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Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Emulation Services for PSTN Modem Calls

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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 framework of IP emulation services for PSTN modem call types. The present document provides, inter alia, a description of:

- the problem statement;
- summary of IP bearer services;
- overview of media-type configurations; and
- a collection of example network use cases;

for PSTN modem calls.

1.1 Conventions

1.1.1 SDP Offer/Answer protocol variants

The present document provides example signalling syntax. There are two models for the Session Description Protocol (SDP) concerning the indication and negotiation of media and transport capabilities:

- the name "*legacy* SDP Offer/Answer" indicates SDP Offer/Answer according RFC 3264 [i.1];
- the name "*revised* SDP Offer/Answer" indicates SDP Offer/Answer according RFC 5939 [i.4] and RFC MediaCapNeg [i.5].

The two SDP Offer/Answer *protocol variants* differ in terms of supported SDP *syntax*, but also in terms of negotiation logic on *semantical* level (see also Annex D).

1.1.2 Configuration (Codec) list

The term *media configuration* (briefly configuration) is more comprehensive than the term codec. A media configuration covers typically media type, media format, all media format attributes, media transport stack, media transport capacity and all associated parameter values.

The term *configuration list* represents consequently a list of configurations within the SDP *media description* block.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

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2.1 Normative references

The following referenced documents are necessary for the application of the present document.

Not applicable.

2.2 Informative references

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] IETF RFC 3264 (2002): "An Offer/Answer Model with the Session Description Protocol (SDP)".

[i.2] IETF RFC 4040 (2005): "RTP Payload Format for a 64 kbit/s Transparent Call".

[i.3] IETF RFC 4733 (2006): "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals".

[i.4] IETF RFC 5939 (2010): "SDP Capabiliy Negotiation".

[i.5] Draft IETF RFC MediaCapNeg (201X): "SDP media capabilities Negotiation", {draft-ietf-mmusic-sdp-media-capabilities}.

NOTE: *MediaCapNeg* is a draft version of IETF RFC, the reference will be updated when it is published formally.

[i.6] Draft IETF RFC MiscCapNeg (201X): "Miscellaneous Capabilities Negotiation in the Session Description Protocol (SDP)", {draft-ietf-mmusic-sdp-misc-cap}.

NOTE: *MiscCapNeg* is a draft version of IETF RFC, the reference will be updated when it is published formally.

[i.7] Draft IETF RFC ConnCapNeg (201X): "Session Description Protocol (SDP) Extension For Setting Up Audio Media Streams Over Circuit-Switched Bearers In The Public Switched Telephone Network (PSTN)", {draft-ietf-mmusic-sdp-cs}.

NOTE: *ConnCapNeg* is a draft version of IETF RFC, the reference will be updated when it is published formally.

[i.8] ITU-T Recommendation G.168: "Digital network echo cancellers".

[i.9] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

[i.10] ITU-T Recommendation V.150.1: "Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs".

[i.11] ITU-T Recommendation V.151: "Procedures for end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay".

[i.12] ITU-T Recommendation V.152: "Procedures for supporting voice-band data over IP networks".

[i.13] ITU-T Recommendation V.153: "Interworking between T.38 and V.152 using IP peering for realtime facsimile services".

[i.14] ITU-T Recommendation T.38: "Procedures for real-time Group 3 facsimile communication over IP networks".

[i.15] ETSI TS 123 228: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS); Stage 2 (3GPP TS 23.228)".

[i.16] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) (3GPP TS 24.229)".

[i.17] ETSI TS 126 114: "Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction (3GPP TS 26.114)".

[i.18] ETSI TS 129 238: "Universal Mobile Telecommunications System (UMTS); LTE; Interconnection Border Control Functions (IBCF) - Transition Gateway (TrGW) interface; Ix interface; Stage 3 (3GPP TS 29.238)".

[i.19] ETSI TS 129 332: "Universal Mobile Telecommunications System (UMTS); Media Gateway Control Function (MGCF) - IM Media Gateway (IM-MGW); Mn interface (3GPP TS 29.332)".

[i.20] ETSI TS 182 012: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation Sub-system (PES); Functional architecture".

[i.21] ETSI TS 183 002: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); H.248 Profile Version 3 for controlling Access and Residential Gateways".

[i.22] ETSI TS 183 018: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Resource and Admission Control: H.248 Profile Version 3 for controlling Border Gateway Functions (BGF) in the Resource and Admission Control Subsystem (RACS); Protocol specification".

[i.23] ETSI TS 183 036: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification".

[i.24] ETSI TS 183 043: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation; Stage 3 specification".

[i.25] ETSI ES 283 003: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 [Release X], modified]".

[i.26] ETSI ES 283 049: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); H.248 Profile for controlling Trunking Media Gateways (TMG) [Endorsement of 3GPP TS 29.332 (V7), modified]".

[i.27] ETSI TS 129 231: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Application of SIP-I Protocols to Circuit Switched (CS) core network architecture; Stage 3 (3GPP TS 29.231 version 9.0.0 Release 9)".

[i.28] IETF RFC 3388: "Grouping of Media Lines in the Session Description Protocol (SDP)".

[i.29] IETF RFC 3407: "Session Description Protocol (SDP) Simple Capability Declaration".

[i.30] Draft ITU-T Recommendation H.248.80: "Gateway control protocol: Usage of the revised SDP offer/answer model with H.248".

NOTE: This ITU-T Recommendation is still in work, the reference will be updated when it is published formally.

[i.31] ITU-T Recommendation Q.115.0: "Protocols for the control of signal processing network elements and functions".

[i.32] ETSI ES 283 012 (V2.1.1): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking; Trunking Gateway Control Procedures for interworking between NGN and external CS networks [Endorsement of 3GPP TS 29.412 (R8)]".

[i.33] ITU-T Recommendation H.248.31: "Gateway control protocol: Adaptive jitter buffer package".

[i.34] IETF RFC 3362 (2002): "Real-time Facsimile (T.38) - image/t38 MIME Sub-type Registration".

[i.35] TISPAN Temporary Document 08TD241 (2005-09): "H.248 Trunking GW Profile - Voiceband Data (VBD) Service".

[i.36] TISPAN Temporary Document ACWG3-H248 TD08 (2005-11): "H.248 Trunking GW Profile - Progress on clause 5.17.2.3 VBD Procedures".

[i.37] ETSI TS 124 292 (V9.3.0): "Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Core Network subsystem Centralized Services; Stage 3 (3GPP TS 24.292 Release 9)".

[i.38] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163)".

3 Definitions and abbreviations

3.1 Definitions

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

PSTN modem call: *voiceband data* call originating/terminating in a PSTN domain

NOTE: The term *voiceband data* (VBD) is an umbrella term for all kind of *teleservices* which using a "data-oriented transport" in the frequency band of the narrowband voice spectrum (which is a 3,1-kHz-band). The data-oriented transport is realized by *modem* protocols (definition as in clause 3.13 of ITU-T Recommendation V.152 [i.12]), as defined e.g. within the ITU-T V.x-series of Recommendations. Teleservices may be categorized into three major applications areas: facsimile, text-based communication and general data services.

XoIP emulation service (for PSTN modem calls): emulation service in IP networks, based on appropriated gateway technologies for interworking voiceband data information between the PSTN and IP networks

NOTE: Example emulation services for the three main VBD application areas, which may be summarized as (by using notation "application/transport"):

- **Facsimile/modem:** Gateway technologies for PSTN-to-IP interworking see e.g. ITU-T Recommendation V.152 [i.12] for *pass-through* mode and ITU-T Recommendation T.38 [i.14] as *packet-relay* mode;
- **Text/modem:** Gateway technologies for PSTN-to-IP interworking see e.g. ITU-T Recommendation V.152 [i.12] for *pass-through* mode and ITU-T Recommendation V.151 [i.11] as *packet-relay* mode; and
- **Data/modem:** Gateway technologies for PSTN-to-IP interworking see e.g. ITU-T Recommendation V.152 [i.12] for *pass-through* mode and ITU-T Recommendation V.150.1 [i.10] as *packet-relay* mode.

3.2 Abbreviations

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

ACL	Answered Configuration (Codec) List
AGCF	Access Gateway Control Function
AGW	Access GateWay
ARGW	Access Residential media GateWay
AS	Application Server
B2BUA	Back-to-back User Agent (SIP)
BICC	Bearer-Independent Call Control
CMD	Clearmode
CMDoIP	Clearmode over IP (RFC 4040 [i.2])
CSCS	Call/Session Control Server
DSP	Digital Signal Processor
EC	Echo Canceller
FoIP	Facsimile over IP (T.38)
G3FE	Group 3 Facsimile Equipment
GW	Gateway
IAF	Internet-aware Fax device (T.38)
IBCF	Interconnect Border Control Function
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
JBIG	Joint Bi-level Image Experts Group (T.38 coding scheme)
MG	Media Gateway (H.248)

MGC	Media Gateway Controller (H.248)
MMR	Modified Modified Read (T.38 coding scheme)
MoIP	(Data) Modem over IP (V.150.1)
NCL	Negotiated Configuration (Codec) List
NNI	Network Network Interface
NTE	Network Telephony Event
NTEoIP	Network Telephony Event over IP (RFC 4733 [i.3])
O/A	(SDP) Offer/Answer
OCL	Offered Configuration (Codec) List
PBX	Private Branch Exchange
PCL	Preferred Configuration (Codec) List
PES	PSTN Emulation Subsystem
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RGW	Residential GateWay
RTP	Real Time Protocol
SCL	Supported Configuration (Codec) List
SDP O/A	SDP Offer/Answer
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TGW	Trunking GateWay
ToIP	Text over IP (V.151)
TrGW	Transition Gateway (3GPP)
UA	User Agent (SIP)
UNI	User Network Interface
VBD	Voiceband Data
VBDoIP	Voiceband Data over IP (V.152)
VGW	Voice Gateway (SIP)
VoIP	Voice over IP
XoIP	X over IP (with 'X' as placeholder for "IP application protocol 'X'")

3.2.1 Reference Points

For the purposes of the present document, the following network reference points are referred:

- TISPAN specific reference points: Ia, P1.
- Common reference points of TISPAN and 3GPP: Gm, Ici, Nb, Nc, Mn, Mp, Mw.
  See also clause 3.2 in TS 123 228 [i.15] with regards to the 3GPP defined reference points.

4 Problem Statement

There is a pending issue with *voiceband data (VBD) calls* in IMS, PES and IMS-based PES with regards to the unambiguous indication and negotiation of possible bearer emulation services in IP networks. The term voiceband data (VBD) originates from data traffic using a modem-based transport via the voice frequency spectrum of a PSTN voice line. VBD calls represent all PSTN modem call types, i.e. the three main categories of *fax/modem*, *text/modem* and *data/modem* calls, which are originating or terminating in a PSTN/ISDN domain.

The problem of the *emulation* of PSTN modem calls via IP network (thus, an *IP bearer emulation service*) is well-known, because there are in the meanwhile experiences of approx. 10 years of network operation -, but this difficulty is not yet solved for all cases. There are still ongoing issues notified in the daily process of VoIP network operation. This challenge is controllable in specific scenarios like private VoIP networks (enterprise solutions), single-vendor public VoIP networks or/and markets with sufficient information about all legacy modem techniques which are still in service.

That specific rationale was/is the root cause why the problem was so far disregarded in existing TISPAN releases. Other reasons coming from the different handling of modem calls in mobile and fixed networks, which preventing straightforward solutions for converged NGN/IMS infrastructures (due to the different handling of a terminating modem call in a mobile terminal). It was in the meanwhile also observed, that not every implementation of VBD IP emulation services is fully compliant to the underlying standardized technologies, or/and sufficiently tested against the universe of legacy, often market-specific, often still proprietary modem types. And there is also insufficient commitment to feed back experiences in the standardization/maintenance process for enhancing the aimed technologies (like e.g. V.15X-series, T.38, etc.), knowing that already a lot of PSTN experts and modem expertise is just gone ("and that loss of knowhow implies a difficult and expensive reverse engineering process when trying to fix PSTN / modem problems").

However, the technologies for VBD IP bearer emulation services may be not blamed for that situation, the problem of PSTN modem calls in IMS & VoIP NGNs is primarily still an implementation and interoperability issue (see e.g. also *SIP Forum* activities to fix fax/modem calls: FoIP task group http://www.sipforum.org/content/view/310/252/) between PSTN-to-IP gateways (or IP endpoints). Any interoperability problem between two peering gateways should be firstly addressed by:

1) sufficient and unambiguous specification of the aimed *media configuration* for a VBD IP emulation service; and

2) the explicit *declaration* and *negotiation* between the involved parties.

Item #1 may be addressed by *explicit* usage of V.152 [i.12] or a packet relay technique like T.38 [i.14] for fax/modem calls ("but then *all* T.38 parameters are provided as defined in T.38 [i.14]").

Item #2 relates to the *negotiation* (and re-negotiation) process on call/session control level, thus subject of SIP and the embedded SDP Offer/Answer protocol in IMS and IMS-based PES networks. A sufficient SDP O/A negotiation is often lacking today (due to the implicit assumption of partially/entirely expected media configurations (defaults) of the remote side; or/and rudimentary support of SDP O/A).

The present document is not providing any explicit change indications for TISPAN specifications. The purpose of the present document is rather a summary of the problem statement and the indication of possible changes in order to relax this dilemma.

The problem may be introduced and studied at the simplest network scenario of a pure IMS, single domain configuration, see next clause.

5 Example scenario as introduction

Figure 1 illustrates the example of a single network domain for IMS and IMS-based PES. The outlined problem should be firstly solved for the simple case of such intra-IMS calls before looking at interconnection scenarios of multiple domains or private-to-public configurations.

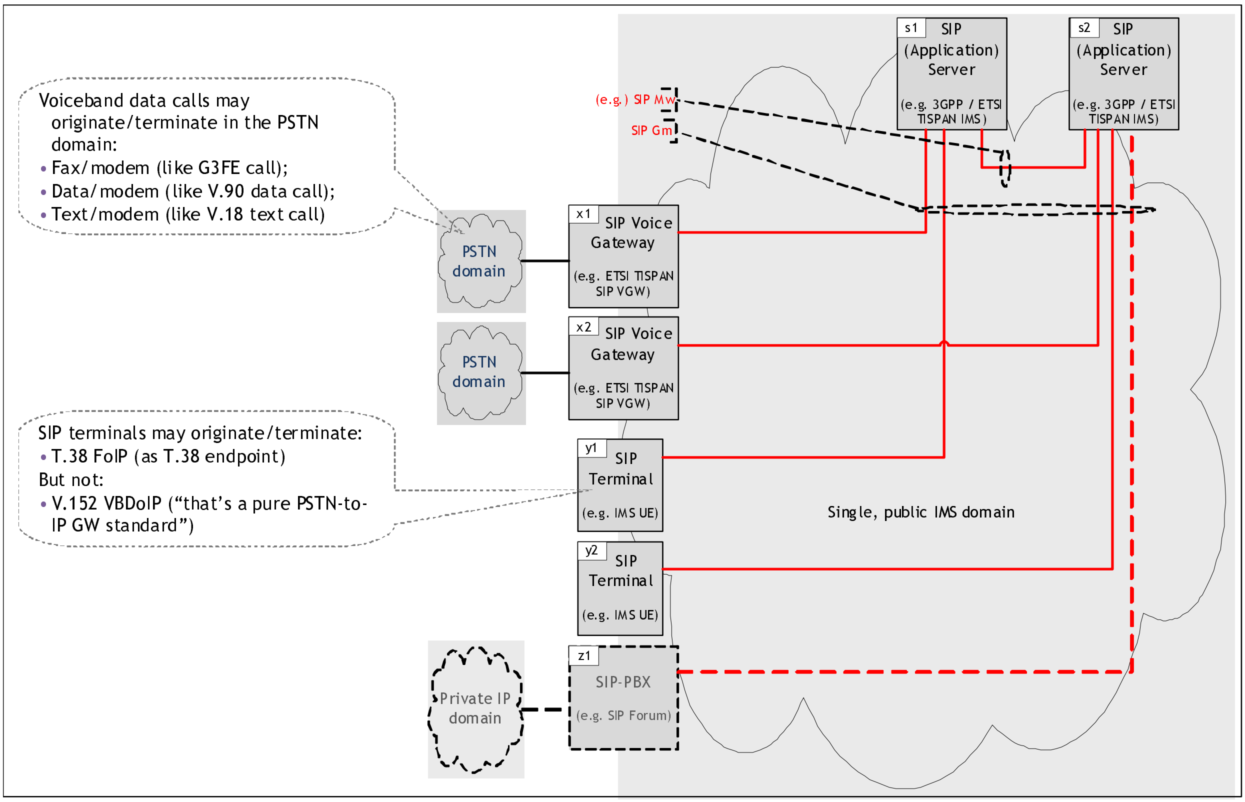

Figure 1: Mix of SIP VGWs (IMS-based PES) & SIP UEs (IMS)

A PSTN (access) domain is interconnected via SIP VGWs to the IMS domain in IMS-based PES solutions. PSTN modem calls originate/terminate in that PSTN domains (e.g. a SIP signalling session from x1 to x2 via s1 and s2). However, such a call may also terminate in an IMS UE (or even originate) if that IP terminal provides correspondent capabilities for handling modem-based VBD services (e.g. a SIP session from x1 to y2 via s1 and s2).

The focus is rather on PSTN traffic and not on ISDN, - due to the typical ratio of PSTN:ISDN traffic (in favour of PSTN) and the lacking capability of PSTN signalling with regards to the indication of a "PSTN bearer service", (see note).

NOTE: ISDN is different, e.g. a call originating from a G3FE looks like a speech telephony service to the network, but an ISDN fax terminal (G4FE) could provide an explicit indication for a fax service.

Further: there is typically following ratio between speech telephony and modem calls: "PSTN speech" >> "PSTN modem". The PSTN modem calls may be furthermore classified in *three types* (note: such a categorization is also justified by the availability of correspondent IP packet relay techniques (like T.38 FoIP, V.151 ToIP, V.150.1 MoIP)): Figure 2 provides an example *distribution* for PSTN modem calls. Such a distribution is primarily driven by the installed base of terminal equipment for the particular modem calls.

The (estimated) traffic profile of PSTN modem calls is significant for any IMS deployment. For instance, the SDP Offerer is in charge of providing:

a) appropriate media configurations for all modem types; and

b) fall-back media configurations in case that specific IP bearer emulation services may not be supported by the peering side.

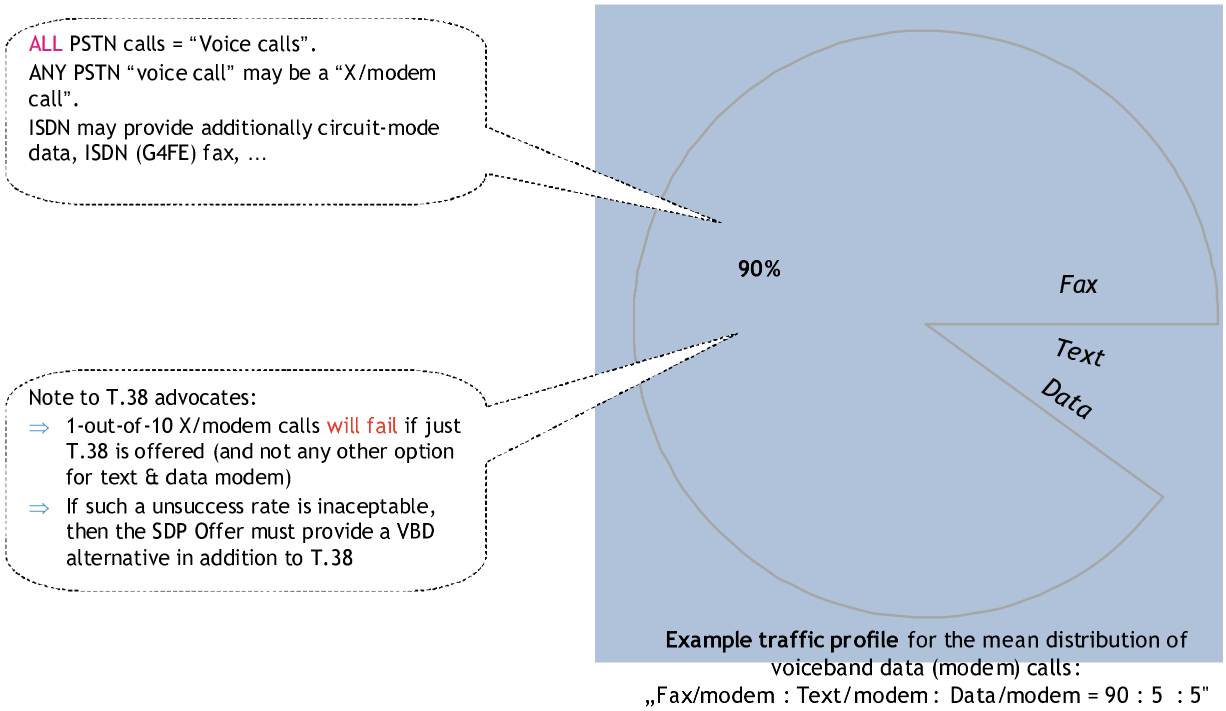

Figure 2: Traffic Profile for PSTN Modem Calls (example)

6 XoIP bearer services for emulating PSTN modem calls - Existing technologies and recommendations

There is a single "pass-through" technology defined, VBDoIP according ITU-T Recommendation V.152 [i.12], for *all three* PSTN modem call categories. V.152 may be thus used in general, and should be also the fall-back option if a packet-relay technique is not supported / not wished / etc.

ITU-T Recommendations T.38 [i.15], V.151 [i.12] and V.150.1 [i.11] are the correspondent packet relay methods for FoIP, ToIP and MoIP. Media configurations for packet relay operations are much more expensive than VBDoIP V.152 due to the significant difference of required resources in terms of DSP processing time and memory for the interworking function.

The focus may be therefore just on V.152 and T.38 for above outlined example traffic profile of PSTN modem calls (e.g. such a distribution would not justify any implicit support of V.150.1 per se).

Any network solution needs also to consider pre-V.152 VBDoIP support because V.152 was just published during TISPAN R1 timeframe, thus is not yet integral part of all R1 specifications. The mode may be labelled as "pseudo-VBD" (pVBDoIP) because:

> The SDP Offerer is unable to provide *explicit* signalling elements (like V.152) for the VBDoIP service, and thus merging the "audio mode" and "VBD mode" in a single mode, called pseudo-VBD mode. Such a single mode requires a configuration which allows to transport audio and VBD signals with the same media (codec) configuration. pVBDoIP is therefore typically using G.711 [i.9] without silence suppression, without adaptive Jitter Buffer control, without gain control, without noise reduction, … overlaid by a G.168 [i.8] compliant echo canceller (EC).

Pseudo-VBD is further even proprietary in the context of SIP VGWs (due to a lacking specification of this gateway mode in earlier TISPAN releases (R1, R2) like in contrary to some H.248 profiles; note: the situation improved with TISPAN R3 by an explicit definition of a "non-V.152" mode in TS 183 043 R3 [i.24]).

6.1 Indication and Negotiation of Media Configuration(s)

The PSTN-IP gateways (here SIP VGW), (see note 1), of the example IMS domain support V.152 and T.38, and the IMS UE could provide support for T.38 (as Internet-aware fax endpoint).

NOTE 1: Out of scope here: other SIP-controlled PSTN-IP gateways like SIP analog terminal adaptors (ATA) or SIP PBXs.

All SIP UA entities involved in the SDP O/A process consequently support the *SDP information elements* for V.152 and T.38 for the *explicit indication* and *negotiation* of media configurations. (Again the note that a *complete set* of SDP elements (all attributes) is needed as specified in T.38 [i.14], and not just a sub-set.)

Further, SDP O/A provides a legacy negotiation protocol according to RFC 3264 [i.1], but also enhancements by latest IETF updates (sometimes called "revised SDP O/A"). The negotiation process could benefit from that extended negotiation protocol, particularly if a decision between a packet-relay and pass-through mode is envisioned, or/and V.152 would be considered in sub-modes (e.g. V.152 non-assured vs V.152 assured transport modes), or/and backward compatibility to non-V.152 implementations needs to be solved, or/and support of multiple T.38 transport modes, or/and support of multiple T.38 assurance levels (FEC, redundancy), or/and etc.

Multiple media configuration options and a negotiation via revised SDP O/A may be a significant step forward in relaxing the interoperability problem in heterogeneous, multi-vendor deployment scenarios.

The IMS SIP profile is already providing support of revised SDP O/A protocol elements, see ES 283 003 [i.25] and the underlying TS 124 229 [i.16] ("SDPCapNeg support since 3GPP R7 and MediaCapNeg since 3GPP R9").

NOTE 2: The *full* negotiation of media configurations for V.152, T.38, etc. via revised SDP O/A requires both: RFC 5939 [i.4] and MediaCapNeg [i.5].

Table 1 on SDP support is derived from what precedes.

Table 1: SDP Negotiation Capability support versus SDP V.152/T.38 support - Matrix of SIP device specific options

SDP compatibility of SIP product implementation				Negotiation Capabilities	
				Support of *Legacy* SDP O/A (RFC 3264)	Support of *Revised* SDP O/A („SDPCapNeg“, „MediaCapNeg“, ...)
SDP for **V.152**	N	SDP for **T.38**	N		
			Y		
	Y	SDP for **T.38**	N		
			Y		

The SDP capabilities of a particular implementation needs to be considered by a service provider concerning his codec preferrences.
The SDP capabilities affecting the construction of an OCL and negotiation.

The outlined status of supported SDP capabilities by a SIP UA and B2BUA instance in an IMS domain is considered in the SDP O/A negotiation process.

Note to *packet relay* methods (like T.38 FoIP):

The VBD packet relay traffic flow may either:

a) re-use the IP transport connection of the voice flow ("replaced voice"); or

b) use a separate, dedicated IP transport connection ("muted voice").

Both approaches are justified from application perspective (i.e. there might be different PSTN teleservices like a "fax-only call" versus an "alternate speech-fax call").
However, independent of the selected transport method, an explicit indication and negotiation between both parties is necessary.

6.2 User plane - Handling of signal processing functions

There are a number of signal processing functions with respect to the "voiceband" traffic: echo control (echo cancellation, echo suppression), noise reduction, automatic level (gain) control, etc., which may be located at the border of PSTN-IP networks, embedded in VoIP gateways (see clause 6.2 in ITU-T Recommendation Q.115.0 [i.31]).

Such kind of voiceband related signal processing functions may inherently interact with voiceband data traffic. However, possible interactions are already clarified by ITU-T Recommendations and/or ETSI TISPAN specifications.

The impact of echo control (EC) and JB (jitter buffer) control is further outlined in following clauses.

6.2.1 GW behaviour (by SIP VGW, H.248 MG)

V.152 compliance guarantees correct EC and JB handling because clause 6/V.152 defines JB freezing and clause 6.2/V.152 defines "EC and VBD mode" behaviour.

This might be an issue in non-V.152 gateways if not a similar behaviour would be supported as defined by V.152.

6.2.2 GW control of EC and JB (in SIP VGW, H.248 MG)

6.2.2.1 H.248 MG

6.2.2.1.1 Echo control

The H.248 protocol provides explicit signalling capabilities, e.g. the *tdmc/ec* property is used for EC control with respect to "half-way EC" types, located at "near-end side" (i.e. PSTN domain). The EC control differs slightly between VoIP and VBDoIP: the MGC could provide a strict EC control in case of VoIP, however, the H.248-signalled EC settings would be overruled by V.152 behaviour in case of an autonomous transitioning to VBDoIP by the MG ("V.152 takes precedence over H.248 here"). Consequently in case of a transition back to VoIP the H.248-signalled EC settings will be re-applied.

Correct EC behaviour might be an issue for non-V.152 gateways.

6.2.2.1.2 Jitter buffer control

The H.248 protocol defines also means for JB control like e.g. the Adaptive Jitter Buffer package according ITU-T Recommendation H.248.31 [i.33]. However, ETSI TISPAN could not identify any demand for additional support of H.248.31. It is implicitly expected that JB behaviour in case of VBD traffic follows the guidelines of V.152.

6.2.2.2 SIP VGW

6.2.2.2.1 Echo control

SIP/SDP lacks signalling capabilities for echo control. SIP VGWs should be V.152 compliant in order to ensure correct EC behaviour in case of VBD.

6.2.2.2.2 Jitter buffer control

SIP VGWs should be V.152 compliant in order to ensure correct JB behaviour in case of VBD detection.

6.2.3 Call control level: indication of EC towards "remote side"

There is usually not any need for EC indications across IMS via call control signalling (SIP) due to the type and location of ECDs at the border between PSTN-IMS, see clause 4.3 in ES 283 012 [i.32]. This covers both PSTN UNI to IMS and PSTN NNI to IMS scenarios. The G.168 "cancelled end" (also known as "near end") is subject of the PSTN segment. EC control is consequently subject of a local decision by the call control instance.

However, there might be more complex end-to-end scenarios like e.g. a chaining of PSTN (UNI|NNI) to IMS to PSTN NNI to IMS to PSTN (NNI|UNI) again, i.e. interim PSTN domains. Such scenarios may raise the question of coordinated EC control across all involved PSTN-IMS gateways, e.g. whether IMS call control signalling should provide support of EC indication and control towards remote gateway control instances. Such an evaluation and correspondent stage 2 work is beyond the scope of the present document. However, compliance to G.168 and V.152 by all PSTN-IP gateways in the end-to-end path should be sufficient for end-to-end PSTN modem calls (due to above outlined GW behaviour).

7 Media-type Configurations for PSTN modem calls

7.1 Overview

Any SDP *Offered Configuration List* (OCL; briefly "*Offered Codec List*") for PSTN modem calls provides *audio* support. There might be also packet relay support for "auxiliary information" like inband application control, modem signals etc. (via RFC 4733 [i.3] RTP packet types). A more complete view of *example* media-type/-format specific configurations is illustrated in Figure 3.

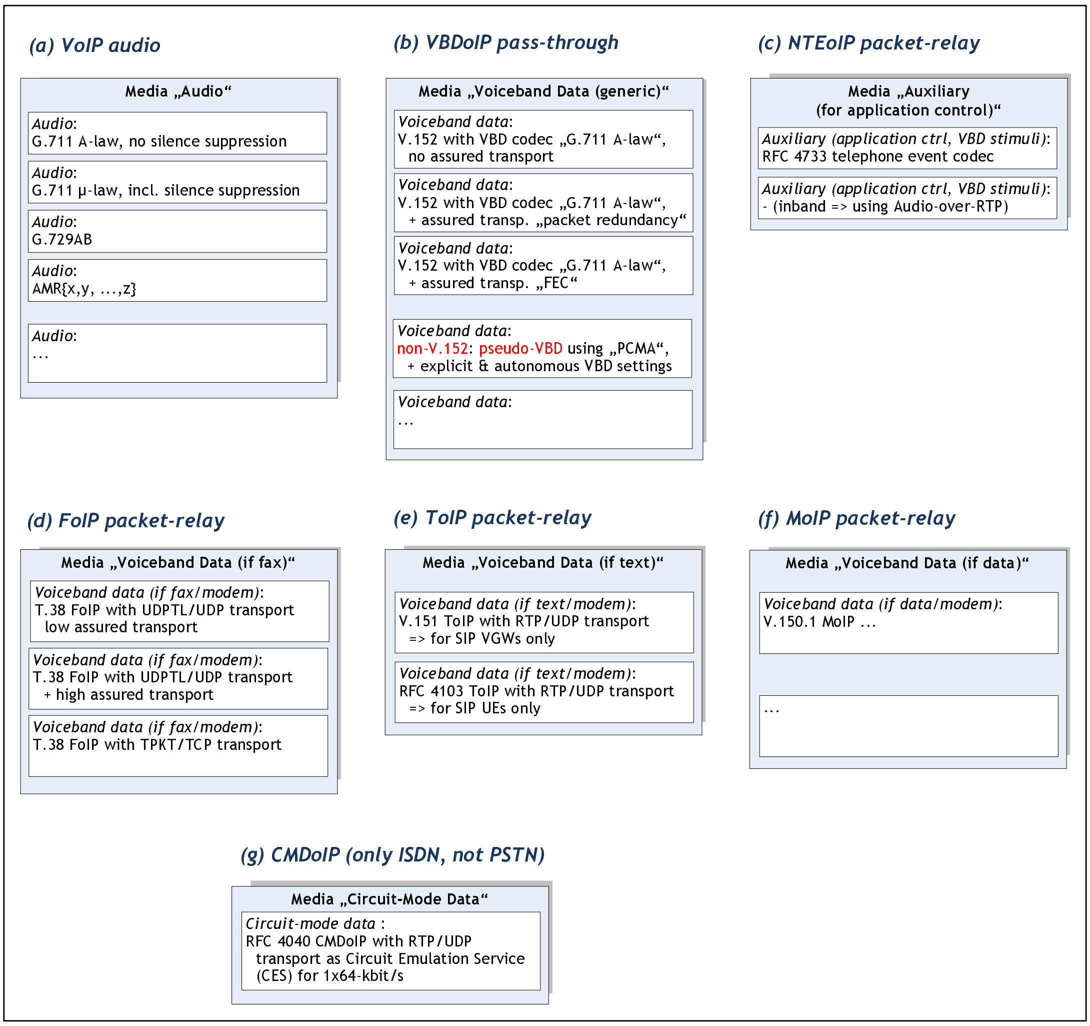

Figure 3: Sets of example Media-type specific Configurations

7.1.1 VBDoIP-type specific Configurations

VBDoIP pass-through, V.152 specific configurations are characterized by following *major* capabilities:

1) V.152 VBD codec type; and

2) V.152 transport mode ("non-assured" versus "assured" types).

7.1.2 FoIP-type specific Configurations

7.1.2.1 Major Configurations

FoIP packet relay, T.38 specific configurations are characterized by following *major* capabilities:

1) T.38 protocol version ("there are five versions (0 to 4) up to now");

2) T.38 transport mode ("there are the three modes UDTPL/UDP, TPKT/TCP, RTP/UDP");

3) T.38 data rate management method;

4) T.38 error correction;

5) T.38 fax transcoding ('none', 'MMR' or 'JBIG'); and

6) T.38 supported modem (primarily due to V.34 versus non-V.34-G3FE device").

There are more T.38 configuration parameters, however, existing T.38 interoperability problems are primarily related to above listed items. It is recommended to explicitly indicate and negotiate at least these T.38 configuration parameters, - despite the fact of possible redundancy (NOTE 1) -, in order to maximize the likelihood of T.38 service delivery, which would then allow to optimize the successful T.38 call rate (per IMS domain; per T.38 domain).

NOTE: Example of possible redundancy: the signalling of T.38 versions 0, 1 or 2 would *implicitly* indicate the *non-support* of modem types V.33 and V.34. However, legacy T.38 endpoint implementations would be enforced to commit explicitly each individual T.38 configuration parameter (if explicitly requested by SDP offer), which may allow to enhance a reliable service.

7.1.2.2 Signalled versus provisioned T.38 Configurations

A default T.38 configuration could be provisioned in *all* T.38 entities of a particular T.38 domain, which would allow to omit the indication, signalling and negotiation of all T.38 configuration details between T.38 on-ramp- and off-ramp gateways or IAFs.

The provisioning approach however is not recommended, primarily due to:

- Default values for T.38 parameters were not entirely specified for T.38 versions 0 to 3, thus a potential source of ambiguity.

- It may be not excluded that different T.38 versions could use different default value settings.

- Horizon of a single T.38 domain may span multiple PES or/and IMS domains (e.g. T.38 endpoints located in different provider domains, and each provider has its own preferred T.38 default settings).

7.2 Preferred Configuration/Codec Lists (PCL)

7.2.1 Basic principles

The originating SIP UA (e.g. SIP VGW where the PSTN modem call enters the IMS domain) constructs an (application-specific) *Offered Configuration List* (OCL), based on supported SDP capabilities, based on SDP O/A protocol support, based on VBD IP emulation service support, based on provider preferences, based on knowledge about capability support by peer devices, or/and based on present IP network conditions, or/and based on QoS objectives, or/and etc.

NOTE: It may be supposed (due to simplicity) that the considered SIP endpoints recognize the *preferences* of the network operator / service provider: which allows then to equate the OCL with the PCL (*Preferred Configuration List*). The sent Offer by the originating SIP UA represents thus already the PCL.

Table 2 shows some examples for PCLs for some of the outlined conditions. One of these conditions is related to the preferences for audio signals: assumption here to media format G.729AB first and G.711 [i.9] as secondary audio codec.

Revised SDP O/A provides the advantage of the declaration (and negotiation) of *potential configurations*. That is an essential feature for the negotiation of alternatives, which is required when trying to tackle the issue of PSTN modem calls.

Table 2: Some examples for Preferred Configuration/Codec Lists

SDP compatibility of SIP product implementation				language & intelligence for SDP-level negotiations	
				Support of *Legacy* SDP O/A (RFC 3264)	Support of *Revised* SDP O/A („SDPCapNeg“, „MediaCapNeg“, ...)

SDP for V.152	N	SDP for T.38	N	PCL = {
				G.711 in pseudo-VBD for combined “Audio & VBD”,
				no T.38
				}
					PCL = {
					p1: G.711 in pseudo-VBD for combined “Audio & VBD”,
					no T.38;
					p2, p3, ...: ...
					}

			Y	PCL = {
				G.729AB | G.711 Audio, T.38oUDPTL/UDP
				}
					PCL = {
					p1: G.729AB | G.711 Audio, T.38oUDPTL/UDP;
					p2, p3, ...: ...
					}

	Y	SDP for T.38	N	PCL = {
				G.729AB | G.711 Audio, V.152 VBD, no T.38;
				}
					PCL = {
					p1: G.729AB | G.711 Audio, V.152 VBD, no T.38;
					p2, p3, ...: ...
					}

			Y	PCL = {
				G.729AB | G.711 Audio, T.38oUDPTL/UDP
				}
					PCL = {
					p1: G.729AB | G.711 Audio, V.152 VBD, no T.38;
					p2: G.729AB | G.711 Audio, T.38oUDPTL/UDP
					p3, p4, ...: ...
					}

Note 1: all Offers for VoIP shall provide „RFC 4733 telephone event codec“ for auxiliary media components.					
Note 2: assumptions => no occurence of text/modem & data/modem; but if yes, then re-negotiation ...					

7.2.2 Signalling aspects

Configuration preferences in signalling messages could be indicated implicitly by an ordering principle (e.g. list) or explicitly by a correspondent attribute. Following mechanism is available for SIP/SDP signalling:

1) SDP Offer/Answer protocol for SIP:

- "*legacy* SDP Offer/Answer": indication of preference by list order, - at the level of SDP media descriptions and at the level of media formats within a media description;

  Reference: clause 5.1 of RFC 3264 [i.1]: [...] `In all cases, the formats in the "m=" line MUST be listed in order of preference, with the first format listed being preferred. In this case, preferred means that the recipient of the offer SHOULD use the format with the highest preference that is acceptable to it.`

- "*revised* SDP Offer/Answer": explicit indication by numerical value assigned to potential configurations and session configurations;

2) SDP extensions, orthogonal to Offer/Answer protocol:

- V.152 defined SDP attribute "`a=pmft:`" as "*indicating preference of VoIP relay mechanisms above VBD*", thus applicable for T.38 FoIP, V.151 ToIP and V.150.1 MoIP together with V.152 VBDoIP related media descriptions.

Discussion:

- the V.152 SDP "`a=pmft:`" method is superfluous in case of "*revised* SDP Offer/Answer", but essential for "*legacy* SDP Offer/Answer";
- the V.152 SDP "`a=pmft:`" codepoint is not (yet) registered with IANA;
- "*legacy* SDP Offer/Answer" with V.152 SDP "`a=pmft:`" attributed media descriptions leads to a combination of two preference indication methods, which needs to be carefully applied on Offerer and Answerer side in order to avoid misinterpretations.

A general, explicit and unambiguous preference scheme is provided by "*revised* SDP Offer/Answer", which allows also to deprecate the usage of the V.152 SDP "`a=pmft:`" attribute. Recommendations are summarized in clause 8.

7.3 Interoperability check

The various PCLs from Table 2 could be sent by the SDP Offerer, dependent on the depicted local conditions, but potentially also due to remote or network wide conditions. However, each of the outlined example PCLs provides its inherent merits and shortcomings. Some high-level conclusions (not exhaustive) may be already derived, see Table 3.

Table 3: Interoperability check - High-level conclusion

SDP compatibility of SIP product implementation				language & intelligence for SDP-level negotiations: Support of *Legacy* SDP O/A (RFC 3264)	language & intelligence for SDP-level negotiations: Support of *Revised* SDP O/A („SDPCapNeg“, „MediaCapNeg“, ...)
SDP for V.152	N	SDP for T.38	N	PCL = { G.711 in ... VBD", no T... } **issue** (pseudo-VBD)	PCL = { p1: G ... BD", p2, p } **issue** (pseudo-VBD in actual config.; alternatives in pcfgs ...)
			Y	PCL = { G.729 } **issue** (settings for non-fax modem?)	PCL = { p1: G. p2, p } **ok** (potential config. for non-fax modem may be included)
	Y	SDP for T.38	N	PCL = { G.729 } **ok** (V.152 support)	PCL = { p1: G. p2, p } **ok**
			Y	PCL = { G.729 } **issue** (settings for non-fax modem?; V.152 may be re-negotiated)	PCL = { p1: G. p2: G. p3, p4 } **ok** (due to potential config. ...)
Note 1: all Offers for VoIP shall provide „RFC 4733 telephone event codec“ for auxiliary media components.					
Note 2: assumptions => no occurence of text/modem & data/modem; but if yes, then re-negotiation ...					

Colour code: tackling the interoperability problem for PSTN modem calls should be easier with "green" PCLs, and is inherently more difficult with "red" PCLs.

7.4 Grouping of multiple media configurations

7.4.1 Introduction

A *single* PSTN modem call may require *multiple configurations* with different media type/format settings in 'parallel'. Typical examples are teleservices with alternate communication phases, like alternate speech-fax or text telephony. There are in general the two phases of *speech* and *X/modem* (e.g. *facsimile/modem* or *text/modem*; NOTE 1) at PSTN side. The two phases may be mapped on two separate XoIP emulation services on IP side (NOTE 2), like a VoIP RTP session for the speech phases and a FoIP UDPTL session (or ToIP RTP session) for *facsimile/modem* (or *text/modem*) transfer phases. There would be then two IP transport connections (for the different media type/format) in the IP network user plane, associated to a single SIP session in the control plane.

Such a PSTN modem call emulation needs two grouped media configurations.

NOTE 1: Alternate calls with *speech* and *data/modem* phases are not defined for PSTN.

NOTE 2: Using V.152 VBDoIP as alternative would lead to a single IP transport connection, carrying a single RTP session, carrying two media formats (discriminated by RTP PT value) for speech and VBD information. Thus, the application of V.152 would obsolete the need for "media grouping".

7.4.2 Signalling capabilities for grouped media configurations

7.4.2.1 Call control: SIP/SDP

There are two options:

1) RFC 3388 [i.28] defines the SDP attribute "`a=group:`", which may be used together with the *FID* semantics (for Flow IDentification) for media grouping.

2) "*revised* SDP Offer/Answer" provides the concept of *session configurations* (via SDP attribute "`a=sescap:`") as a binding element for multiple media configurations.

Discussion:

- the RFC 3388 [i.28] SDP "`a=group:`" method is superfluous in case of "*revised* SDP Offer/Answer", but essential for "*legacy* SDP Offer/Answer".

7.4.2.2 Gateway control: H.248

The H.248 protocol element *ReserveGroup* (on LocalControl descriptor) level provides explicit support for media grouping.

7.4.3 Interworking aspects

SIP Voice Gateways (VGW) may support different SDP capability sets (e.g. due to different TISPAN releases). Interworking aspects are discussed in clause C.1.

7.5 Declaration of dedicated media configurations

7.5.1 Introduction

There is a deficiency in "*legacy* SDP Offer/Answer" concerning the declaration of media configuration, see RFC 3407 [i.29]:

> "*For example, an endpoint may support G.711 audio (over RTP) as well as* ***T.38 fax relay*** *(over UDP or TCP). Unless the endpoint is willing to support two media streams at the same time, this cannot currently be expressed in SDP. Another example involves support for* ***multiple codecs.*** *An endpoint indicates this by including all the codecs in the "m=" line in the session description. However, the endpoint thereby also commits to simultaneous support for each of these codecs. …*"

The declaration of (latent) media configuration is an essential method for PSTN modem calls, like e.g. the declaration of configurations T.38 FoIP for *fax/modem* or V.151 ToIP for *text/modem* calls.

7.5.2 Signalling capabilities for declaration of media configurations

7.5.2.1 Call control: SIP/SDP

There are two options:

1) RFC 3407 [i.29] defines some SDP attributes as an extension of "*legacy* SDP Offer/Answer";

2) "*revised* SDP Offer/Answer" provides the concept of *potential configurations*, an inherent declaration mechanism.

Discussion:

- the RFC 3407 [i.29] defined SDP extensions are superfluous in case of "*revised* SDP Offer/Answer", but essential for "*legacy* SDP Offer/Answer".

NOTE: RFC 3407 [i.29] is not (yet) supported by any SIP profile (and H.248 profile) specification from ETSI TISPAN and 3GPP.

7.5.2.2 Gateway control: H.248

H.248 provides a master/slave resource reservation and allocation method, in contrast to the SIP/SDP client/server or peer-to-peer mode. The MGC may handle capability declarations and "latent configurations" (see also clause 7 in ITU-T Recommendation H.248.80 [i.30], which also indicates possible future extensions).

NOTE: H.248.80 is not (yet) supported by any H.248 profile specification from ETSI TISPAN and 3GPP. However, above paragraph is just referring to the analysis section in H.248.80, and not referring any H.248.80-defined protocol extensions.

7.6 Mode transitioning

7.6.1 Introduction

Each media configuration for the XoIP bearer endpoint represents a mode of operation. The top level modes are called *Audio* mode (VoIP), *VBD* mode (VBDoIP), *Fax relay* mode (FoIP), etc., see [i.10], [i.12] and [i.14]. An XoIP bearer connection endpoint with multiple modes enabled may be modelled as state machine. Any mode change represents then state transitioning behaviour.

7.6.2 Mode transitioning behaviour

There are two fundamental transitioning behaviour (see Table 4), which needs to be taken into account when considering emulation services for PSTN modem calls, particularly in heterogeneous network environments like e.g. IMS-based PES to PSTN UNI/NNI calls (see clause 9.4.1) with its mix of SIP- and H.248-controlled PES endpoints.

Table 4: Principle state transitioning behaviour

Mode:	Strict-controlled transitioning	Autonomous transitioning
Characteristics:	- Control plane signalling (e.g. application control protocols like SIP, gateway control protocols like H.248) is involved in state transitioning. - Point in time for state transition dependent on signalling scenario. - The two state machines of the two involved PES endpoints (GWs) are tightly coupled via network control plane.	- GW local decision for state transitioning, without any mandatory control plane event. - Fast state transitioning as soon as unambiguous detection done. - The two state machines of the two involved PES endpoints (GWs) are independent or loosely coupled (e.g. in case of RTP NTE stimuli) via network user plane only.

The preferences for the selected state transitioning behaviour are in general related to following key performance aspects:

1) processing load in network control plane: autonomous state transitioning allows to offload SIP servers (e.g. AS, x-CSCS, IBCF) and H.248 MGCs from signalling message processing and state switch transitioning logic;

2) user experience and call processing performance: state transitioning for PSTN modem calls implies realtime requirements due to the conversational nature of modem-based communication services.

7.6.3 Mode transitioning support for V.152 and T.38 emulation services

Support of different mode transitioning behaviour across the various control plane protocols (like e.g. SIP and H.248, but also H.323, MGCP, etc.) was not consistent (see Table 5).

Table 5: Mode transitioning support for V.152 and T.38 emulation services

Mode:	ITU-T Rec. V.152		ITU-T Rec. T.38	
	SIP-controlled V.152 gateway	H.248-controlled V.152 gateway	SIP-controlled T.38 gateway (or IAF)	H.248-controlled T.38 gateway
Strict-controlled transitioning	NO. Basically not aimed by V.152 (note 1)		YES. (see clause D.2.2.4.2/T.38)	YES. (see clause E.2.2.1/T.38)
Autonomous transitioning	YES	YES	NO (for T.38 2007 and before). YES (for T.38 after 2007 (note 2); see clause D.2.2.4.3/T.38)	YES. (see clause E.2.2.2/T.38; (note 3))

NOTE 1: The revision of V.152 by ITU-T may provide additional support of "strict-controlled transitioning", see clause 10.3/V.152.

NOTE 2: The latest revision of T.38 (which will be T.38 Version 4) by ITU-T defines "autonomous transitioning" for SIP, too. Which allows thus a facilitated operation between T.38 endpoints with autonomous transitioning, independent of SIP or H.248 control.
SIP VGWs: autonomous transition in case of SIP-controlled T.38 implies that both "controlling SIP UA" entities do support *revised* SDP Offer/Answer.

NOTE 3: H.248 GWs: autonomous transition in case of H.248-controlled T.38 implies support of the H.248 ReserveGroup property. This signalling capability is not (yet) supported by all relevant H.248 profiles from ETSI TISPAN and 3GPP.

Some conclusions:

- Autonomous transitioning looks more advantageous than strict-controlled transitioning, but the application depends on the network use case and/or involved control plane protocols inclusive their compliance against ITU-T Recommendation V.152 [i.12] and T.38 [i.14] publications.

- Autonomous transitioning is e.g. possible already in TISPAN R1 and R2 in PES-only scenarios (see clause 9.8).

- Strict-controlled transitioning was initially implemented in SIP VGW devices due to the missing definition by T.38 versions 0 to 3.

- Consistent autonomous transitioning is possible in future, when T.38 Version 4 would be supported by SIP, together with "*revised* SDP Offer/Answer" based negotiation of T.38 as *latent* configuration.

- Interoperation between a strict-controlled transitioning gateway and an autonomous transitioning gateway is fundamentally possible and not in contradiction:
  - the call control instance (e.g. SIP UA in a SIP server or SIP VGW, or MGC entity) knows (e.g. in case of H.248 gateways by the defined call-dependent procedures of the underlying H.248 profile) whether the associated V.152 / T.38 gateway instance is enabled for autonomous or strict-controlled state transitioning;
  - the autonomous transitioning gateway may be enabled for additional event reporting (of V.152 / T.38 stimuli) due to possible triggering of the peering strict-controlled transitioning gateway.

8 Requirements analysis & recommendations

Above high-level analysis may be summarized by following recommendations concerning the alleviation of interoperability issues:

i) Best case: SIP entity with support of:

  a) RFC 5939 (transport [i.4]) and MediaCapNeg (media [i.5]); and

  b) V.152;

ii) If one party is not supporting SDPCapNeg or MediaCapNeg, then V.152 offer should provide a VBD service with the lowest probability of potential interop issues.

iii) If one party is not supporting V.152, then T.38 may be tried.

iv) If one party is not supporting V.152, and if one party is not supporting T.38 or non-fax modem call, then pseudo-VBD should be tried, using G.711 [i.9] for both audio and VBD "as a single codec".

Such recommendations may be mapped on the following proposed stage 2 requirements. If those proposed requirements are later on included in a document containing normative provisions such as a Technical Specification or an ETSI Standard, the word "will" should be replaced by "shall".

1) Any SIP entity, involved in SDP Offer/Answer negotiations and/or the inspection of SDP payloads, WILL support RFC 3264 [i.1] and SHOULD support SDPCapNeg and MediaCapNeg.

2) Any SDPCapNeg/MediaCapNeg capable SIP entity WILL use SDPCapNeg/MediaCapNeg for generating Offers.

3) Any SIP entity WILL support SDP elements for V.152 VBDoIP service negotiations.

4) Any SIP entity WILL support SDP elements for T.38 FoIP service negotiations.

5) Any SIP entity MAY support SDP elements for V.151 ToIP service negotiations.

6) Any SIP entity MAY support SDP elements for V.150.1 MoIP service negotiations.

7) RFC 4733 [i.3] "telephone-event" codec WILL be supported in IMS (see TS 124 229 (3GPP 24.229) [i.16]).

Generation of *Preferred Configuration/Codec Lists* (for PSTN modem calls):

1) Configuration preferences are in general *conditional* due to the relation to policies (like operator preferences, see clause 7.2.1);

2) in case of lacking policies, then V.152 VBDoIP SHOULD be preferred over packet relay methods, due to the independence of the modem application;

3) usage of "*revised* SDP Offer/Answer" is recommended (due to the discussion in clause 7.2.2).

Grouping of multiple media configurations (if required for PSTN modem calls; see clause 7.4):

1) "*revised* SDP Offer/Answer" is recommended due to the embedded method for media grouping;

2) RFC 3388 [i.28] SDP "`a=group:`" method is deprecated due to inherent deficiencies for PSTN modem calls (see clause C.1.2).

Declaration of media configurations (as beneficial for PSTN modem calls with T.38 FoIP or V.151 ToIP emulation service; see clause 7.5):

1) "*revised* SDP Offer/Answer" is recommended due to the embedded method for *latent configurations*;

2) RFC 3407 [i.29] SDP extensions: would be required for "*legacy* SDP Offer/Answer".

Applied *mode transitioning* behaviour (for PSTN modem calls):

1) Autonomous transitioning is recommended (due to outlined benefit in clause 7.6.2) whenever possible;

2) Strict-controlled transitioning should be the exception, but often the only option like e.g. scenario #1 (clause 9.1) in pure TISPAN R1/R2 environments.

9 Use cases

This clause provides a non-exhaustive list of example use cases.

9.1 Scenario #1 - IMS-based PES scenario, intra-IMS call between two SIP gateways

The IMS-based PES processing model (see Figure 1 in TS 183 043 [i.24]) identifies two SIP gateways types:

- SIP VGW: the embedded SIP UA entity got an external SIP interface towards a P-CSCF, using SIP Gm signalling,
  clause 9.1.1 discusses a scenario between two SIP VGWs;

- SIP AGCF: the embedded SIP UA entity got an external SIP interface towards an S-CSCF, using SIP Mw signalling,
  clause 9.1.2 discusses a scenario between two SIP AGCFs.

9.1.1 Scenario #1.1 - Two SIP VGWs

The problem may be studied at the simplest network scenario of a pure IMS, single domain configuration. Figure 4 illustrates the example of a single network domain, here with just IMS-based PES gateways (x1, x2). The outlined problem should be firstly solved for the simple case of such intra-IMS calls before looking at interconnection scenarios of multiple domains or private-to-public configurations.

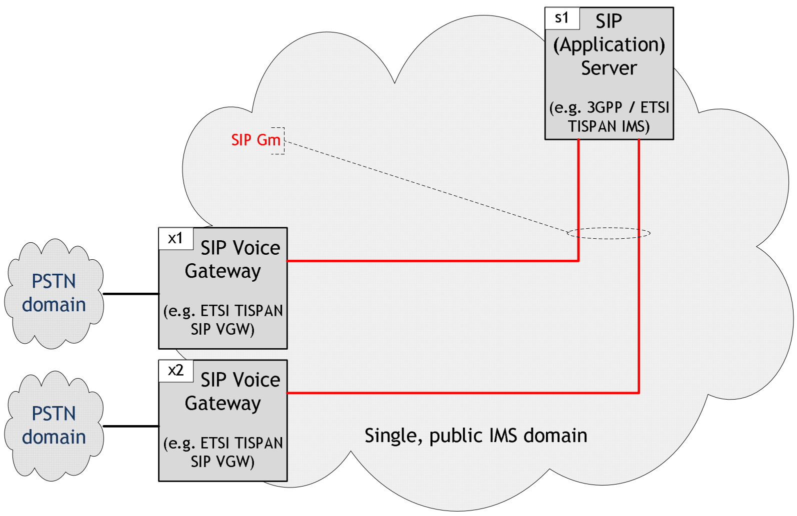

Figure 4: IMS-based PES scenario, intra-IMS call between two SIP Voice gateways (the SIP VGW is connected to a SIP server of type P-CSCF)

"Codec negotiation" (i.e. indication & negotiation of media & bearer configurations) in IP network:

- just SIP/SDP at Gm (not any other signalling interfaces involved).

9.1.2 Scenario #1.2 - Two SIP AGCFs

Figure 4a provides a scenario between two SIP AGCFs.

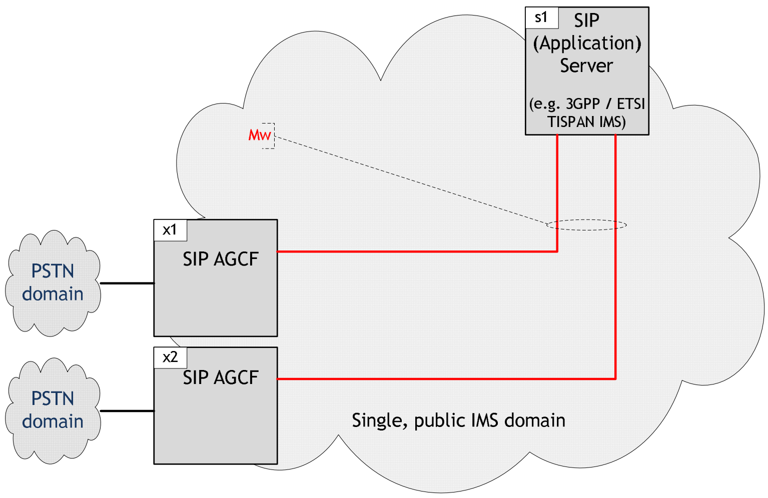

Figure 4a: IMS-based PES scenario, intra-IMS call between two AGCF (the SIP AGCF is connected to a SIP server of type S-CSCF)

The AGCF provides:

- SIP/SDP at Mw (for call control signalling with a S-CSCF); and
- H.248 for gateway control signalling between the AGCF embedded MGC entity and an external MGF.

9.2 IMS & IMS-based PES scenarios, general intra-IMS call

9.2.1 Scenario #2.1 - IMS & IMS-based PES scenario, general intra-IMS call

Figure 5 provides additional IMS equipment: e.g. because a T.38 call may origin/terminate in IP terminals directly. The peering to private IP networks via a SIP PBX (z1) is also indicated (but not discussed by this contribution).

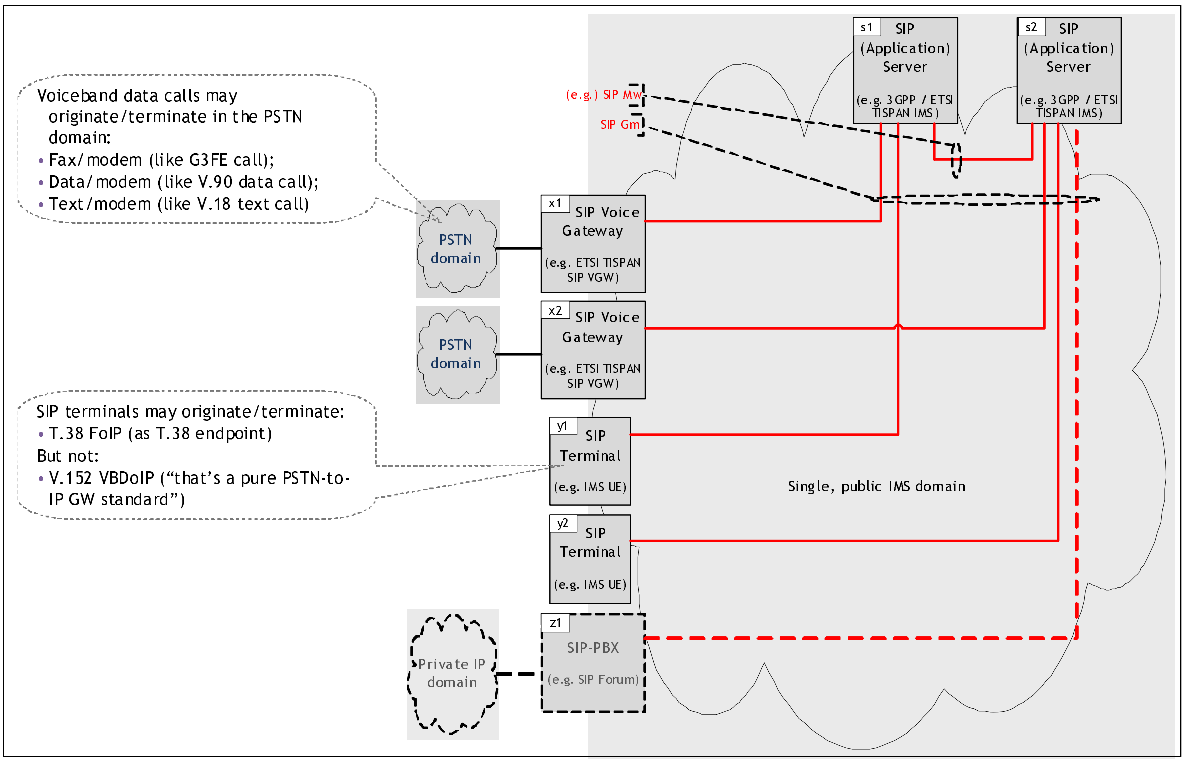

Figure 5: Mix of SIP VGWs (IMS-based PES) & SIP UEs (IMS)

A PSTN (access) domain is interconnected via SIP VGWs to the IMS domain in IMS-based PES solutions. PSTN modem calls originate/terminate in that PSTN domains (e.g. a SIP signalling session from x1 to x2 via s1 and s2). However, such a call may also terminate in an IMS UE (or even originate) if that IP terminal provides correspondent capabilities for handling modem-based VBD services (e.g. a SIP session from x1 to y2 via s1 and s2).

"Codec negotiation" (= indication & negotiation of media & bearer configurations) in IP network:

- other IMS SIP/SDP interfaces besides Gm.

9.2.2 Scenario #2.2 - IMS & IMS-based PES scenario, intra-IMS call, unsuccessful negotiation

Figure 6 depicts the case of possible unsuccessful negotiations (due to SCL limitations) between x1, y1 and t1.

EXAMPLE: Call between y1 (Offerer) and x1 (Answerer):

- device y1 provides T.38 IAF capability, but of course not any support for V.152 and non-V.152 VBD (due to IP terminal type)
  => SCL_{y1} = {VoIP audio codecs: a_1, … a_n | FoIP: T.38/UDPTL | VBDoIP: -}

- SIP VGW x1 (or H.248 ARGW t1 [i.21]) does not support T.38, but V.152 only
  => SCL_{x1} = {VoIP audio codecs: a_1, … a_m | FoIP: - | VBDoIP: V.152 PCMA, V.152 PCMU}

- conclusion: $NCL_{y1\text{-}x1}$ empty ($SCL_{y1} \cap SCL_{x1}$) for PSTN modem emulation traffic

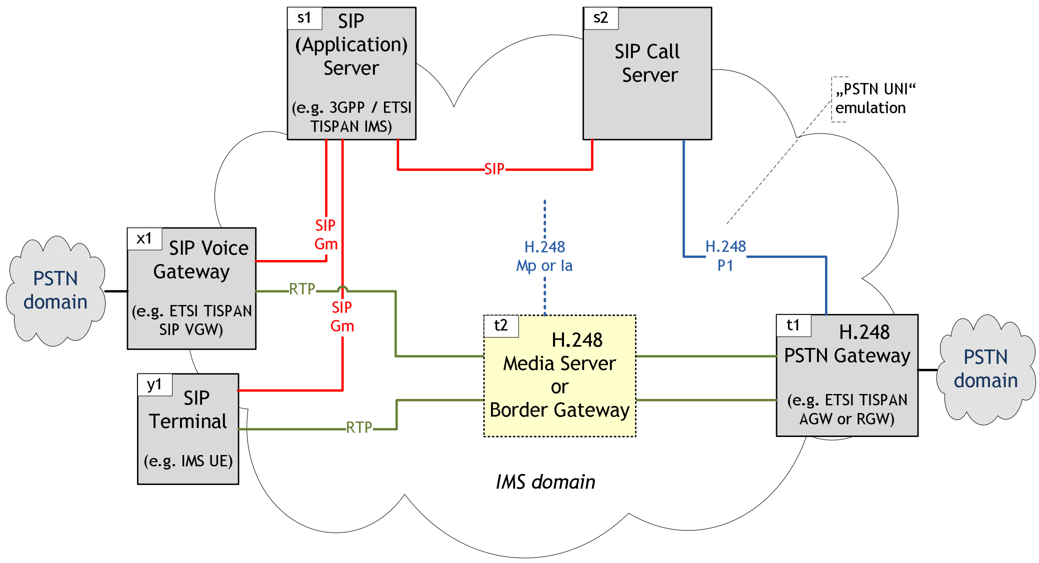

Figure 6: IMS & IMS-based PES to PSTN UNI (intra-IMS call) - Unsuccessful E2E negotiations demand for an e.g. media server

Possible solution:

Option 1: IP media-path routed via (Mp-controlled) media server;

Option 2: IP media-path routed via (Ia-controlled) border gateway:

- media-aware mode with V.152-to-T.38 interworking (according ITU-T Recommendation V.153 [i.13]);
- NAPT-less mode in order to keep the single routing domain (of the single IMS provider domain).

9.3 Scenario #3 - Inter-IMS call between two provider domains (IMS peering)

H.248 border gateways (here: t1 and t2) are positioned in the IP media-path (bearer-path) when peering IMS provider domains (see Figure 7). The codec negotiation via SDP Offer/Answer is subject of the *Ici* interface between providers. E.g. there could be an "end-to-end" negotiation via SIP between the SIP gateways x1 and x2.

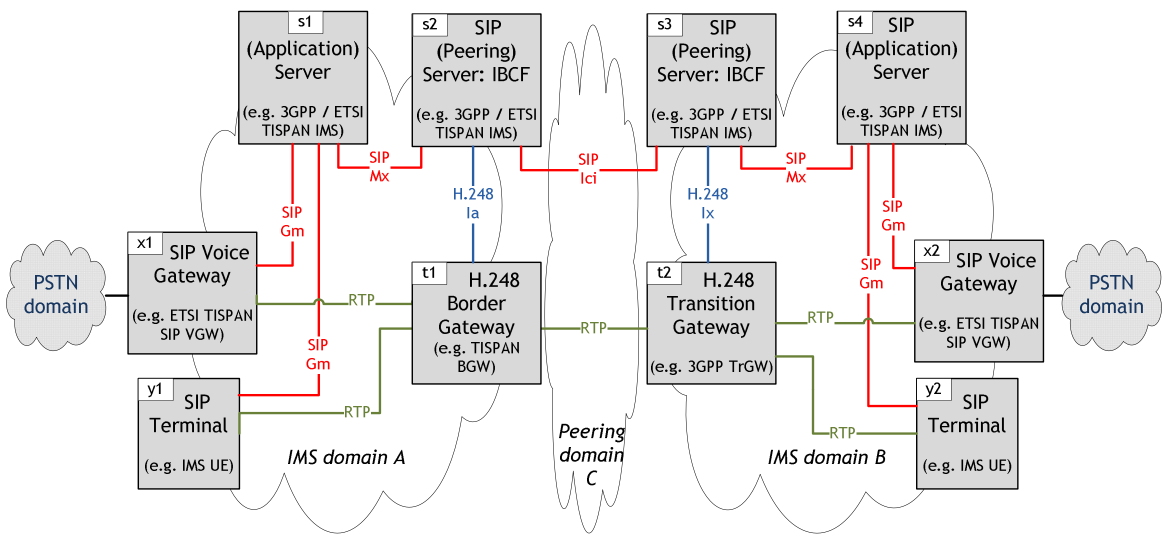

Figure 7: Inter-IMS call between two provider domains

"Codec negotiation" (= indication & negotiation of media & bearer configurations) in IP network:

- additional IMS SIP/SDP interfaces (here *Ici*);
- additional policy control (gateway control) interfaces for H.248-based policy enforcement: here H.248 Ia profile(s) for ETSI border gateway/routers [i.22] and H.248 Ix profile for 3GPP border gateway/routers (called Transition Gateway (TrGW) [i.18]).

9.4 Scenario #4 - IMS-PSTN UNI call

H.248 residential or access gateways (RGW; AGW) may be located in the bearer-path between an IMS and PSTN domain (Figure 8). The gateway location relates to PSTN UNI.

"Codec negotiation" (= indication & negotiation of media & bearer configurations) in IP network:

- IMS SIP/SDP interfaces;
- H.248 ARGW profile (at P1) for ETSI ARGWs [i.21].

9.4.1 Scenario #4bis - IMS-based PES to PSTN UNI (single IMS provider)

Figure 8 illustrates a mix of scenario #1 and #4, under the condition that the IMS domain is operated by a single provider (thus, intra-IMS call scenario).

The capabilities (media, transport) of the media plane devices x1, y1 and t1 allow the successful negotiation of end-to-end emulation services for PSTN modem calls. The supported capabilities may be abstracted by the concept of *Supported Codec Lists* (SCL; i.e. here SCL_{x1}, SCL_{y1} and SCL_{t1}). Successful negotiation means that the final *Negotiated Codec List* (NCL) provides at least one media configuration for PSTN modem call traffic.

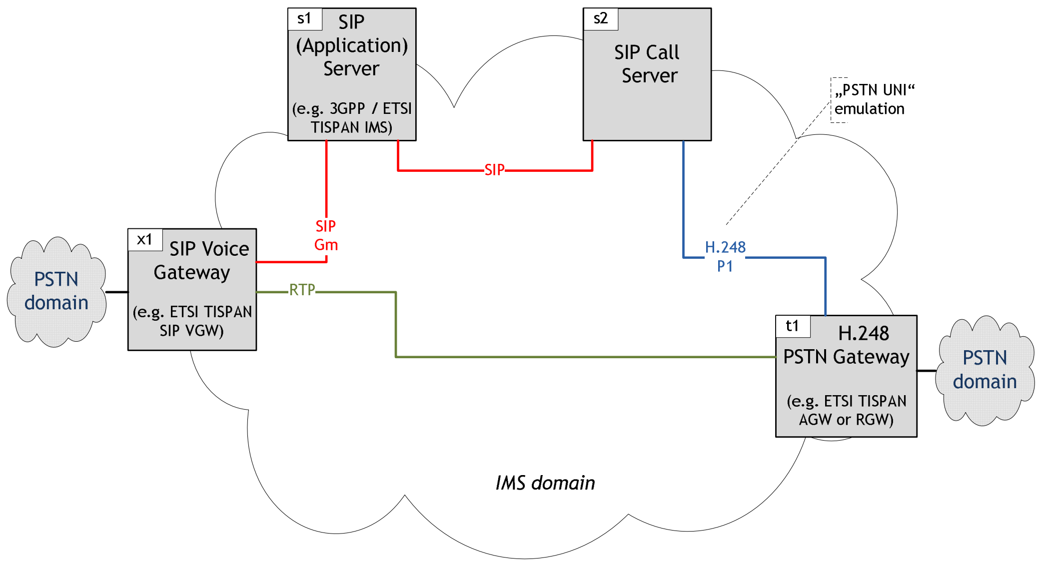

Figure 8: IMS-based PES to PSTN UNI (intra-IMS call) - Successful negotiations possible

9.5 Scenario #5 - IMS-PSTN NNI call

H.248 trunking gateways (TGW) may be located in the bearer-path between an IMS and PSTN domain (Figure 9). The gateway location relates to PSTN NNI.

"Codec negotiation" (= indication & negotiation of media & bearer configurations) in IP network:

- IMS SIP/SDP interfaces;
- H.248 TGW profile (at Mn).

9.6 Scenario #6 - IMS-PSTN (general)

Figure 9 provides a summary of the PSTN UNI & NNI interworking scenarios (#4 & #5). It may be reminded again that there are slightly different objectives concerning the termination of a PSTN modem call in a 3GPP user equipment versus TISPAN scenarios (see TISPAN R1 discussions, e.g. temporary documents *08TD241* [i.35] (2005-09) or *ACWG3-H248* TD08 [i.36] (2005-11)).

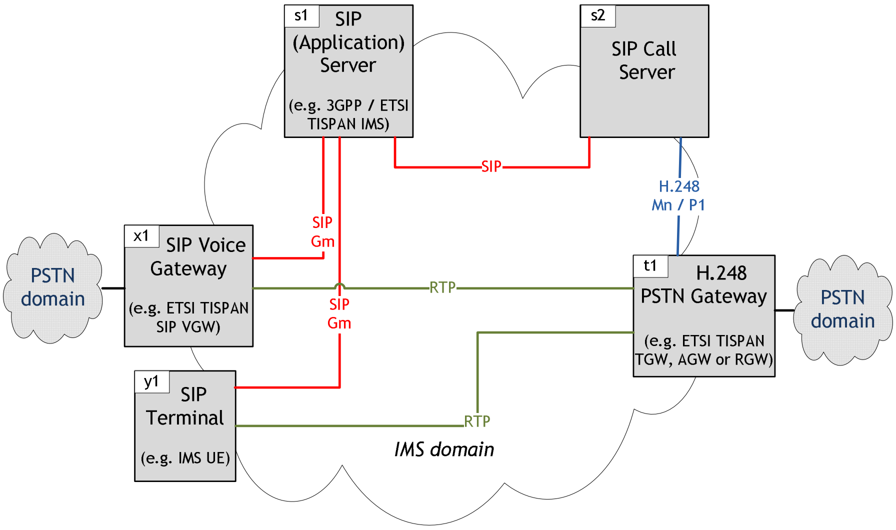

Figure 9: Two use cases - (a) IMS-PSTN UNI call (= H.248 ARGW profile at P1); (b) IMS-PSTN NNI call (= H.248 TGW profile at Mn)

9.7 Scenario #7 - PES-IMS call between two provider domains

H.248 border gateways (here: t1) may ("not mandatory") positioned in the IP media-path (bearer-path) when connecting an IMS and PES network (see Figure 10). This scenario is not adding new aspects with regards to PSTN modem calls.

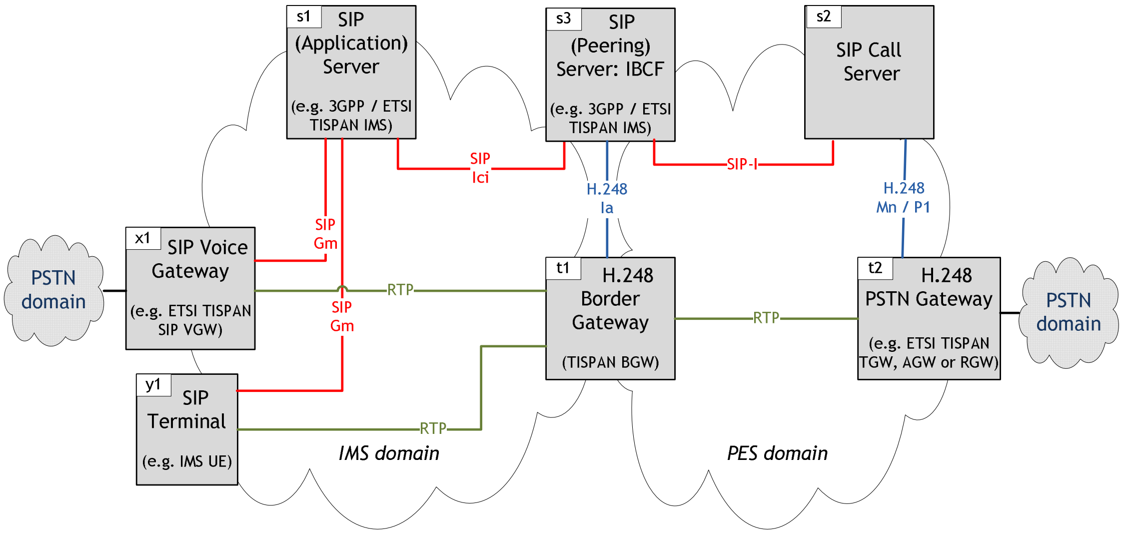

Figure 10: PES-IMS call between two provider domains

9.8 Scenario #8 - PES only

Figure 11 illustrates a PES only configuration. There is not any native SIP/SDP interface here. This scenario is thus out of scope of SDP Offer/Answer procedures.

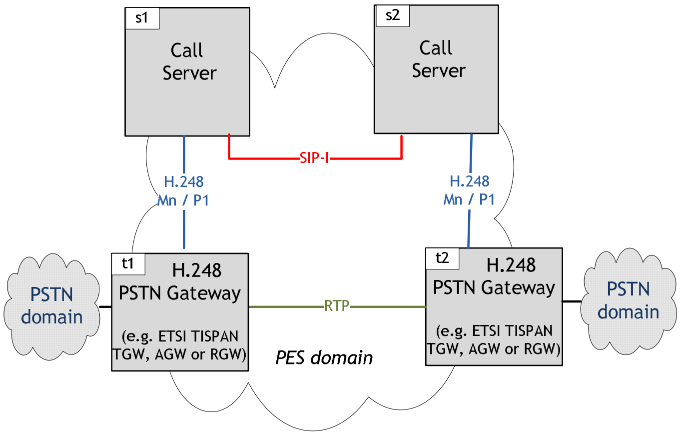

Figure 11: PES-only call, single domain

9.9 Scenario #9 - Others

Real world deployment have to respect potential interworking with BICC-, H.323- or MGCP-control VoIP networks, or other control plane protocols. Such scenarios are out of scope of this contribution.

9.10 Summary - Use Cases vs Signalling Capabilities

Table 6 provides a high-level summary of the example use cases versus involved signalling interfaces and protocols.

Table 6: Involved (TISPAN) *Control Plane Interfaces* versus *Scenario*

Scenario		SIP/SDP Gm	SIP/SDP Mx / Mw / Ici	SIP-I … / Nc	H.248/SDP P1	H.248/SDP Mn	H.248/SDP Ia / Ix	others (out of scope)
No.	Type	[1]	[2]	[3]	[4]	[5]	[6]	[7]
#1.1	IMS-based PES (UNI-UNI)	X	—	—	—	—	—	
#1.2	IMS-based PES (NNI-NNI)	—	X	—	—	—	—	
#2	IMS / IMS-based PES (general)	X	X	—	—	—	—	
#3	IMS Peering	X	X	—	—	—	X	
#4	IMS to PSTN UNI	X	X	—	X	—	—	
#5	IMS to PSTN NNI	X	X	—	—	X	—	
#6	IMS to PSTN (general)	X	X	—	X	X	—	
#7	IMS to PES (general)	X	X	X	X	X	X	
#8	PES only	—	—	X	X	X	—	
#9	others							

NOTES: Major references for control plane protocol specifications:
[1] TS 183 043 [i.24], ES 283 003 [i.25]; TS 124 229 [i.16]
[2] same as [1]
[3] TS 129 231 [i.27]
[4] TS 183 002 [i.21]
[5] ES 283 049 (=TS 129 624 / TS 129 424) [i.27]; TS 129.332 [i.19]
[6] TS 183 018 [i.22], TS 129 238 [i.19]
[7] H.323, MGCP, BICC (e.g. 3GPP Nc for IP-based Nb)

Annex A: Support of XoIP bearer services for PSTN modem calls: Inventory of correspondent TISPAN specifications

This annex provides an overview of ETSI Specifications versus support of IP bearer emulation services for PSTN modem calls. This overview provides a snapshot from end of 3GPP Release 9. There are two key aspects as outlined by clause 4:

1) SDP Offer/Answer protocol capabilities
   - status: TS 124 229 [i.16] support of:
     a) legacy SDP O/A (RFC 3264 [i.1]) and
     b) revised SDP O/A for:
        - *transport* capabilities (RFC 5939 [i.4]); and
        - *media* capabilities (MediaCapNeg [i.5]);
   - further: *examples* needed for SDP Offers, similar to TS 126 114 [i.17], Annex A
2) *SDP syntax* for required media configurations for emulation services V.152, T.38, …
   - status: see Table A.1 on next page (used colour code: status of supported capabilities is indicated in red; required capabilities are indicated in green).

Table A.1: ETSI Specifications versus support of IP bearer emulation services for PSTN modem calls (Status: end of 3GPP Release 9)

Document	Version	Title	pseudo-VBD (pVBDoIP)	V.152 (VBDoIP)	T.38 (FoIP)	V.151 (ToIP)	V.150.1 (MoIP)
TS 182 012 [i.20]	V2.1.4 (2008-03)	IMS-based PSTN/ISDN Emulation; Functional architecture	Status: stage 2 is still open Required: see V.152	Status: stage 2 is still open Required: V.152 service may be mentioned in clause 11 "mode of operation"	Status: stage 2 is still open Required: V.152 service may be mentioned in clause 11 "mode of operation"	Status: stage 2 is still open Required: V.152 service may be mentioned in clause 11 "mode of operation"	Status: stage 2 is still open Required: V.152 service may be mentioned in clause 11 "mode of operation"
TS 183 043 [i.24]	V2.3.1 (2009-03)	IMS-based PSTN/ISDN Emulation; Stage 3 specification	Status: nothing mentioned at all how PSTN modem calls shall be treated by the IMS/PES domain Required: see V.152	Status: V.152 not supported Required: mandatory support of V.152	Status: T.38 not supported "Support of T.38 is outside the scope of TISPAN NGN present Release." Required: optional support of T.38: statement that V.152 may be used alternatively for fax/modem calls	Status: V.151 not supported Required: statement that V.152 shall be used for text/modem calls	Status: V.150.1 not supported Required: statement that V.152 shall be used for text/modem calls
TS 183 036 [i.23]	V2.1.1 (2009-01)	ISDN/SIP interworking; Protocol specification *ISDN only Emulation services for PSTN modem calls missing*	Status: unclear; Table 5.1.1.1.4-2 mentions the option of a 2nd G.711 codec with dynamic PT, but the exact mapping on a SDP offer is open Required: see V.152	Status: V.152 not supported Required: mandatory support of V.152	Status: T.38 supported, but application limited on ISDN; indication via Q.931 HLC IE … Required: optional support of T.38 (note 1); statement that V.152 may be used alternatively for fax/modem calls (note 2)	Status: V.151 not supported Required: statement that V.152 shall be used for text/modem calls	Status: V.150.1 not supported Required: statement that V.152 shall be used for text/modem calls

Document	Version	Title	pseudo-VBD (pVBDoIP)	V.152 (VBDoIP)	T.38 (FoIP)	V.151 (ToIP)	V.150.1 (MoIP)
ES 283 003 [i.25]	V2.6.1 (2008-08)	IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS **24.229**…].	Status: XoIP bearer emulation services: neither SDP for V.152 nor for T.38 mentioned (NOTE 3), thus any of above XoIP bearer emulation services for PSTN modem/calls is not defined Codec/configuration indication & negotiation: *legacy* SDP O/A (RFC 3264 [i.1]) mandatory, *revised* SDP O/A (SDPCapNeg and MediaCapNeg) fully supported (for above XoIP configurations) Required: mandatory support of V.152 for SIP VGWs, optional support of T.38 for SIP VGWs / mandatory for SIP UEs; SDP O/A examples for XoIP negotiations (similar to the examples in TS 126 114 [i.17], Annex A for IMS MMoIP UEs)				

NOTE 1: Table 5.1.1.1.4-2 "*Coding of SDP media description lines from BC/HLC to SIP*" provides a column for SDP attributes, but the table is lacking an explicit list of T.38 SDP attributes. There are ten T.38 parameters defined for T.38 versions 0 to 3. T.38 version 4 provides an additional parameter ("*T38ModemType*"). T.38 versions 0 to 2 do not yet support V.34-capable G3FE. It is therefore required to signal (indicate and negotiate) at least T.38 parameters "*T38FaxVersion*" and "*T38FaxRateManagement*". If the probability of possible interoperability issues (with existing T.38 implementations) shall be minimized, then it is recommended to signal all T.38 parameters. If not, then at least the T.38 default values should be explicitly specified in the applied SIP profile, or an explicit reference to Table H.1/T.38 should be provided.

NOTE 2: Table 5.1.1.1.4-2 "*Coding of SDP media description lines from BC/HLC to SIP*" should be extended by an explicit list of V.152 SDP attributes, which are not listed yet.

NOTE 3: Relevant references are Tables A.319 and A.330 in TS 124 229 [i.16] with regards to "*zero or more session / media attribute lines (a=)*", which missing yet an explicit list of all SDP attributes for V.152 and T.38.

A.1 T.38 and V.152 call support in IMS releases?

The lacking of an explicit list of SDP attributes in the IMS SIP profile specification (see Table A.1; Tables A.319 and A.330 in TS 124 229 [i.16]) raises the question whether T.38 or/and V.152 calls are supported at all in existing IMS releases (i.e. 3GPP pre-R10, TISPAN pre-R3). There are two possible positions:

A.1.1 "Open" IMS position

The "open IMS position" represents a proceeding by allowing the (fast) introduction of new IMS applications, without any explicit update of the IMS SIP profile concerning lower level details like each SDP attribute type and possible attribute parameters. The SIP messages would just provide a container function for such SDP information elements, which are only understood by the SIP endpoints (like IMS UE, AS).

The RFC 3362 [i.34] MIME type for T.38 is already supported (by 29.163, 24.229), but allows only the indication of a T.38 media stream in SDP, but not any indication and negotiation of T.38 configuration settings in detail.

T.38 and V.152 IMS calls may be already supported in an "open IMS" environment, because the T.38 and V.152 endpoints support anyway the correspondent SDP attributes (as part of their compliance against these ITU-T Recommendations).

A.1.2 "Strict service control" IMS position

This approach implies the full specification of all required signalling elements in the IMS SIP profile. Any present 24.229 SIP/SDP controlled T.38 FoIP service would therefore violate T.38, because not supporting the mandatory T.38 parameter from Annex D/T.38 (this Annex is normative for SIP/SDP-controlled T.38 endpoints). Leading to the conclusion: that SIP profile 24.229 (and thus IMS) does not (yet) support the indication and negotiation of T.38 configurations.

A.1.3 Discussion

The "strict service control" IMS approach is currently followed by 3GPP and TISPAN, primarily motivated by security concerns due to untrusted SIP traffic.

However, the security situation might be more relaxed in case of PSTN modem calls: the majority of use cases (in clause 9) relates to call endpoints located in PSTN domains (which are trusted domains per se), with IP emulation services between PSTN-to-IMS gateway types (thus SIP endpoints located in network elements, rather than SIP user equipment).

Annex B: Support of *Revised* SDP Offer/Answer Syntax by 3GPP SIP Profile specifications

This annex provides an inventory with respect of protocol support for *revised* SDP Offer/Answer Syntax by 3GPP SIP Profile specifications. This overview provides a snapshot from end of 3GPP Release 9. Following 3GPP specifications are relevant in scope of considered use cases (see clause 9) for PSTN modem call emulation services:

- TS 124 292 (V9.3.0): "Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia CoreNetwork subsystem Centralized Services; Stage 3 (3GPP TS 24.292 Release 9)".
- TS 124 229 (V9.3.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) (3GPP TS 24.229 Release 9)".

Table B.1: Support of *Revised* SDP Offer/Answer Syntax by 3GPP SIP Profile specifications

No.	SDP extension	IETF reference	Required for PSTN modem call emulation service	Support by TS 124 229 [i.16] *SIP Gm Profile*	Support by TS 124 292 [i.37] *SIP "ICS" Profile (see note 2)*	Support by TS 129 163 [i.38] *Interworking between IMS and Circuit Switched networks*
1	RFC **'SDPCapNeg'**: Framework for Revised SDP Offer/Answer model and SDP syntax extension concerning transport capabilities	Andreasen, F., "SDP Capability Negotiation", RFC 5939 [i.4]	YES	YES	YES	NO (note 1)
2	RFC '**MediaCapNeg**': SDP syntax extension for *media capabilities*	Gilman, R., et al., "*SDP media capabilities Negotiation*", draft-ietf-mmusic-sdp-media-capabilities [i.5]	YES	YES	YES	NO (note 1)
3	RFC '**ConnCapNeg**': SDP syntax extension for connection type capabilities	Garcia-Martin, M. and S. Veikkolainen, "*Session Description Protocol (SDP) Extension For Setting Up Audio Media Streams Over Circuit-Switched Bearers In The Public Switched Telephone Network (PSTN)*", draft-ietf-mmusic-sdp-cs [i.6]	NO (optional)	YES	YES	NO (note 1)
4	RFC '**MiscCapNeg**': SDP syntax extension for further capabilities concerning SDP lines "b=", "c=" and "i="	Garcia-Martin, M. et al., "*Miscellaneous Capabilities Negotiation in the Session Description Protocol (SDP)*", draft-ietf-mmusic-sdp-misc-cap [i.7]	NO (optional)	YES	YES	NO (note 1)
NOTE 1: TS 129 163 defines SDP Offer/Answer information, which needs to be consistent with TS 124 229. It is thus expected that TS 129 163 [i.38] will be aligned with TS 124 229 [i.16] concerning supported SDP Offer/Answer capabilities. NOTE 2: Centralized Services (i.e. anchor point of service located in IMS domain).						

Annex C: Discussion of SIP/SDP interworking by example scenarios

This annex provides some example scenarios, as illustration for interworking discussion.

C.1 Media grouping

Media grouping is described in clause 7.4.

C.1.1 Example 1 - Two SIP VGWs with different SDP Capability Sets, but successful usage of RFC 3388 media grouping

Figure C.1 illustrates an example use case. There are two SIP Voice Gateways (VGWs x1 and x2), with different SDP capabilities supported:

- **SIP VGW$_{x2}$** provides support of *legacy* SDP Offer/Answer, but also the SDP elements according RFC 3388 [i.28] in order to address the *media grouping* goal (for grouping audio and image media types), and RFC 3407 [i.29] in order to address the *capability declaration* goal (for early declaration of T.38 support and possible later usage; see clause 7.5).

- **SIP VGW$_{x1}$** provides support of *revised* SDP Offer/Answer plus SDP elements according RFC 3388 [i.28] (and RFC 3407 [i.29]) due to backward compatibility reasons.

NOTE 1: The functionality of RFC 3388 [i.28] and RFC 3407 [i.29] is covered by *revised* SDP Offer/Answer. They would be consequently not needed if all SIP VGWs would support *revised* SDP Offer/Answer. However, due to backward compatibility reasons they might be beneficial (e.g. in order to provide an improved emulation s ervice, better than just possible with *legacy* SDP Offer/Answer).

NOTE 2: This clause is only about RFC 3388 [i.28] in order not to mix too much capabilities. The subject of RFC 3407 [i.29] is similar, but out of scope of this clause.

There is only one SIP profile involved, the TS 124 229 "SIP *Gm* Profile", for the capability indications and negotiations between the SIP VGWs.

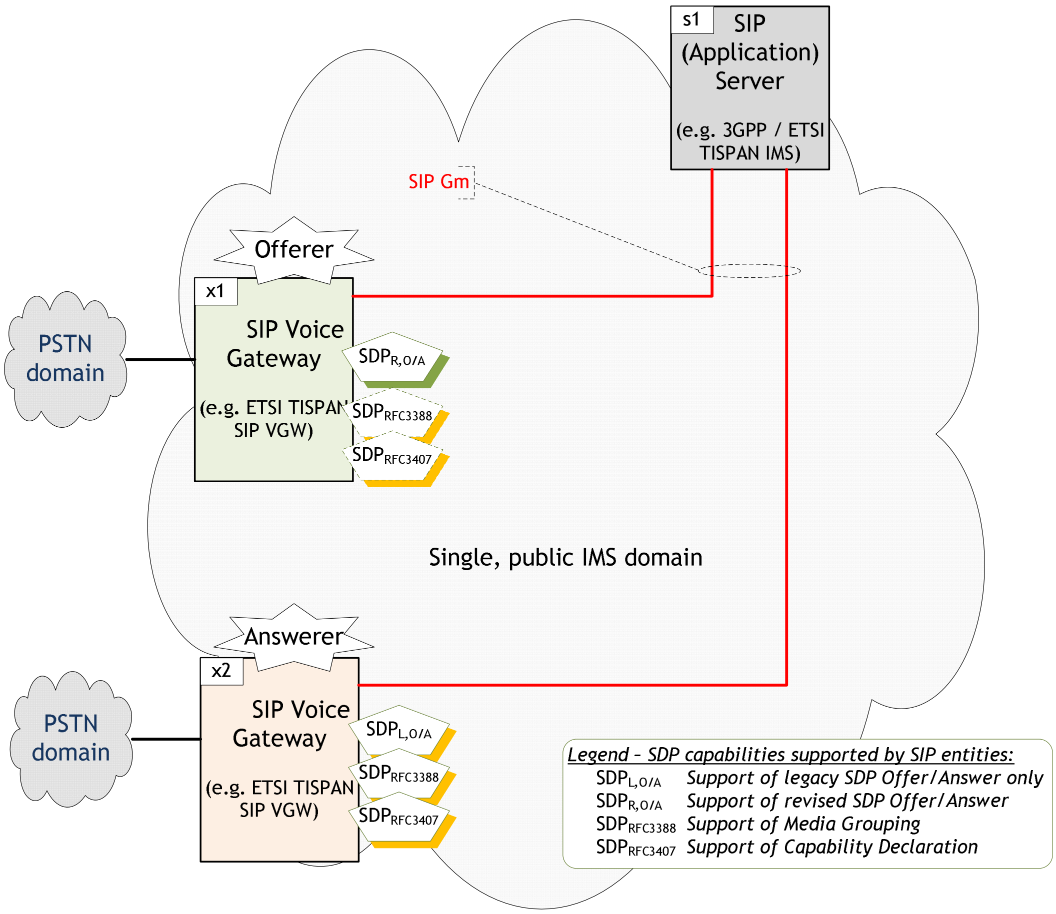

Figure C.1: Use case - IMS-based PES scenario, intra-IMS call between two SIP gateways *x1* and *x2* with different SDP capability sets supported

Figure C.2 illustrates the session configuration, *offered* in this example:

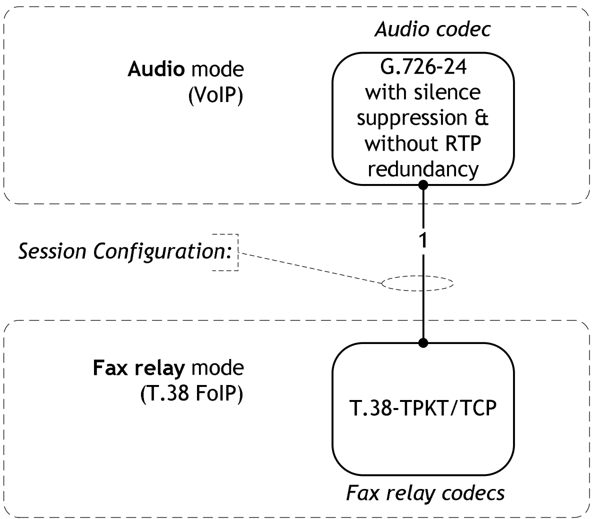

Figure C.2: Potential session configurations *offered* in example 1

IMS call establishment direction from *x1* (Offerer) and *x2* (Answerer):

- Offered configuration list (by x1), see Table C.1:
  OCL_{x1} = {VoIP audio codecs: ADPCM-24 incl silence suppression | FoIP: FoTPKT/TCP | VBDoIP: -}

- Answered configuration list (by x2), see Table C.2:
  $ACL_{x2} = OCL_{x1}$

- conclusion, - negotiated configuration list: $NCL_{x1-x2} = OCL_{x1}$

Table C.1: SDP example ((shortened SDP description)) in *Revised* SDP Offer/Answer syntax - OFFER

SDP offer (embedded in SIP INVITE)

<pre>…
1) OFFER (embedded in SIP INVITE):
…
; ACTUAL CONFIGURATION (due to backward compatibility)
a=group:FID 1 2
m=audio 49170 RTP/AVP 111 112
a=rtpmap:111 G726-24/8000   ; ADPCM (in 24 kbps mode) as audio codec
a=ptime:…
a=rtpmap:112 CN/8000   ; Comfort Noise transport with dynamic PT value
a=mid:1
m=image 49172 tcp t38
a=T38FaxRateManagement:localTCF
a=… (… additional T.38 TPKT attributes should
             be included)
a=mid:2
;
; SESSION CONFIGURATIONs
a=sescap:1 1,2 ; VoIP = G.711, FoIP = T.38 TPKT/TCP
 	
; LATENT CONFIGURATIONs for T.38
a=tcap:2 tcp   ; T.38 FoTPKT/TCP transport variant
a=mcap:1 t38   ; T.38 FoIP codec (subtype = 't38')
a=acap:11 T38FaxVersion:4
; Transport-dependent T.38 parameters for TPKT/TCP
a=acap:12 T38FaxRateManagement:localTCF
a=acap:13 (… all T.38 TPKT attributes should
             be included)
a=lcfg:2 mt=image t=2 m=1 a=-ms:11,…
 	
; POTENTIAL CONFIGURATION
a=tcap:1 RTP/AVP
a=mcap:2 G726-24/8000 ; audio codec
a=mcap:3 CN/8000   ; comfort noise
a=acap:1 ptime…
 	
; Preferences
a=pcfg:1 mt=audio t=1 m=2,3 pt=1:111,112 a=-ms:1</pre>

Offered (1) potential configuration (as session configurations due to 'voice' and 'facsimile'): - Preference 1: Audio (G.726-24 with enabled silence suppression) and T.38 FoTPKT/TCP. The T.38 configuration is indicated as latent configuration. The audio mode is specified as "a=pcfg:1". The session configuration concept is required due to the two media types.

Table C.2: SDP example ((shortened SDP description)) in *Revised* SDP Offer/Answer syntax - ANSWER

SDP answer (embedded in SIP INVITE)

<pre>…
2) ANSWER (embedded in SIP message):
…
a=group:FID 1 2
m=audio 49170 RTP/AVP 111 112
a=rtpmap:111 G726-24/8000   ; ADPCM (in 24 kbps mode) as audio codec
a=ptime:…
a=rtpmap:112 CN/8000   ; Comfort Noise transport with dynamic PT value
a=mid:1
m=image 49172 tcp t38
a=T38FaxRateManagement:localTCF
a=… (… additional T.38 TPKT attributes should
             be included)
a=mid:2</pre>

The Answerer supports all requested configurations and provides a positive acknowledge. The Answerer does not support *Revised* SDP Offer/Answer syntax, thus the *actual configuration* remains.

Conclusions from example 1:

- RFC 3388 [i.28] media grouping may be used, independent of *revised* SDP Offer/Answer syntax. RFC 3388 [i.28] would be obsolete if both sides are complaint to *revised* SDP Offer/Answer.
- If one party is not supporting *revised* SDP Offer/Answer, then it is still better to used RFC 3388 [i.28] than without any grouping approach.

C.1.2 Example 2 - RFC 3388 deficiencies

Same configuration and SDP capabilities as in Figure C.1, but multiple T.38 options offered by the SIP VGWs. Figure C.3 illustrate the two possible session configurations, *offered* in this example.

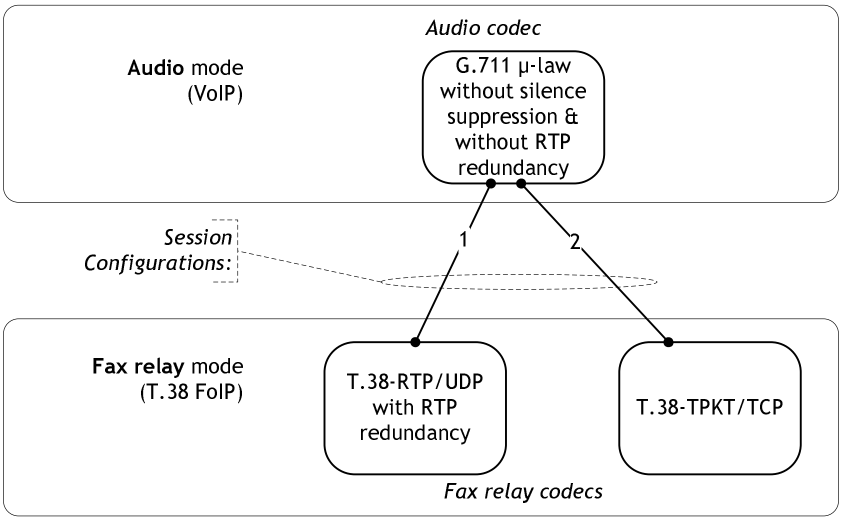

Figure C.3: Potential session configurations *offered* in example 3

Table C.3 provides an attempt for using RFC 3388 [i.28] in this example, which is **not** possible.

Table C.3: SDP example ((shortened SDP description)) in *Revised* SDP Offer/Answer syntax - Correct syntax of the SDP_R_O/A block, but incorrect syntax in the legacy part

SDP offer (embedded in SIP INVITE)
… 1) OFFER (embedded in SIP INVITE): … ; ACTUAL CONFIGURATION (due to backward compatibility) **a=group:FID 1 2** m=audio 49170 RTP/AVP 0 **a=rtpmap:0 PCMU/8000 ; explicit listing, as group member 1** a=ptime:… **a=mid:1** m=audio 49170 RTP/AVP 121 0 100 a=rtpmap:100 t38/8000 a=fmtp:100 T38FaxRateManagement=transferredTCF a=rtpmap:121 red/8000 a=fmtp:121 100/100 **a=mid:2** m=image 49172 tcp t38 a=T38FaxRateManagement:localTCF **a=mid:2** {***Note: that is the required mid value, however not possible according RFC 3388 "The identification tag MUST be unique within an SDP session description."***} ; ; SESSION CONFIGURATIONs a=sescap:1 1,2 ; VoIP = G.711, FoIP = T.38 RTP/UDP a=sescap:2 1,3 ; VoIP = G.711, FoIP = T.38 TPKT/TCP ; LATENT CONFIGURATIONs for T.38 a=tcap:1 RTP/AVP ; T.38 FoRTP/UDP transport variant a=tcap:2 tcp ; T.38 FoTPKT/TCP transport variant a=mcap:2 t38 ; T.38 FoIP codec (subtype = 't38') a=mcap:3 red/8000 ; RTP packet redundancy a=mfcap:3 %2%/%2% ; RFC 2198 redundancy format (T.38) a=mfcap:2 T38FaxRateManagement=transferredTCF a=acap:11 T38FaxVersion:4 ; Transport-dependent T.38 parameters for RTP/UDP a=acap:12 (… additional T.38 RTP attributes should be included, if required) ; Transport-dependent T.38 parameters for TPKT/TCP a=acap:21 T38FaxRateManagement:localTCF a=acap:22 (… additional T.38 TPKT attributes should be included) a=lcfg:2 mt=audio t=1 m=2,3 pt=2:121,3:100 a=-ms:11,… a=lcfg:3 mt=image t=2 m=2 a=-ms:11,… ; POTENTIAL CONFIGURATION a=mcap:1 PCMU/8000 ; audio codec ; Preferences a=pcfg:1 mt=audio t=1 m=1 pt=1:0 a=-ms:
Offered (2) potential configurations (as session configurations due to 'voice' and 'facsimile'): - Preference 1: Audio (PCMU) and T.38 FoRTP/UDP inclusive packet redundancy - Preference 2: Audio (PCMU) and T.38 FoTPKT/TCP The two T.38 options are indicated as latent configurations. The audio mode is specified as "a=pcfg:1". There are consequently two session configurations indicated.

Conclusions from example 2:

- RFC 3388 [i.28] media grouping is **not applicable** here.
- In general: RFC 3388 [i.28] can be not used in case of *multiple media groups* in *legacy* SDP Offer/Answer. This is an issue, e.g. if a PES endpoint supports multiple *T.38 versions* ("each T.38 version relates to a separate T.38 configuration"), or multiple transport modes, or/and multiple protection schemes for a particular transport mode. Just limiting on a single, specific T.38 configuration is either reducing successful interoperability, or implies a series of subsequent Offer/Answer cycles ("which may be impossible due to realtime constraints").

NOTE: The *revised* SDP Offer/Answer block is not affected because RFC 3388 [i.28] elements are just not required for the indication of "media groups", which is related to "session configurations" here.

C.2 Example 3 - More SIP VGWs with different SDP Capability Sets

There might be much more heterogeneity in real networks. Figure C.4 depicts an example with three SIP VGWs and different SDP capability sets supported.

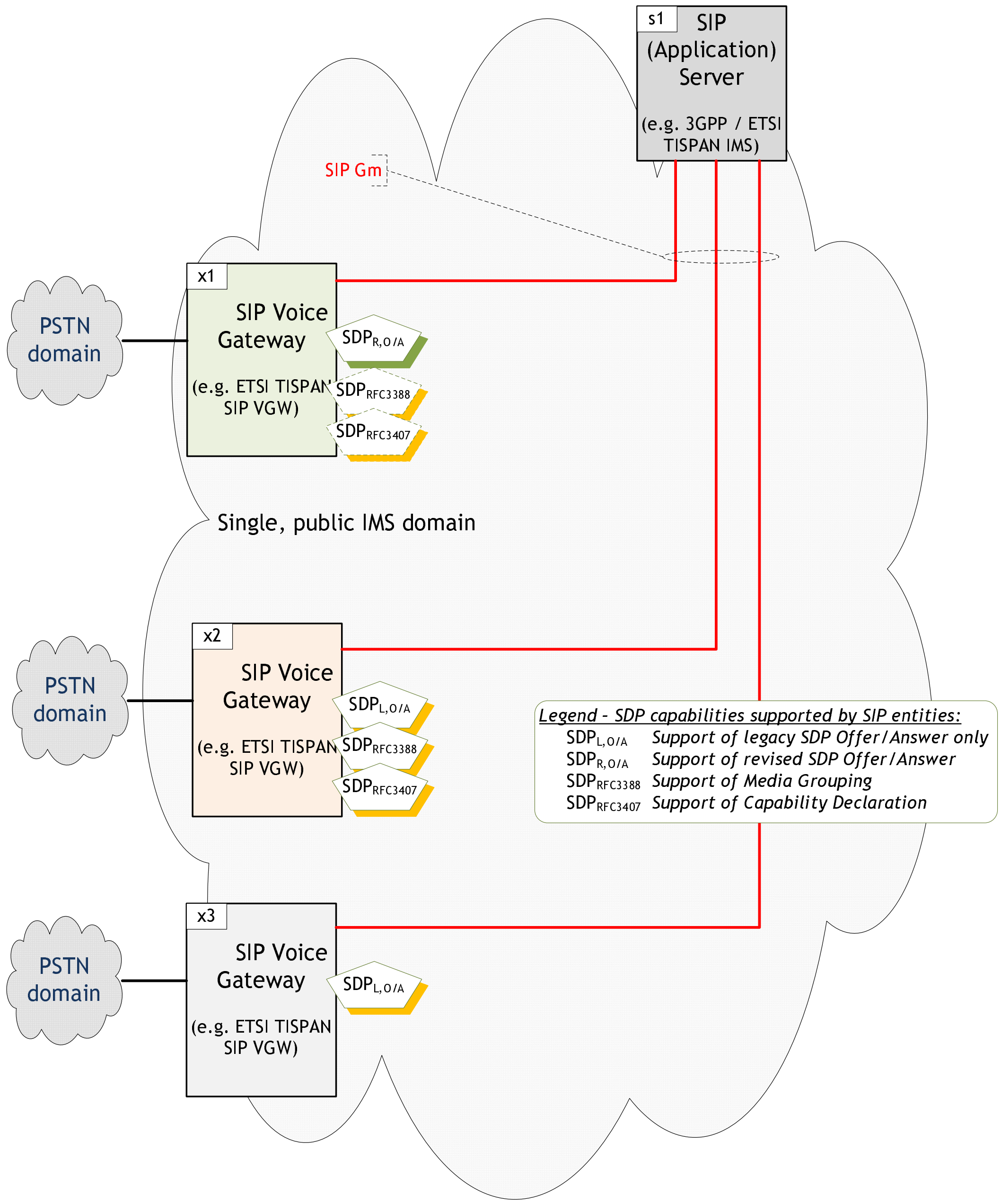

Figure C.4: Use case - IMS-based PES scenario, intra-IMS call - Three SIP gateways x1, x2 and x3 with different SDP capability sets supported

Some observations from example 3:

- Interworking between "*legacy* SDP Offer/Answer"-only gateways: e.g. **SIP VGW**$_{x2}$ may offer RFC 3388 [i.28] to **SIP VGW**$_{x3}$. Answerer **SIP VGW**$_{x3}$ would just ignore unsupported SDP syntax.

- Interworking between a "*legacy* SDP Offer/Answer" gateway (e.g. **SIP VGW$_{x2}$**) and a "*revised* SDP Offer/Answer" gateway (e.g. **SIP VGW$_{x1}$**), - there are two scenarios, dependent on the offerer and answerer role distribution:
    - Offerer **SIP VGW$_{x2}$** may just sent a *legacy* SDP Offer, which allows Answerer **SIP VGW$_{x1}$** to conclude the supported SDP Offer/Answer protocol capabilities.
    - Offerer **SIP VGW$_{x1}$** may sent a *revised* SDP Offer, which would mean that Answerer **SIP VGW$_{x2}$** would consider the *actual configuration* only (due to ignorance of unsupported SDP syntax).
    - Both interworking directions are unproblematic.
- Interworking between "*revised* SDP Offer/Answer"-only gateways: not any issues due to same level of SDP support.

Thus, interworking on SDP syntax level is not any problem in all scenarios. Unsupported SDP capabilities may lead to limited emulation services (in terms of quality or resource efficiency) for PSTN modem calls.

Annex D: "SDP Offer/Answer protocol variants" - Negotiation Phases

This annex evaluates an abstracted negotiation phase (embedded in overall call establishment phase) in order to point out the major differences between the two SDP Offer/Answer protocol variants (see clause 1.1.1).

Figure D.1 illustrates an example scenario: a SIP device at the edge of a PSTN and IMS domain (e.g. SIP VGW x1), acting as SDP Offerer when considering an outgoing PSTN call request. The SIP VGW supports the *revised* SDP Offer/Answer protocol variant.

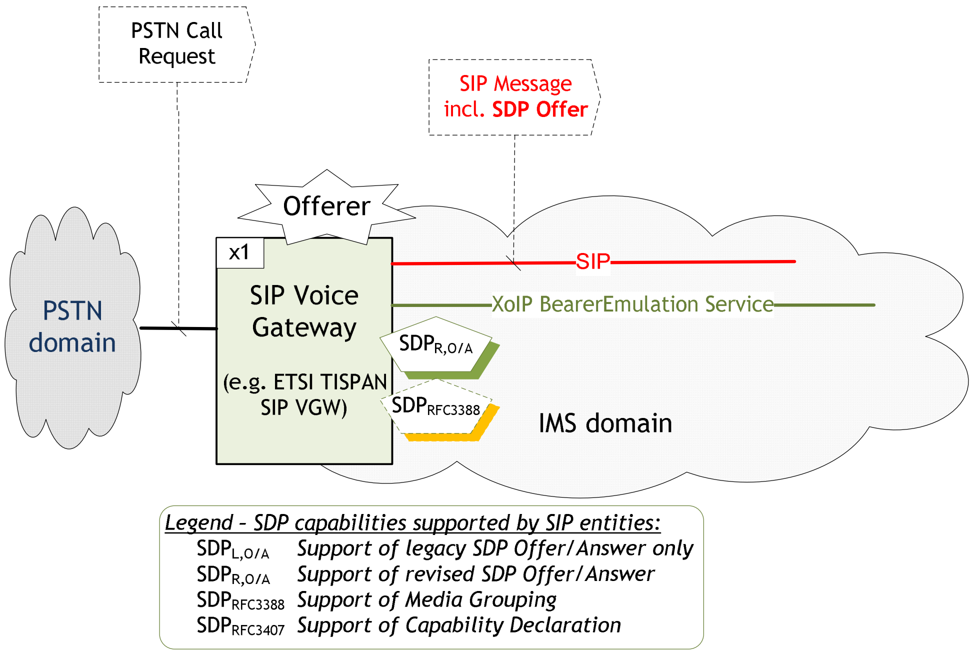

Figure D.1: Example scenario: SIP device (e.g. SIP VGW) as SDP Offerer, supporting *Revised* SDP Offer/Answer

The negotiation process (embedded in overall call establishment phase) may be abstracted in some phases (see Figure D.2). Relevant, in the scope of the present document, are following phases:

- negotiation preparation phase (B), given by:
  1) analysis of PSTN call request in terms of requested service capabilities;
  2) determination of required IP emulation service(s), which translates to a set of candidate SDP media configurations;
  3) production of a *generic* Offered Configuration List (OCL), taking into account:
     - locally supported configurations ($SCL_{Offerer}$); and
     - preferred configurations (PCL; e.g. given by operator policies);
  4) dependent on locally supported SDP Offer/Answer protocol variant;
  5) generation of the *real* OCL
     a) either in $SDP_{R,O/A}$ syntax; or
     b) in pure $SDP_{L,O/A}$ syntax.

NOTE 1: The mapping function between the *generic* OCL and *real* OCL differs between the SDP Offer/Answer protocol variants.

- *actual negotiation* phase between SIP entities (C).

NOTE 2: The negotiation logic differs between the SDP Offer/Answer protocol variants. There might be e.g. multiple O/A cycles in case of *legacy* SDP O/A, instead of a single cycle in *revised* SDP O/A.

NOTE 3: The Offerer falls back in *legacy* SDP O/A negotiation logic when the first SDP Answer is indicating that the remote entity is *not* supporting *revised* SDP O/A.

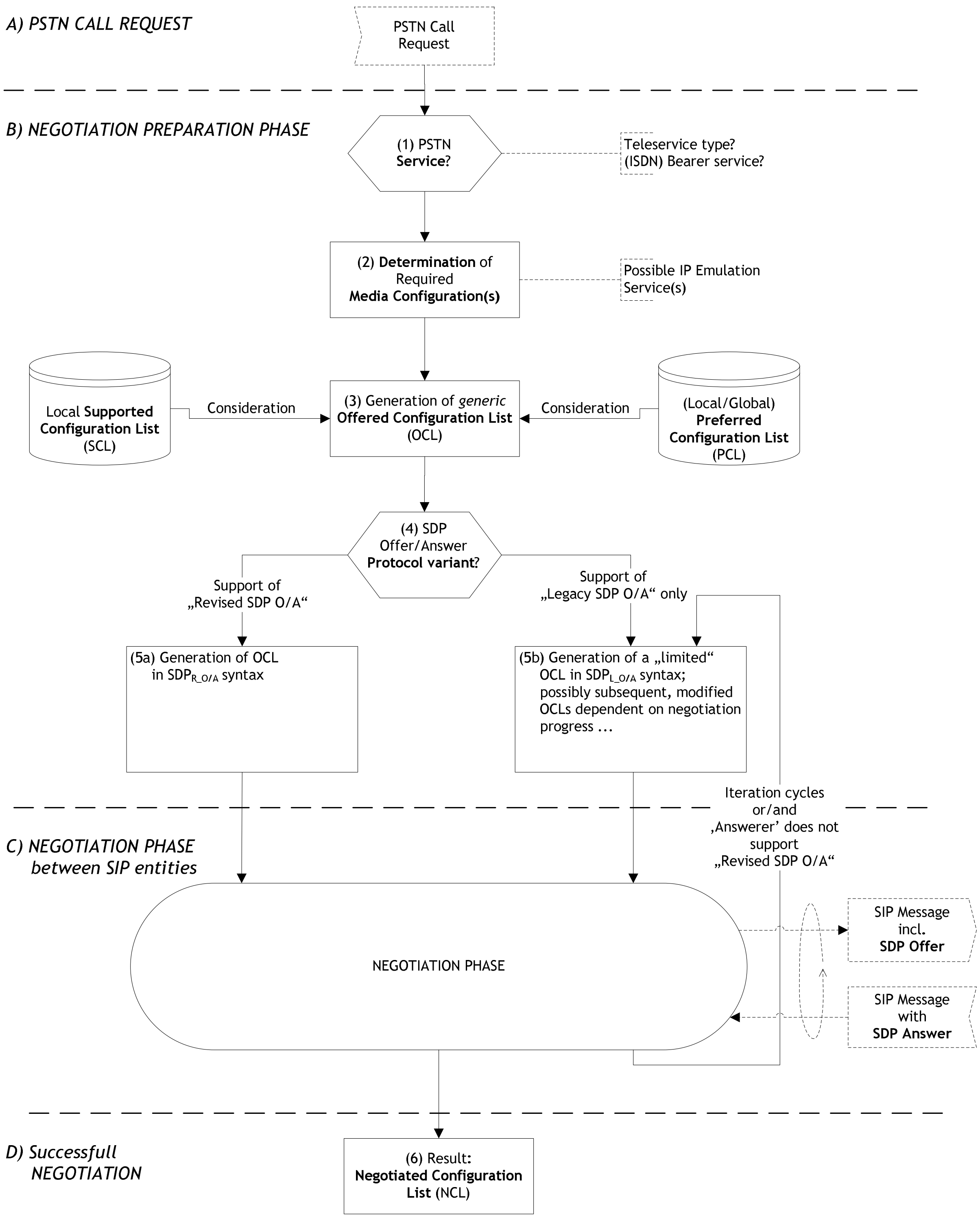

Figure D.2: Generic Negotiation Phase (embedded in overall Call Establishment Phase) - here SIP device (e.g. SIP VGW) acting with SDP Offerer role

Thus, besides the pure syntactical differences between the two SDP Offer/Answer protocol variants, the correspondent semantic leads to slightly different processing logic like e.g. in terms of the production of media configuration lists and negotiation procedures.

Annex E: Bibliography

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

ITU-T Recommendation Q.1400: "Architecture framework for the development of signalling and OA&M protocols using OSI concepts".

History

Document history		
V3.1.1	September 2010	Publication