 8 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 9000 RTP/AVP 31
a=sendrecv
	(2) – H.248 ADD request from MGC towards MG:
	
	MEGACO/3 [11.9.19.65]:12345
	Transaction = 31205 {
	Context = $ {
	Add = $ {
	Media{
	Stream=1{ ; NOTE 1
	LocalControl{
	Mode=SendReceive,
	ReservedGroup = OFF,
	ReservedValue = ON},
	Local{
	v=0
	c=IN IP4 $
	m=audio $ RTP/AVP 8 $
	a=rtpmap:$ telephone-event/8000
	a=fmtp:101 0-15},
	Remote{
	v=0
	o=- 1234 0 IN IP4 172.17.2.31 ; NOTE 2
	s=SIP Call
	c=IN IP4 172.17.2.31
	t=0 0
	m=audio 6000 RTP/AVP 8 101
	/* from SIP m=audio line and attributes */
	a=rtpmap:101 telephone-event/8000
	a=fmtp:101 0-15}
	},
	/* of Stream 1 */
	Stream=2{
	..**LocalControl**{
	Mode=SendReceive},
	ReservedGroup = OFF,
	ReservedValue = ON},
	Local{
	v=0
	c=IN IP4 $
	m=video $ RTP/AVP 31},
	Remote{
	v=0
	o=- 1234 0 IN IP4 172.17.2.31
	s=SIP Call
	c=IN IP4 172.17.2.31
	t=0 0
	m=video 9000 RTP/AVP 31
	/* from SIP m=video line and attributes

<table>
<tr><td colspan="2">Example of media descriptions for audio, RFC 4733 [22] packet types for telephony events and video</td></tr>
<tr><td>Reference:</td><td>-</td></tr>
<tr><td>Description:</td><td>In this example, "A" wishes to establish separate audio/video streams, one for normal audio, one for telephone-events and one for video.
"A" offers separate streams, one audio for PCMA and RFC 4733 [22] tones (for DTMF), another one for video.
"B" accepts all media types.</td></tr>
<tr><td>SIP/SDP ('A' side):</td><td>H.248/SDP ('B' side):</td></tr>
<tr><td></td><td><pre>*/
}}
/* of Stream 2 */

}}}
…}</pre></td></tr>
<tr><td>(4) – Outgoing "Answer"
(see discussion in clause 6.2):<pre>v=0
o=- 0 0 IN IP4 89.0.222.229
     /* or o=- 1 0 IN IP4 89.0.222.229 */
s=H.248 Context
c=IN IP4 89.0.220.229
t=0 0
m=audio 2000 RTP/AVP 8 101
    /* from H.248 Stream 1 LD */
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=video 4000 RTP/AVP 31
     /* from H.248 Stream 2 LD */
a=sendrecv</pre></td><td>(3) – H.248 reply from MG towards MGC:<pre>MEGACO/3 [2.3.19.70]:6789
Reply = 31205 {
 Context = C1 {
  Add = T1 {
   Media{
Stream=1{
  LocalControl{
   Mode=SendReceive,
   ReservedGroup = OFF,
   ReservedValue = ON},
  Local{
   v=0
   o=- 0 0 IN IP4 89.0.222.229 ; NOTE 2
   s=H.248 Context
   c=IN IP4 89.0.220.229
   t=0 0
   m=audio 2000 RTP/AVP 8 101
   a=rtpmap:101 telephone-event/8000
   a=fmtp:101 0-15}
},
/* of Stream 1 */
Stream=2{
..LocalControl{
   Mode=SendReceive,
   ReservedGroup = OFF,
   ReservedValue = ON},
  Local{
   v=0
   o=- 1 0 IN IP4 89.0.222.229
   s=H.248 Context
   c=IN IP4 89.0.220.229
   t=0 0
   m=video 4000 RTP/AVP 31}
}
/* of Stream 2 */

}}
…}</pre></td></tr>
<tr><td colspan="2">NOTE 1: Two H.248 Streams are used. Stream "1" is for audio and for RFC 4733 [22] information, and stream "2" for video.
NOTE 2: MGC did decide (a) to include "s=", "o=" and "t=" lines in the H.248/SDP, and did decide (b) to re-use these lines unmodified from SIP/SDP side.
(c) The MG must return the lines with the received values.</td></tr>
</table>

6.2 H.248/SDP to SIP/SDP Example

6.2.1 General Mapping

In this case, consider a BGF (MG) which has multiple H.248 StreamDescriptors, each containing a single session description each with a single Media Description line. Consider the mapping of the H.248 LD and LocalControl descriptor returned from the MG in clause 6.1 into SIP/SDP. To perform the mapping, the following steps are taken:

i) a single SIP session description is created; and

ii) a Media Description line is created per H.248 StreamDescriptor;

iii) the H.248 StreamMode is reflected in the Media Attribute line of SIP/SDP.

The resulting SIP/SDP block is shown as step 4 in table 6.1

6.2.2 Specific SDP Lines: Timing ("t=" Line)

In addition, rules of Offer/Answer also have an influence on the mapping between H.248/SDP and SIP/SDP. Specifically, for the initial Offer/Answer exchange, the same Timing ("t=") line must appear. In table 6.1, both blocks of SDP contained the default setting for the Timing line. However, if an Offer had been initially received from SIP, then the Answer would have had to echo the received Timing line in the Answer.

6.2.3 Specific SDP Lines: Media Descriptions ("m=" Line)

In addition, there must be the same number of Media Description ("m=") lines in an Offer/Answer exchange. In the example mapping in clause 6.1 from SIP to H.248, it was assumed that the H.248 entity supported multiple media types. However, in the case where the H.248 end supported audio only, then there would not be any associated SDP at the H.248 end for the unsupported media type of video. In this case, the Answer to SIP would include the "m=video" line with suitable parameters to denote that the video stream was disabled/unsupported. This is shown as step 4 in table 6.2.3.

The video stream is disabled by a combination of null IP port number (in the "m=" line) and Media Attribute line set to inactive.

Table 6.2.3: Re-Formatting SDP Elements: Example of an Audio/Video Description

<table>
<tr><td colspan="3">Example of media descriptions for audio, RFC 4733 [22] packet types for telephony events and video</td></tr>
<tr><td>Reference:</td><td colspan="2">Variation of example from table 6.1</td></tr>
<tr><td>Description:</td><td colspan="2">In this example, "B" accepts the audio types, but rejects the video codec.</td></tr>
<tr><td colspan="2">SIP/SDP ('A' side):</td><td>H.248/SDP ('B' side):</td></tr>
<tr><td colspan="2">(1) – Incoming "Offer":

Same as in table 6.1.</td><td>(2) – H.248 ADD request from MGC towards MG:

MEGACO/3 [11.9.19.65]:12345
Transaction = 31205 {
 Context = $ {
  Add = $ {
   Media{
Stream=1
  LocalControl{
   Mode=SendReceive,
   ReservedGroup = OFF,
   ReservedValue = ON},
  Local{
   v=0
   c=IN IP4 $
   m=audio $ RTP/AVP 8 $
   a=rtpmap:$ telephone-event/8000
   a=fmtp:$ 0-15},
  Remote{
   v=0
   o=- 1234 0 IN IP4 172.17.2.31
   s=SIP Call</td></tr>
</table>

Example of media descriptions for audio, RFC 4733 [22] packet types for telephony events and video	
Reference:	Variation of example from table 6.1
Description:	In this example, "B" accepts the audio types, but rejects the video codec.
SIP/SDP ('A' side):	**H.248/SDP** ('B' side):

	c=IN IP4 172.17.2.31
	t=0 0
	m=audio 6000 RTP/AVP 8 101
	/* from SIP m=audio line and attributes */
	a=rtpmap:101 telephone-event/8000
	a=fmtp:101 0-15}
	},
	/* of Stream 1 */
	Stream=2{
	..**LocalControl**{
	Mode=SendReceive,
	ReservedGroup = OFF,
	ReservedValue=ON}, ; NOTE 3
	Local{
	v=0
	c=IN IP4 $
	m=video $ RTP/AVP 31},
	Remote{
	v=0
	o=- 1234 0 IN IP4 172.17.2.31
	s=SIP Call
	c=IN IP4 172.17.2.31
	t=0 0
	m=video 9000 RTP/AVP 31
	/* from SIP m=video line and attributes */
	}}
	/* of Stream 2 */
	
	}}}
	…}

(4) – Outgoing "**Answer**":
 	
v=0
o=- 0 0 IN IP4 89.0.222.229
s=H.248 Context
c=IN IP4 89.0.220.229
t=0 0
m=audio 2000 RTP/AVP 8 101
/* from H.248 audio Stream Descriptor */
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
m=**video 0** RTP/AVP 31 ; NOTE 2
/* echoed back to SIP and disabled */
a=inactive ; NOTE 2
	(3) – H.248 reply from MG towards MGC:
	
	MEGACO/3 [2.3.19.70]:6789
	Reply = 31205 {
	Context = C1 {
	Add = T1 {
	Media{
	Stream=1{
	LocalControl{
	Mode=SendReceive,
	ReservedGroup = OFF,
	ReservedValue = ON},
	Local{
	v=0
	o=- 0 0 IN IP4 89.0.222.229
	s= H.248 Context
	c=IN IP4 89.0.220.229
	t=0 0
	m=audio 2000 RTP/AVP 8 101
	a=rtpmap:101 telephone-event/8000
	a=fmtp:101 0-15}
	},
	/* of Stream 1 */
	Stream=2{
	LocalControl{
	Mode=SendReceive,
	ReservedGroup = OFF,
	ReservedValue=ON},
	Local{} ; NOTE 3/* of Stream 2 */
	
	}}

Example of media descriptions for audio, RFC 4733 [22] packet types for telephony events and video	
Reference:	Variation of example from table 6.1
Description:	In this example, "B" accepts the audio types, but rejects the video codec.
SIP/SDP ('A' side):	**H.248/SDP** ('B' side):
	…}
NOTE 1: Video is rejected by the MG, i.e. there is finally only Stream '1'. NOTE 2: Video is disabled via indications in "m= " and "a= " lines. NOTE 3: In this example the MGC sets ReservedValue=ON, which enables the MG to report "insufficient resources " for this particular stream according to H.248.1v3 [3] chapter 7.1.8. If the MGC would set ReservedValue=OFF, the MG has to return an error descriptor with Error Code 510 for the entire command, if not being able to support at least one of the requested resources. Thus in this case the H.248 context establishment fails. Note that Stream level error descriptors are not defined in H.248.1v3 [3])	

6.2.4 Specific SDP Lines: Origin ("o=" Line)

For subsequent Offer/Answer exchanges, the rules of clause 5. would apply - i.e. the same "t=" line, the same "s=" line, the "o= " line unchanged apart from an incremented *session version* field (<sess-version>) and the same number of "m=" lines. Therefore, in the case where the above SDP block had been previously sent but the audio stream had now become disabled (e.g. due to bearer modification at the H.248 end), the following new Offer could be made, step 5 in table 6.2.4.

Table 6.2.4: Re-Formatting SDP Elements: Example of an Audio/Video Description

Example of media descriptions for audio, RFC 4733 [22] packet types for telephony events and video	
Reference:	Further variation of example from table 6.1
Description:	In this example, there is a subsequent SIP Offer/Answer cycle. The 2nd Offer leads to modifications in the "o= " line and a reduction of "m= " lines.
SIP/SDP ('A' side):	**H.248/SDP** ('B' side):

(5) – Incoming 2nd "**Offer**"
(1st Offer see step 1 in table 6.1):
 	
`v=0`
`o=- 0 `**`1`**` IN IP4 89.0.222.229 ; NOTE 1`
`s=H.248 Call`
`c=IN IP4 89.0.220.229`
`/* IP address of 0.0.0.0 could also be sent */`
`t=0 0`
`m=audio `**`0`**` RTP/AVP 8 101 ; NOTE 2`
`a=rtpmap:101 telephone-event/8000`
`a=fmtp:101 0-15`
`a=inactive`
`m=video 0 RTP/AVP 31 ; NOTE 3`
`a=inactive`
	(6) – H.248 **MODIFY** request from MGC towards MG:
	
	Not considered.

(8) – Outgoing "**Answer**": Not considered.	(7) – H.248 reply from MG towards MGC: Not considered.
NOTE 1: The value of <sess-version> is incremented. NOTE 2: All audio codecs are disabled. NOTE 3: The video codec is disabled.	

6.3 Network Examples

6.3.1 Pure PES scenario

Originating and terminating side could be both located in the PSTN/ISDN. Such a scenario shall be called "pure PSTN emulation subsystem "scenario. The PSTN/ISDN bearers are connected via H.248 Access or Trunking MGs to the IP domain. There will be thus either an AMG-to-AMG, AMG-to-TMG (see figure 6.3.1) or TMG-to-TMG network configuration. One or two MGs would be involved. In case of two MGs might be only one MGC for both, or each MG is controlled individually by an MGC. Only the last use case is subject of this report because there will be an SIP/SDP interface for call/session control signalling between both network elements housing the MGC instances. SIP-I is applied at this interface, which relates to MIME encoding of ISUP in SIP messages.

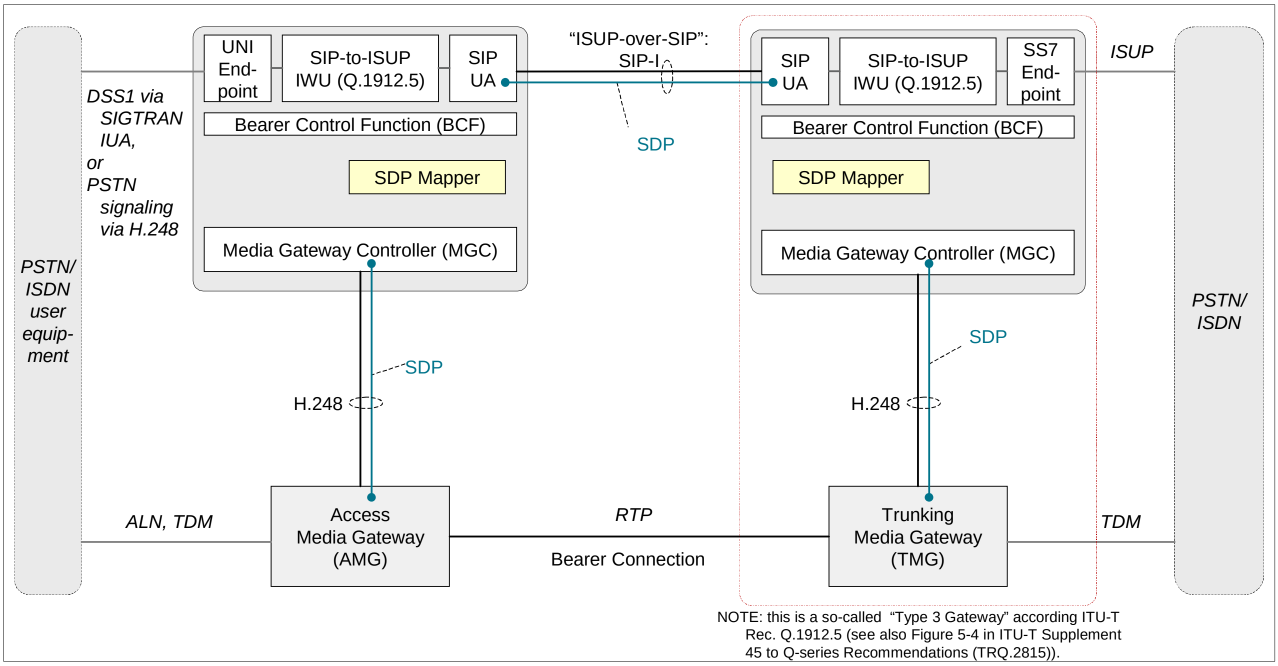

Figure 6.3.1: Pure PES scenario with SIP-I/SDP between "MGCs" (here: AMG-to-TMG scenario with two MGC entities)

The H.248 interfaces are defined by the H.248 Access/Residential MG Profile (ES 283 002 [17]) and the H.248 Trunking MG Profile (ES 283 024 [18]).

The SIP/SDP interface is based on "encapsulated ISUP over SIP" (called SIP-I), according EN 383 001 [19] (which is based on ITU-T Recommendation Q.1912.5 [8]; in particular relates SIP-I, which relates to profile C in Q.1912.5 [8]). Following is relevant for this report concerning SDP usage:

- Conventions for representation of SDP information in Q.1912.5 [8] is based on RFC 2327 [7].
- Coding of SDP media description lines from TMR/USI elements is generally described in clause 7.1.1/Q.1912.5 [8].

NOTE: See also clause 5.1.1.1 "Mapping from ISUP bearer to RTP using SDP" in TR 183 014 [20] for PSTN/ISDN emulation.

- SDP media description data within a SIP-I message (relates to SIP content type "application/SDP") is reduced to a very minimum due to the encapsulated ISUP message (relates to SIP content type "application/ISUP").

In summary, there are not any extra SDP mapping rules for SIP-I/SDP in addition to SIP/SDP.

6.3.2 End-to-end Offer/Answer scenario with a RFC 3264-based SIP interface

Media (e.g. codec types) negotiation, determination and/or selection may require the conversion of SDP descriptors.

6.3.2.1 Overview

The protocol elements of SIP/SDP and H.248/SDP for negotiation or determination of session/media information are slightly different. SIP/SDP is based on the model of IETF RFCs 3264 [4] and 4317 [6], whereas H.248 is using the concept of "reserve properties" (ReservedValue and ReservedGroup; see clauses 7.1.7 and 7.1.8 in H.248.1 [3] for syntax, usage and resource reservation rules).

6.3.2.2 Two Audio Streams

This example is based on the recommendation given by paragraph 2.4/RFC 4317 [6]. An "offer" (1) is initiated by SIP UA (A). Figure 6.3.2.2 illustrates the result after negotiation. Potential interfaces to DNS servers from SIP UA, call server or MG are omitted in figure 6.3.2.2. The remaining bearer connection segment towards the called party is also omitted.

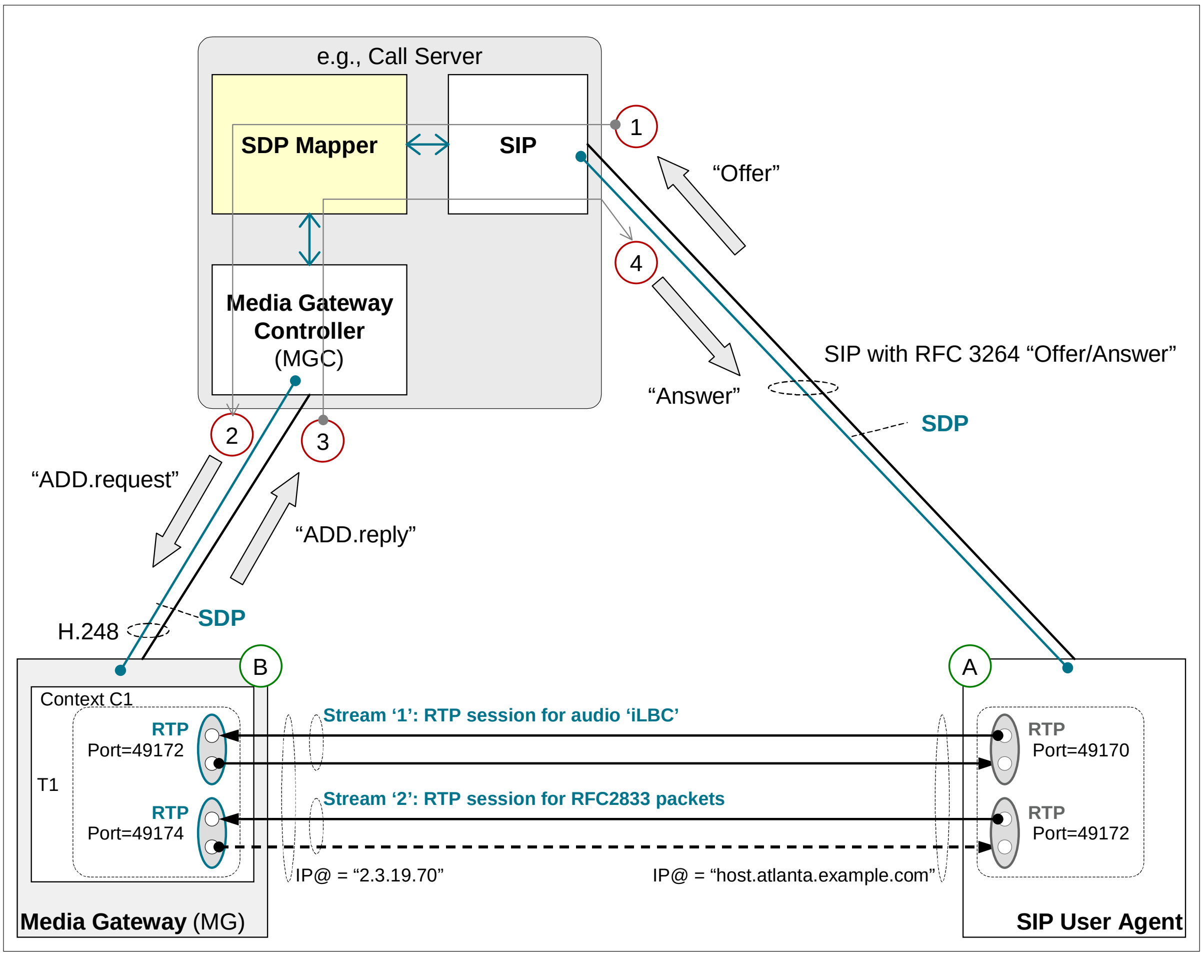

Figure 6.3.2.2: Four step signalling scenario - Result after Negotiation

The given SIP "answer" (4) with the chosen audio codec "iLBC" (internet Low Bit Rate Codec; see IETF RFCs 3951 [15] and 3952 [16]) requires the distinction of sub-cases (see clauses 6.3.2.2.1 and 6.3.2.2.2), which are underlining again the H.248 reservation rules and resulting formatting rules of H.248/SDP information.

6.3.2.2.1 H.248 MG does not support G.711 (as Audio Codec)

See table 6.3.2.2.1.

Table 6.3.2.2.1: Re-Formatting SDP Elements: Example of Two Audio Streams

<table>
<tr><td colspan="2">Example of two audio streams</td></tr>
<tr><td>Reference:</td><td>paragraph 2.4/RFC 4317 [6]
NOTE1: this might be a theoretical example, not applicable to TISPAN scenarios.
NOTE2: in principle there is no need to setup a dedicated stream for telephony events according to RFC4733 [22], as RTP packets for normal audio and RTP packets for telephony events can be unambiguously differentiated based on RTP payload types.</td></tr>
<tr><td>Description:</td><td>In this example, "A" wishes to establish separate audio streams, one for normal audio and the other for telephone-events.
"A" offers two separate streams, one audio with two codecs and the other with RFC 4733 [22] tones (for DTMF).
"B" accepts both audio streams choosing the iLBC codec and telephone-events.</td></tr>
<tr><td>Assumptions:</td><td>H.248 MG supported resource component types:
no support of G.711 as audio codec</td></tr>
<tr><td>SIP/SDP ('A' side):</td><td>H.248/SDP ('B' side):</td></tr>
<tr><td>(1) – Incoming "Offer":

v=0
o=alice 2890844526 2890844526 IN IP4 host.atlanta.example.com
s=
c=IN IP4 host.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0 97
a=rtpmap:0 PCMU/8000
a=rtpmap:97 iLBC/8000
m=audio 49172 RTP/AVP 98
a=rtpmap:98 telephone-event/8000
a=sendonly</td><td>(2) – H.248 ADD request from MGC towards MG:
MEGACO/3 [11.9.19.65]:12345
Transaction = 31205 {
 Context = $ {
  Add = $ {
   Media{
    Stream = 1 { ; NOTE 1
     LocalControl {
      Mode = SendReceive, ; NOTE 2
      ReservedGroup = OFF, ; NOTE 3
      ReservedValue = OFF}, ; NOTE 4
     Local { ; NOTE 5
v=0 ; NOTE 6
c=IN IP4 $; NOTE 7
m=audio $ RTP/AVP 0 97 ; NOTE 7,18
a=rtpmap:97 iLBC/8000
}
     Remote { ; NOTE 5
v=0
c=IN IP4 host.atlanta.example.com ; NOTE 19
m=audio 49170 RTP/AVP 0 97 ; NOTE 18
a=rtpmap:97 iLBC/8000
}}}
    Stream = 2 { ; NOTE 1
     LocalControl {
      Mode = ReceiveOnly, ; NOTE 8
      ReservedGroup = OFF, ; NOTE 9
      ReservedValue = OFF},
     Local {
v=0
c=IN IP4 $; NOTE 7
m=audio <b>$ RTP/AVP 98 ; NOTE 7
a=rtpmap:98 telephone-event/8000
}
     Remote {
v=0
c=IN IP4 host.atlanta.example.com; NOTE 19
m=audio 49172 RTP/AVP 98
a=rtpmap:98 telephone-event/8000
}}}
...}</td></tr>
<tr><td>(4) – Outgoing "Answer":

v=0
o=bob 2808844564 2808844564 IN IP4</td><td>(3) – H.248 reply from MG towards MGC:
MEGACO/3 [2.3.19.70]:6789
Reply = 31205 {
 Context = C1 {</td></tr>
</table>

Example of two audio streams	
Reference:	paragraph 2.4/RFC 4317 [6] NOTE1: this might be a theoretical example, not applicable to TISPAN scenarios. NOTE2: in principle there is no need to setup a dedicated stream for telephony events according to RFC4733 [22], as RTP packets for normal audio and RTP packets for telephony events can be unambiguously differentiated based on RTP payload types.
Description:	In this example, "A" wishes to establish separate audio streams, one for normal audio and the other for telephone-events. "A" offers two separate streams, one audio with two codecs and the other with RFC 4733 [22] tones (for DTMF). "B" accepts both audio streams choosing the iLBC codec and telephone-events.
Assumptions:	H.248 MG supported resource component types: no support of G.711 as audio codec

SIP/SDP **('A' side):**

```
host.biloxi.example.com
s=
c=IN IP4 host.biloxi.example.com
t=0 0
m=audio 49172 RTP/AVP 97
a=rtpmap:97 iLBC/8000
m=audio 49174 RTP/AVP 98
a=rtpmap:98 telephone-event/8000
a=recvonly
```

H.248/SDP **('B' side):**

```
  Add = T1 {
   Media{
    Stream = 1 {               ; NOTE 10
     Local {
v=0
o=- 0 0 IN IP4 2.3.19.70    ; NOTE 11
s=-                         ; NOTE 12
t=0 0                       ; NOTE 13
c=IN IP4 19.65.9.11         ; NOTE 14
m=audio 49172 RTP/AVP 97    ; NOTE 15
a=rtpmap:97 iLBC/8000       ; NOTE 16
}
     Remote {
v=0
o=- 0 0 IN IP4 2.3.19.70    ; NOTE 11
s=-                         ; NOTE 12
t=0 0                       ; NOTE 13
c=IN IP4 host.atlanta.example.com;
NOTE 19
m=audio 49170 RTP/AVP 97
a=rtpmap:97 iLBC/8000
}}
    Stream = 2 {
     Local {
v=0
o=- 0 0 IN IP4 2.3.19.70    ; NOTE 11
s=-                         ; NOTE 12
t=0 0                       ; NOTE 13
c=IN IP4 19.65.9.11         ; NOTE 14
m=audio 49174 RTP/AVP 98    ; NOTE 17
a=rtpmap:98 telephone-event/8000
}
     Remote {
v=0
c=IN IP4 host.atlanta.example.com;
NOTE 19
m=audio 49172 RTP/AVP 98
a=rtpmap:98 telephone-event/8000
}}
…}
```

<table>
<tr><td colspan="2">Example of two audio streams</td></tr>
<tr><td>Reference:</td><td>paragraph 2.4/RFC 4317 [6]
NOTE1: this might be a theoretical example, not applicable to TISPAN scenarios.
NOTE2: in principle there is no need to setup a dedicated stream for telephony events according to RFC4733 [22], as RTP packets for normal audio and RTP packets for telephony events can be unambiguously differentiated based on RTP payload types.</td></tr>
<tr><td>Description:</td><td>In this example, "A" wishes to establish separate audio streams, one for normal audio and the other for telephone-events.
"A" offers two separate streams, one audio with two codecs and the other with RFC 4733 [22] tones (for DTMF).
"B" accepts both audio streams choosing the iLBC codec and telephone-events.</td></tr>
<tr><td>Assumptions:</td><td>H.248 MG supported resource component types:
no support of G.711 as audio codec</td></tr>
<tr><td>SIP/SDP ('A' side):</td><td>H.248/SDP ('B' side):</td></tr>
<tr><td colspan="2">NOTE 1: Two H.248 Streams are required. Stream '1' is for audio and stream '2' for RFC 4733 [22] information.
NOTE 2: It is supposed that the SDP mapper translates a lacking attribute in media codec specifications of SIP/SDP "offers" (here: missing line 'a=sendrecv') into H.248 StreamMode "SendReceive".
NOTE 3: Default value "Off" is used because there is only a single "media group" in Stream '1'.
NOTE 4: The MG is to reserve a single set of the property values indicated. This is the actual "negotiation" decision: selection of one out of two possible codecs. [The decision is here given by the SIP/SDP "answer" in the discussed example of paragraph 2.4/RFC 4317 [6].].
NOTE 5: Symmetrical codec usage is considered, thus "SDP mapper" is using same media types in H.248 LD and RD.
NOTE 6: MGC did decide to delete 's=', 'o=' and "t=" lines.
NOTE 7: Resource management of resource component types related to logical/physical IP interfaces ("IP addresses", "IP ports") shall be under MG responsibility. The MGC is therefore applying wildcarding here.
NOTE 8: SIP/SDP "sendonly" is mapped to H.248 StreamMode and inverted to value "RecvOnly", because this is a unidirectional communication only: from RTP endpoint "A" to "B".
NOTE 9: ReservedGroup is also false due to single media element.
NOTE 10: The protocol elements of LocalControl Descriptor ("StreamMode", "ReservedGroup" and "ReservedValue") are omitted in the reply.
NOTE 11: The MG inserts an 'o=' line in his reply (see table 5b, rule (3)). In the example here is the numerical IP address of the MG's IP interface for H.248 signalling transport used.
NOTE 12: The MG insert an 's=' line in his reply (see table 5b, rule (3)). It has to be noted that there is a small difference (in above example) of the 's=' line encoding at H.248/SDP and SIP/SDP interface.
NOTE 13: The MG inserts an 't=' line in his reply (see table 5b, rule (3)).
NOTE 14: IP LA equals to "19.65.9.11" selected by H.248 MG.
NOTE 15: IP LP equals to 49172 selected by H.248 MG for iLBC RTP packets.
NOTE 16: Codec "iLBC" chosen by H.248 MG.
NOTE 17: IP LP equals to 49174 selected by H.248 MG for RFC 4733 [22] packets.
NOTE 18: The attribute can be removed from the H.248 Local/Remote descriptor as the transport protocol in the m-line is RTP/AVP, which denotes RTP (RFC3550 [24]) used under the RTP Profile for Audio and Video Conferences with Minimal Control (RFC3551 [25]) running over UDP and therefore the meaning of payload type 0 is defined without any ambiguity.
NOTE 19: The MGC may insert a FQDN in the "c= " line. This does not imply that the MG must resolve the symbolical address into a numerical IP address (e.g. based on a DNS query). A resolution is only done on MG side when a resolved address is required.</td></tr>
</table>

Example illustrates again the basic SDP mapper functions beside the "negotiation aspect" here:

1. Usage of H.248 reserve properties in order to control/influence resource negotiation/determination (here see note 4, table 6.3.2.2.1).

2. Selection of the 2nd codec "iLBC" by MG because the 1st order codec is not supported in this scenario.

3. Mapping of (multiple) SIP/SDP "media streams" on (multiple) H.248 Streams.

4. Using single H.248 Streams for RTP and RTCP packet flows together. Alternatively could be individual H.248 Streams used.

5. Mapping of SIP/SDP attributes "recvonly", "sendrecv" and "sendonly" on correspondent H.248 StreamMode properties (e.g. a SIP/SDP "sendonly" must be inverted to a H.248 StreamMode "ReceiveOnly"). This function includes the deletion of correspondent "a=" line in the H.248/SDP descriptor, and insertion (if required) in the SIP/SDP descriptor respectively.

6. Format adaptation of "s=" lines, e.g., in case dedicated default specifications (e.g. by an H.248 Profile).

7. Format adaptation of "o=" lines: relevant is here the "o=" line usage at the SIP/SDP interface, i.e. the SDP mapper could replace a default "o=" line, as received from the MG, by another "o=" line towards SIP UA.

6.3.2.2.2 H.248 MG does support also G.711 (as Audio Codec)

This requires a rearrangement of the codec list in the H.248 ADD.request (2) in order to get SIP "answer" (4). See table 6.3.2.2.2.

Table 6.3.2.2.2: Re-Formatting SDP Elements: Example of Two Audio Streams

Example of two audio streams	
Reference:	see previous clause 6.3.2.2.1 table 6.3.2.2.1
Description:	see previous clause 6.3.2.2.1 table 6.3.2.2.1
Assumptions:	H.248 MG supported resource component types: support of all requested codec types
SIP/SDP ('A' side):	**H.248/SDP** ('B' side):

(1) – Incoming "**Offer**":
 	
see previous clause 6.3.2.2.1
	(2) – H.248 ADD request from MGC towards MG:
	`MEGACO/3 [11.9.19.65]:12345`
	`Transaction = 31205 {`
	`Context = $ {`
	`Add = $ {`
	`Media{`
	`Stream = 1 {`
	`LocalControl {`
	`Mode = SendReceive,`
	`ReservedGroup = OFF,`
	`ReservedValue = OFF},`
	`Local {`
	`v=…`
	`c=…`
	`m=audio $ RTP/AVP 97 0 ; NOTE 1`
	`a=rtpmap:97 iLBC/8000`
	`}`
	`Remote {`
	`v=…`
	`c=…`
	`m=audio 49170 RTP/AVP 97 0 ; NOTE 1`
	`a=rtpmap:97 iLBC/8000`
	`}}}`
	`Stream = 2 {`
	…
	}
	…}

(4) – Outgoing "**Answer**": see clause 6.3.2.2.1	(3) – H.248 reply from MG towards MGC: ...
NOTE: The list of codec types is now flipped because the MG must apply the H.248 reservation rule: "If ReservedGroup is "False " and ReservedValue is "False ", then … "The MG chooses the **first** alternative in Local for which it is able to support at least one alternative in Remote." (see paragraph 7.1.8/H.248.1). Note that the particular codec selection preference list hast to be known by the MGC/SDP mapper (e.g. by means of configuration management).	

6.3.3 End-to-end scenario with ES 129 163 call procedures

For further studies.

7 Mapping aspects between SDP versions

7.1 Introduction

SDP is still evolving, new RFCs will obsolete old RFCs. There is therefore a RFC-dependency concerning compatibility between SDP specifications, either on protocol level itself (e.g. RFC 2327 [7] vs RFC 4566 [1]), or concerning procedures (e.g. for "SDP capability negotiations" like offer/answer model).

This RFC-dependency affects each SDP interface itself, but also SDP mapping between SIP/SDP and H.248/SDP. For instance, when looking at the core SDP specification then there could be theoretically up to four mapping rules for the four combinations of {SIP/SDPv1 to H.248/SDPv1; SIP/SDPv1 to H.248/SDPv2; SIP/SDPv2 to H.248/SDPv1; SIP/SDPv2 to H.248/SDPv2}.

The RFC-dependency should not be an issue for the very majority of interworking scenarios, but there are a few cases which need more consideration (e.g. in case of syntax changes or capability extensions).

This section provides some first guidelines (see next clauses).

7.2 High-level guidelines

There might be different possibilities:

1. "Full parser" method

   This means that the SDP decoder is able to parse all possible SDP information elements as defined by relevant SDP RFCs. E.g. such an SDP decoder implementation would understand both RFC 2327 [7] and RFC 4566 [1].

2. Provisioning of SDP support information

   The SDP mapper (see figure 5) could benefit from knowledge about the particular SDP support at the SIP/SDP and H.248/SDP interface. Such kind of information could be beneficial for an optimization of SDP interworking between both interfaces.

3. Dynamic auditing of SDP support information

   A more flexible method is supported for H.248/SDP interfaces by the capabilities defined by ITU-T Recommendation H.248.49. Appendix I/H.248.49 provides a comparison of SDP variants between RFC 4566 [1] and RFC 2327 [7].

4. Others.

7.3 Behaviour in case of "not supported SDP elements "

Not supported SDP syntax and/or information elements may basically lead to an "ignore" action, or a reply with an appropriate error code by the SDP receiving entity. Specific use cases are subject of SIP and H.248 profile specifications and are thus out of scope of the present document.

History

Document history		
V2.0.0	January 2008	Publication