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Interworking with other networks

Procedures for supporting voice-band data over IP networks

Recommendation ITU-T V.152

ITU-T V-SERIES RECOMMENDATIONS

DATA COMMUNICATION OVER THE TELEPHONE NETWORK

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Recommendation ITU-T V.152

Procedures for supporting voice-band data over IP networks

Summary

Voice-band data traffic has traditionally been transported by circuit switched systems and equipment. With the advent of the networks optimized for the transport of Internet Protocol (IP), and as a result of its considerable growth and pervasive nature, more and more voice-band data traffic is expected to be carried over IP networks.

Given that voice and voice-band data services remain a significant part of telecommunications, there is a need to ensure a high quality of service for voice and voice-band data carried in part, or wholly, via IP. Recommendation ITU-T V.152 defines procedures for equipment that interconnect GSTN networks with IP networks to provide satisfactory, transparent delivery of modulated voice-band data (VBD) as encoded audio content over IP (data modems, facsimile terminals and text telephones).

Annex B defines a method that uses data signal detection and silence insertion in voiceband data that adds a means of providing bandwidth savings during transmission.

Annex C addresses a problem discovered during the implementation of ITU-T V.152 gateways with the transmission of facsimile terminals. While this issue has been resolved by an amendment to Recommendation ITU-T T.30, due to the extremely large numbers of terminals deployed in the field and the low probability that they would be corrected retroactively, it was considered pre-emptive to include a solution in an ITU-T V.152 media gateway.

This Recommendation is complementary to the modem relay and voice-band data Recommendations (Recommendations ITU-T V.150.0 and V.150.1).

This revision includes the updating of references, general editing improvements for clarity and many additional examples of call control. It also integrates changes introduced by Corrigendum 1 (2005) for clarification to clauses 7.1 and 7.1.1; by Corrigendum 2 (2006) for clarifications on the use and control of echo cancellers and the application of IETF RFC 2833 with VBD; and by Amendment 1 (2009) for new Annexes B and C.

History

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Echo canceller, facsimile over IP, gateway, Internet gateway, Internet Protocol, IP gateway, media gateway, media gateway controller, modem over IP, quality of service, speech coding, TDM, TDM-IP gateway, text over IP, textphone over IP, text telephone, VBD, voice-band data, voice gateway, voice over IP, VoIP.

FOREWORD

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In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure, e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

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As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

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Procedures for supporting voice-band data over IP networks

1   Scope

This Recommendation describes the voice-band data (VBD) operation of Voice-over-Internet Protocol (VoIP) gateways and media gateways. The term "VBD" refers only to the use of suitable voice-band codecs for the transport of data payloads via RTP. The VBD procedures described in this Recommendation shall apply to VBD-only capable gateways. An ITU-T V.152 gateway only has the guarantee of interworking with another gateway if that gateway also supports ITU-T V.152.

This does not preclude the interworking with non-ITU-T V.152 gateways; however, there are no guarantees of performance and supported functionality with such systems. The negotiation of a VBD capability does not exclude from a VoIP session any other capabilities such as the transport of audio signals, RFC 4733-based telephone events, ITU-T T.38 facsimile relay, RFC 4103 text relay and ITU-T V.150.1 modem relay, etc.

Declaration of VBD support using the session description protocol (SDP) is detailed in clause 7.1.

Declaration of VBD support using ITU-T H.245 is detailed in clause 7.2.

This Recommendation supports hybrid modes of operation; for example, a device may support a VBD and facsimile relay capability but not a modem relay or text relay capability. In this example of hybrid operation modem and text payloads are transported in the VBD mode, while facsimile payloads could be transported in the ITU-T T.38 facsimile relay mode or in the VBD mode. The negotiation of such hybrid sets of capabilities follows SDP and ITU-T H.245 mechanisms (clause 11).

This Recommendation describes the default mechanism for transitioning into VBD mode via payload-type switching as described in clause 10 and the optional mechanism of utilizing the state signalling events (SSE) messages described in clause 11.

2   References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T G.168]   Recommendation ITU-T G.168 (2004), *Digital network echo cancellers.*

[ITU-T G.701]   Recommendation ITU-T G.701 (1993), *Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms.*

[ITU-T G.711]   Recommendation ITU-T G.711 (1988), *Pulse code modulation (PCM) of voice frequencies.*

[ITU-T G.726]   Recommendation ITU-T G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM).*

[ITU-T G.728]   Recommendation ITU-T G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction.*

[ITU-T G.729]	Recommendation ITU-T G.729 (1996), *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)*.
[ITU-T H.245]	Recommendation ITU-T H.245 (2005), *Control protocol for multimedia communication*.
[ITU-T H.248.1]	Recommendation ITU-T H.248.1v3 (2005), *Gateway control protocol: Version 3*.
[ITU-T H.248.2]	Recommendation ITU-T H.248.2 (2005), *Gateway control protocol: Facsimile, text conversation and call discrimination packages*, plus Amendment 1 (2007), *Discriminated call type enhancement*.
[ITU-T H.248.39]	Recommendation ITU-T H.248.39 (2006), *Gateway control protocol: H.248 SDP parameter identification and wildcarding*.
[ITU-T H.323]	Recommendation ITU-T H.323 (2003), *Packet-based multimedia communications systems*.
[ITU-T T.30]	Recommendation ITU-T T.30 (2005), *Procedure for document facsimile transmission in the general switched telephone network*.
[ITU-T T.35]	Recommendation ITU-T T.35 (2000), *Procedure for the allocation of ITU-T defined codes for non-standard facilities*.
[ITU-T T.38]	Recommendation ITU-T T.38 (2004), *Procedures for real-time Group 3 facsimile communication over IP networks*.
[ITU-T T.120]	Recommendation ITU-T T.120 (1996), *Data protocols for multimedia conferencing*.
[ITU-T V.8]	Recommendation ITU-T V.8 (2000), *Procedures for starting sessions of data transmission over the public switched telephone network*.
[ITU-T V.18]	Recommendation ITU-T V.18 (2000), *Operational and interworking requirements for DCEs operating in the text telephone mode*.
[ITU-T V.21]	Recommendation ITU-T V.21 (1988), *300 bits per second duplex modem standardized for use in the general switched telephone network*.
[ITU-T V.34]	Recommendation ITU-T V.34 (1998), *A modem operating at data signalling rates of up to 33 600 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits*.
[ITU-T V.150.1]	Recommendation ITU-T V.150.1 (2003), *Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs*.
[IETF RFC 768]	IETF RFC 768 (1980), *User Datagram Protocol*.
[IETF RFC 791]	IETF RFC 791 (1981), *Internet Protocol DARPA Internet Program Protocol Specification*.
[IETF RFC 2198]	IETF RFC 2198 (1997), *RTP Payload for Redundant Audio Data*.
[IETF RFC 3261]	IETF RFC 3261 (2002), *SIP: Session Initiation Protocol*.
[IETF RFC 3264]	IETF RFC 3264 (2002), *An Offer/Answer Model with the Session Description Protocol (SDP)*.
[IETF RFC 3388]	IETF RFC 3388 (2002), *Grouping of Media Lines in the Session Description Protocol (SDP)*.
[IETF RFC 3389]	IETF RFC 3389 (2002), *Real-time Transport Protocol (RTP) Payload for Comfort Noise (CN)*.

[IETF RFC 3550]   IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications*.

[IETF RFC 4103]   IETF RFC 4103 (2005), *RTP Payload for Text Conversation*.

[IETF RFC 4566]   IETF RFC 4566 (2006), *SDP: Session Description Protocol*.

[IETF RFC 4733]   IETF RFC 4733 (2006), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*.

[IETF RFC 4756]   IETF RFC 4756 (2006), *Forward Error Correction Grouping Semantics in Session Description Protocol*.

[IETF RFC 5109]   IETF RFC 5109 (2007), *RTP Payload Format for Generic Forward Error Correction*.

3   Definitions

3.1   Terms defined elsewhere

This Recommendation uses the terms defined in [ITU-T G.701].

3.2   Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1   audio mode: In this mode, the channel processes speech signals. The mode may include the use of compression algorithms and other processing functions that are not suitable for the transport of modem or facsimile signals.

3.2.2   general switched telephone network (GSTN): This network includes ATM, PSTN, ISDN, wireless networks and private networks.

3.2.3   ITU-T H.248 gateway: A media gateway that complies with the ITU-T H.248x series of Recommendations.

3.2.4   media gateway (MG): The media gateway converts media provided in one type of network to the format required in another type of network. For example, a MG could terminate bearer channels from a switched circuit network (e.g., DS0s) and media streams from a packet network (e.g., RTP streams in an IP network). This gateway may be capable of processing audio, video and ITU-T T.120 multimedia signals alone or in any combination, and will be capable of full duplex media translations. The MG may also play audio/video messages and performs other interactive voice response (IVR) functions, or may perform media conferencing. For the purpose of this Recommendation, the term media gateway refers to a voice gateway.

3.2.5   media gateway controller (MGC): Controls the parts of the call state that pertain to connection control for media channels in a media gateway.

3.2.6   modem: The term modem in this Recommendation covers all ITU-T V-series modems and text telephones types covered in the annexes of [ITU-T V.18].

3.2.7   modem relay: The transport of modem data across a packet network using modem termination at the gateways.

3.2.8   MoIP gateway: A media gateway that is compliant with the ITU-T V.150x series of Recommendations.

3.2.9   off-ramp gateway: The IP network access point that calls an answering DCE. (Abbreviated to G2.)

3.2.10   on-ramp gateway: The access point that is called by an originating DCE that interfaces to the IP network. (Abbreviated to G1.)

3.2.11 transcoding: Translation from one type of encoded media format to another different media format (examples: ITU-T G.711 A-law to μ-law or vice versa, ITU-T G.711 codec to ITU-T G.726-40K, ITU-T G.711 to a broadband codec that operates at 256 kbit/s, etc.).

3.2.12 VBD gateway: A media gateway that is compliant with this Recommendation.

3.2.13 voice-band data mode: The transport of voice-band data over a voice channel of a packet network with the encoding appropriate for modem signals as defined in clause 6 of this Recommendation.

4   Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

ABNF	Augmented Backus-Naur Form
ACL	Answered Configuration (Codec) List
ANS	Answer Tone
/ANS	Answer tone with phase reversals
ASN.1	Abstract Syntax Notation One
ASNam	Answer Tone
CED	Facsimile CallED tone
CI	Call Indicator Signal
CNG	Facsimile Calling tone
DS0	Digital Signal, level 0
DSD	Data Signal Detector
DTMF	Dual Tone Multi-Frequency
FAX	Facsimile
FEC	Forward Error Correction
FoIP	Facsimile over Internet Protocol
G3FE	Group 3 Facsimile Equipment
GSTN	General Switched Telephone Network
IP	Internet Protocol
IVR	Interactive Voice Response
MG	Media Gateway
MGC	Media Gateway Controller
MoIP	Modem over Internet Protocol
MPS	Multiple Payload Stream
NCL	Negotiated Configuration (Codec) List
O/A	(SDP) Offer/Answer
OCL	Offered Configuration (Codec) List
OLC	Open Logical Channel
PCL	Preferred Configuration (Codec) List

PCMU	Pulse Code Modulation μ-law
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SCL	Supported Configuration (Codec) List
SCN	Switched Circuit Network
SDP	Session Description Protocol
SDP O/A	SDP Offer/Answer
SID	Silence Insertion Description
SIP	Session Initiation Protocol
SPRT	Simple Packet Relay Transport
SS7	Signalling System No. 7
SSE	State Signalling Events
TDM	Time Division Multiplex(ing)
ToIP	Text telephony over Internet Protocol
UDP	User Datagram Protocol
VBD	Voice-band Data
VoIP	Voice over Internet Protocol

5 Conventions

An ITU-T Recommendation, by definition, is not mandatory – compliance is voluntary. The use of the words "shall" and "must" and their negatives "shall not" and "must not" are to be used with care and sparingly. These words are only to be used to express mandatory provisions, when necessary, to give the Recommendation meaning and effect; i.e., if certain values and/or parts of a Recommendation are essential, and the Recommendation will have no meaning if these values and/or parts are not strictly respected or adhered to. Compliance with the Recommendation is achieved only when all mandatory provisions are met. However, the inclusion of mandatory provisions in a Recommendation does not imply, of itself, that compliance with the Recommendation is required of any party.

5.1 Recommendation version

For the purposes of forward and backward compatibility, this Recommendation is assigned a version number, which is defined here.

NOTE – The reader is encouraged to check on the ITU-T website for any normative or informative amendments to this Recommendation.

Version: 1

5.2 SDP Offer/Answer protocol variants

This Recommendation 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 to [IETF RFC 3264];

–   the name "revised SDP Offer/Answer" indicates SDP Offer/Answer according to [b-IETF RFC 5939] and [b-IETF Draft MediaCapNeg].

6   Definition of the VBD mode of operation

Voice-band data is the transport of modem, facsimile, and text telephony signals over a voice channel of a packet network with a codec appropriate for such signals.

For voice-band data (VBD) mode of operation, all voice-band modulated signal samples shall be transported across an IP network using the RTP protocol defined in [IETF RFC 3550].

When in VBD mode, an ITU-T V.152 compliant implementation shall:

–   use a codec that passes voice-band modulated signals with minimal distortion. This codec shall be assigned as the VBD codec with a specific RTP payload type which shall be negotiated with the remote ITU-T V.152 implementations as described in clause 7;

–   have end-to-end constant latency;

–   disable voice activity detection and comfort noise generation during the data transfer phase;

–   disable any DC removal filters that may be integral with the speech encoder used;

and should consider the appropriate application of:

–   forward error correction (FEC) (e.g., [IETF RFC 5109]) or other forms of redundancy (e.g., [IETF RFC 2198]) only if support has been successfully negotiated with the remote ITU-T V.152 implementation;

–   voice packet loss concealment techniques and algorithms that are suitable for modem and facsimile modulations.

6.1   Minimum requirements for VBD mode of operation

For purposes of interoperability, an ITU-T V.152 compliant implementation shall support at least both ITU-T G.711 A-law and ITU-T G.711 μ-law codecs as VBD codecs.

When negotiating the VBD codec, the initiating ITU-T V.152 implementation must include in the offer either PCMA or PCMU (or both) in the list of VBD codecs, though other VBD codecs may be additionally specified. The ITU-T V.152 implementation answering the offer must indicate support for at least one VBD codec, which need not be PCM-based.

Redundancy as per [IETF RFC 2198] and forward error correction as per [IETF RFC 5109] are supported options.

6.2   Echo canceller and VBD mode

Echo cancellation and VBD mode are independent functions in a VBD gateway. The requirement for echo cancellation is not a VBD characteristic, but depends on the propagation delay in combination with a source of echo present in the connection and refers primarily to voice. An ITU-T V.152 compliant implementation does, therefore, not need to provide echo cancellation.

If an echo canceller (EC) is used on the VBD channel, then the EC shall be compliant to [ITU-T G.168]. The gateway autonomous control of the EC shall thus be orthogonal to the ITU-T V.152 audio-VBD state transitioning.

[ITU-T G.168] implies the detection of inband events in both traffic directions, i.e., from circuit-switched and IP network side in case of a VBD gateway.

6.3 IP transport services for VBD

6.3.1 Non-assured VBD mode

The non-assured VBD transport service is defined by the mandatory capabilities of ITU-T V.152, i.e., according to clause 6.1,

– without RTP transport redundancy (according to [IETF RFC 2198]); and

– without forward error correction of RTP packets (according to [IETF RFC 5109]).

The non-assured VBD mode should be sufficient in case of "good" IP network conditions. The non-assured VBD mode may be operated in two sub-modes with respect to the transfer of inband signals, see the following subclauses.

6.3.1.1 Non-assured transport of modem inband signals

This is a non-assured VBD transport service,

– without RTP payload format for telephony events (according to [IETF RFC 4733]).

6.3.1.2 Assured transport of modem inband signals

[IETF RFC 4733] shall be used according to clause 8 (i.e., the IETF RFC 4733 NTE mode shall be used). Usage of IETF RFC 4733 packets implies a negotiation process with the peer PSTN gateway (see also clause 2.4.1 of [IETF RFC 4733]).

6.3.2 Assured VBD mode

The assured VBD mode may be enabled in case of unreliable IP transport conditions. The assured VBD mode shall then use [IETF RFC 4733] for modem inband signals, and [IETF RFC 2198] and/or [IETF RFC 5109] for modem data information. Usage of [IETF RFC 2198] and [IETF RFC 5109] implies a negotiation process with the peer PSTN gateway:

– IETF RFC 2198 SDP elements for negotiation: see clause 5 of [IETF RFC 2198];

– IETF RFC 5109 SDP elements for negotiation: see [IETF RFC 4756].

7 Negotiation of support of VBD and selection of VBD codec and other VBD enhanced functionality

Negotiation of the support and use of VBD data mode, as defined in this Recommendation, shall be carried out at call establishment during the initial exchange of the call capabilities of the endpoints establishing the call, particularly under the following conditions:

– if it is unclear whether the peer gateway is [ITU-T V.152] or non-[ITU-T V.152] compliant;

– if the same audio codec shall be used as ITU-T V.152 VBD codec, which implies the negotiation of a dynamic RTP PT value for the VBD codec;

– if it is already clear that a PSTN modem call may be expected (with a certain probability); and/or

– if ITU-T V.152 VBDoIP is preferred (by the IP service provider) above packet relay mechanisms (see also clause 7.1.2.1).

Mid-call negotiations may be carried out in exceptional cases, see clause 7.3.

Indication of such support entails assigning RTP payload types to VBD as well as the codecs.

The mechanisms for negotiation vary depending on the endpoint's capabilities exchange protocols used, which can be the session description protocol (defined in [IETF RFC 4566]) or [ITU-T H.245]; the call control protocol, such as those defined in [ITU-T H.323], and the session

initiation protocol (SIP), defined in [IETF RFC 3261]; and/or the media gateway control protocols such as those defined in [ITU-T H.248.1] and [b-ITU-T J.171].

This clause shall describe negotiation procedures for mechanisms that use:

– The session description protocol (SDP) defined in [IETF RFC 4566]), such as, but not limited to, SIP terminals/gateways and ITU-T H.248 gateways;

– [ITU-T H.245] that complies with [ITU-T H.323].

This Recommendation does not preclude the gateways from negotiating support of other mechanisms such as IETF RFC 4733 telephone-events, [ITU-T T.38], [ITU-T V.150.1] and/or text relay, for transporting non-voice signals. RTP shall be used for the transport of VBD.

7.1 Negotiation using the session description protocol

For implementations that use the session description protocol, the "gpmd" (general-purpose media descriptor) attribute shall be used to associate payload types in a media information ('m') line with VBD mode.

7.1.0 New SDP attributes

7.1.0.1 VBD support indication (a=gpmd:)

The general form of this attribute line is:

```
a=gpmd:<format> <parameter list>
```

In the context of VBD declaration, the `<format>` must be an RTP/AVP payload type. The `<parameter list>` is a semicolon-separated list of "parameter=value" pairs. For RTP/AVP formats, these pairs address parameters that are not part of their standard MIME definition. For sessions supporting this Recommendation, the parameter of interest is the Boolean 'vbd' that may have the value of 'yes' or 'no'. When set to 'yes', the attribute indicates that the implementation supports VBD mode as described in this Recommendation.

Omission of the "gpmd" attribute with a "vbd=yes" attribute/value pair for any codec in the SDP session description shall be construed as non-support of VBD mode operation as defined in this Recommendation.

Setting vbd=no is an explicit indication that the payload type will not be used for vbd.

Note that this is not the same as omitting the gpmd attribute with vbd.

The payload type marked for voice-band data (VBD) treatment should be a dynamic payload type. It is possible that a codec, such as PCMU, be declared with both static and dynamic payload types, with only one of the two marked for voice-band data use (see example 1 below). If a codec, such a PCMU or PCMA, is declared with only a static payload type, and is also marked for voice-band data use, then this codec must not be used for carrying voice (see example 2 below).

7.1.0.2 Packetization time (a=maxmptime:)

In addition to negotiating support of ITU-T V.152 and the corresponding RTP payload type, an ITU-T V.152 implementation should include the 'maxmptime' attribute (maximum multiple ptime) to indicate the supported packetization period for all codec payload types.

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

This attribute is a media-level attribute. The maxmptime attribute defines a list of maximum packetization time values, expressed in milliseconds, the endpoint is capable of using (sending and receiving) for this connection. There shall be precisely one entry in the list for each `<format>` entry provided in the "m=" line. Each entry is separated by a space. Entry number j in this list defines the maximum packetization time for entry number j in the "m=" line. The first entry in the list shall be a

decimal number whereas subsequent entries in the list shall be either a decimal number or a hyphen. For those media formats where a single maximum packetization rate does not apply (e.g., non-voice codecs such as telephone-event or comfort noise), a hyphen ("-") shall be included at the corresponding location in the list of packetization periods.

When receiving an SDP session description, the maxmptime attribute conveys the list of maximum packetization periods that the remote endpoint is capable of using for this connection; one for each media format in the "m=" line. For media formats whose packetization period is specified as a hyphen ("-"), the VBD gateway shall use one of the maximum packetization periods that was actually specified in the list.

The "a=ptime" attribute, defined in [IETF RFC 4566], shall be ignored if the SDP session description contains the "maxmptime" attribute.

If the "maxmptime" attribute is absent, then the value of the "ptime" attribute, if present, shall be taken as indicating the packetization period for all codecs present in the "m=" line.

If neither the 'ptime' nor 'maxmptime' attribute are present in the SDP session description, then an ITU-T V.152 implementation shall assume the default packetization period defined in [IETF RFC 3550] (which is 20 ms for ITU-T G.711 and ITU-T G.726-32k). An ITU-T V.152 implementation shall not transmit ITU-T V.152 packets with a packetization time greater than the one offered by the remote end.

7.1.0.3   Examples

Several application scenario examples are provided below.

7.1.0.3.1 Example 1 – Multiple audio and multiple VBD codec types

Below is an SDP-related example that indicates support of ITU-T V.152 as per this Recommendation. For clarity purposes, the example only shows the media descriptions of the SDP session description.

Table 1 provides the encoding of example 1 using Legacy SDP Offer/Answer syntax.

Figure 1 illustrates the four possible media configurations, offered in this example.

Table 2 provides the encoding of example 1 in Revised SDP Offer/Answer syntax.

Table 1 – Example SDP encoding – "Example 1" offer in Legacy SDP Offer/Answer syntax

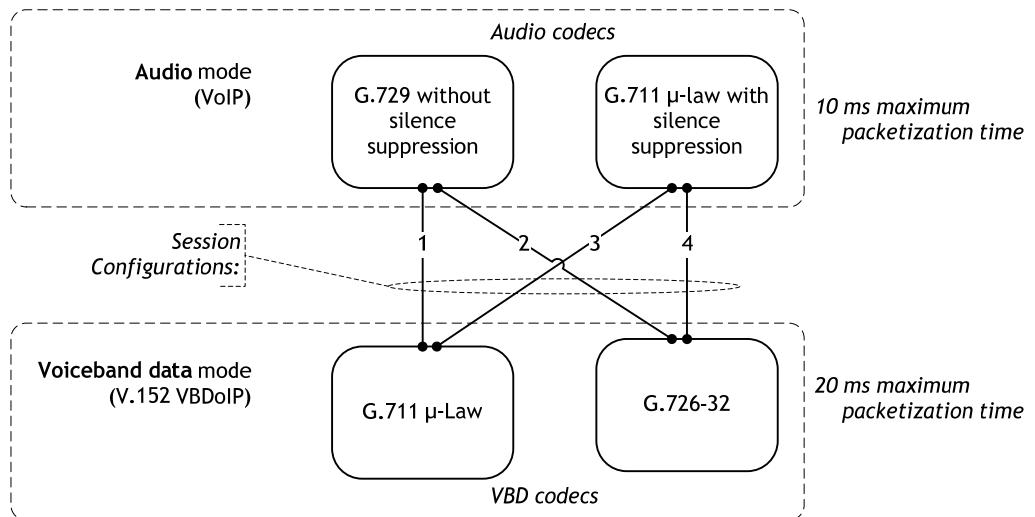| SDP encoding (shortened SDP description)	Comments
```m=audio 3456 RTP/AVP 18 0 13 96 98 99``` ```a=maxmptime:10 10 - - 20 20``` ```a=rtpmap:96 telephone-event/8000``` ```a=fmtp:96 0-15, 34, 35``` ```a=rtpmap:98 PCMU/8000``` ```a=gpmd:98 vbd=yes``` ```a=rtpmap:99 G726-32/8000``` ```a=gpmd:99 vbd=yes```	In the example, static payload type '0' and dynamic payload type '98' each represent the encoding format 'PCMU'. The payload type '0' is not associated with VBD. The payload types '98' (PCMU) and '99' (32 kbit/s ADPCM) are, however, associated with VBD. Concerning the maximum packetization times for each payload type: Voice packets use 10 ms, VBD packets use 20 ms, and a dash is assigned to payload types 13 (silence indication packets) and 96 (IETF RFC 4733) packets) indicating that a maximum ptime is not applicable or necessary.

Figure 1 – Potential media configurations offered in example 1

**Table 2 – Example SDP encoding – "Example 1" offer in
Revised SDP Offer/Answer syntax**

SDP encoding (shortened SDP description)	Comments

```; SESSION CONFIGURATIONs```
 ```a=sescap:1 1,3 ; VoIP = G729, VBDoIP =```
 ```V.152 (PCMU)```
 ```a=sescap:2 1,4 ; VoIP = G729, VBDoIP =```
 ```V.152 (G726)```
 ```a=sescap:3 2,3 ; VoIP = PCMU, VBDoIP =```
 ```V.152 (PCMU)```
 ```a=sescap:4 2,4 ; VoIP = PCMU, VBDoIP =```
 ```V.152 (G726)```
 	
 ```; ACTUAL CONFIGURATION (due to backward```
 ```compatibility)```
 ```m=audio 3456 RTP/AVP 18 0 13 96 98 99```
 ```a=maxmptime:10 10 - - 20 20```
 ```a=rtpmap:96 telephone-event/8000```
 ```a=fmtp:96 0-15, 34, 35```
 ```a=rtpmap:98 PCMU/8000```
 ```a=gpmd:98 vbd=yes```
 ```a=rtpmap:99 G726-32/8000```
 ```a=gpmd:99 vbd=yes```
 ```;```
 ```; POTENTIAL CONFIGURATIONs```
 ```a=tcap:1 RTP/AVP ; transport for VoIP &```
 ```VBDoIP```
 ```a=acap:1 ptime:10   ; for Audio mode```
 ```a=acap:2 ptime:20   ; for VBD mode```
 ```a=mcap:1 G729/8000   ; audio codec 1```
 ```a=mcap:2,5 PCMU/8000 ; audio codec 2 & VBD```
 ```codec 1```
 ```a=mcap:3 CN/8000   ; comfort noise for```
 ```audio c. 2```
 ```a=mcap:4 telephone-event/8000 ; NTE codec```
 ```a=mcap:6 G726-32/8000 ; VBD codec 2```
	Offered (4) session configurations:
	 • Preference 1: Audio (ITU-T G.729), VBD (ITU-T V.152 PCMU) & NTE ([IETF RFC 4733])
	 • Preference 2: Audio (ITU-T G.729), VBD (ITU-T V.152 G.726-32) & NTE ([IETF RFC 4733])
	 • Preference 3: Audio (PCMU including silence suppression), VBD (ITU-T V.152 PCMU) & NTE ([IETF RFC 4733])
	 • Preference 4: Audio (PCMU including silence suppression), VBD (ITU-T V.152 G.726-32) & NTE ([IETF RFC 4733])
	 NOTE – The "a=-ms" operator is used to remove all legacy attribute lines on session and media description level.
	 The two audio and two VBD settings are defined as individual potential configurations, in order to assign individual packetization times. This approach allows to avoid usage of "*a=maxmptime:*" parameter. The concept of session configurations is applied for the four audio/VBD combinations.

**Table 2 – Example SDP encoding – "Example 1" offer in
Revised SDP Offer/Answer syntax**

SDP encoding (shortened SDP description)	Comments

```a=mfcap:4 0-15,32-35 ; value range DTMF & VBD stimuli```
 ```a=mscap:5 gpmd vbd=yes ; for V.152 PCMU```
 ```a=mscap:6 gpmd vbd=yes ; for V.152 G726-32```
 ```a=pcfg:1 t=1 a=-ms:1 m=1,4 pt=1:18,4:96```
 ```a=pcfg:2 t=1 a=-ms:1 m=2,3,4 pt=2:0,3:13,4:96```
 ```a=pcfg:3 t=1 a=-ms:2 m=5 pt=5:98```
 ```a=pcfg:4 t=1 a=-ms:2 m=6 pt=6:99```	

The following main advantages may be noted concerning Revised versus Legacy SDP Offer/Answer:

– the ITU-T V.152 introduced SDP attribute "*a=maxmptime:*" may be replaced by IETF SDP elements;

– there is an explicit indicated, preferred order with regard to the offered media configurations (this is missing in Legacy SDP Offer/Answer: it is unclear, e.g., whether configuration 2 is preferred over configuration 3 or vice versa); and

– the answerer in Legacy SDP Offer/Answer could select both audio codecs and/or both VBD codecs (which may require subsequent Offer/Answer cycles).

7.1.0.3.2 Example 2 – Single audio codec, two VBD codec options

Table 3 provides the encoding of example 2 in Legacy SDP Offer/Answer syntax.

Figure 2 illustrate the two possible media configurations, offered in this example.

Table 3 – Example SDP encoding – "Example 2" offer in Legacy SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments

```m=audio 3456 RTP/AVP 0 18 98```
 ```a=gpmd:0 vbd=yes```
 ```a=rtpmap:98 G726-32/8000```
 ```a=gpmd:98 vbd=yes```
 ```a=ptime:20```
	In this example, the static payload type '0' (PCMU) is marked for VBD treatment, along with the dynamic payload type '98' (mapped to 32 kbit/s ADPCM). Thus, payload type '0' must not be used for carrying voice. It also indicates that the VBD gateway can receive voice and VBD packets with a size of 20 ms.

Figure 2 – Potential media configurations offered in example 2

NOTE – The use of static payload types for VBD is strongly discouraged because there is a danger that a non-ITU-T V.152 system would see a proposal with, say, [ITU-T G.711] VBD and [ITU-T G.729]. However, not understanding the VBD attributes, it may consider ITU-T G.711 as a valid audio codec. However, network operators may prefer that [ITU-T G.711] not be utilized, except in the case where VBD is necessary, and that all voice shall be [ITU-T G.729]. To illustrate this point, consider this offer:

```
m=audio 15400 RTP/AVP 0 18
a=gpmd:0 vbd=yes
```

and consider this answer:

```
m=audio 15400 RTP/AVP 0 18
```

The systems would then, most likely, communicate using [ITU-T G.711], rather than the intended [ITU-T G.729], for voice. This is a possible issue and inherent to Legacy SDP Offer/Answer protocol. This problem may be avoided by explicit potential configuration in Revised SDP Offer/Answer, see Table 4.

Table 4 – Example SDP encoding – "Example 2" offer in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments

<pre>; ACTUAL CONFIGURATION (due to backward
compatibility)
m=audio 3456 RTP/AVP 0 18 98
a=gpmd:0 vbd=yes
a=rtpmap:98 G726-32/8000
a=gpmd:98 vbd=yes
a=ptime:20
;
; POTENTIAL CONFIGURATIONs
a=tcap:1 RTP/AVP ; transport for VoIP &
VBDoIP
a=acap:1 ptime:20 ; common packetization
time
a=mcap:1 G729/8000   ; audio codec
a=mcap:2 PCMU/8000   ; VBD codec 1
a=mcap:3 G726-32/8000   ; VBD codec 2
a=mscap:2 gpmd vbd=yes ; for V.152 PCMU
a=mscap:3 gpmd vbd=yes ; for V.152 G726-32
a=pcfg:1 t=1 a=-ms:1 m=1,2 pt=1:18,2:0
a=pcfg:2 t=1 a=-ms:1 m=1,3 pt=1:18,3:98</pre>
	Offered (2) potential configurations:
	• Preference 1:
	 Audio (ITU-T G.729), VBD
	 (ITU-T V.152 PCMU)
	• Preference 2:
	 Audio (ITU-T G.729), VBD
	 (ITU-T V.152 G.726-32)
	
	If answerer supports Revised SDP Offer/Answer protocol, then either potential configuration 1 or 2 is selected, but not both.

7.1.1   Mechanism for indicating support of ITU-T V.152 using ITU-T H.248/Megaco

Under ITU-T H.248, the media gateway controller (MGC) uses local and remote descriptors to reserve and commit MG resources for media decoding and encoding for the given stream(s) and termination to which they apply. The MG includes these descriptors in its response to indicate what it is actually prepared to support. When text encoding the protocol, the descriptors consist of SDP session descriptions that describe the call capabilities.

Support of ITU-T V.152 shall only be applied on ephemeral terminations via the local and/or remote descriptors.

For an MG to reserve and commit resources for more than one call capability alternative, the MGC must set the ReserveGroup and ReserveValue properties of the LocalControlDescriptor to 'True'.

Thus, if a list of payload types is offered in a local and/or remote descriptor, such as the following example 3 of an Add ephemeral termination command illustrates (the same applies if the command was a Modify or Move), the media gateway will select from the list only those payloads for which it can reserve and commit resources and shall send a reply to the MGC containing the alternatives for the local and/or remote descriptor that it selected, as described in [ITU-T H.248.1]:

Table 5 – Example ITU-T H.248 encoding (example 3a)

ITU-T H.248 text encoding	Comments

<pre>1) MGC request (MGC to MG):
MEGACO/3 [123.123.123.4]:55555
Transaction = 11 {
  Context = $ {
   Add = $ {
    Media {
     Stream = 1 {
      LocalControl {</pre>
	The MGC requests support for four media configurations:
	• two audio configurations ('G.729' and 'PCMU' codec types); and
	• two VBDoIP configurations (VBD codec types 'PCMU' and 'G.726-32').

Table 5 – Example ITU-T H.248 encoding (example 3a)

ITU-T H.248 text encoding	Comments

<pre>   Mode = RecOnly,
      ReserveGroup = ON,
      ReserveValue = ON
     },
     Local {
     v=0
     c=IN IP4 $
     m=audio $ RTP/AVP 18 0 98 99
     a=rtpmap:98 PCMU/8000
     a=gpmd:98 vbd=yes
     a=rtpmap:99 G726-32/8000
     a=gpmd:99 vbd=yes
     }; IP termination for audio and VBD
    }
   }
  }
 }
}</pre>	

<pre>2) MG reply (MG to MGC response):
 	
MEGACO/3 [123.123.123.4]:55555
Transaction = 11 {
  Context = 34444 {
   Add = Te/1 {
    Media {
     Stream = 1 {
      LocalControl {
       Mode = RecOnly,
       ReserveGroup = ON,
       ReserveValue = ON
      },
      Local {
      v=0
      c=IN IP4 11.09.19.65
      m=audio 1970 RTP/AVP 18 0 98 99
      a=rtpmap:98 PCMU/8000
      a=gpmd:98 vbd=yes
      a=rtpmap:99 G726-32/8000
      a=gpmd:99 vbd=yes
      }; IP termination for audio and VBD
     }
    }
   }
  }
}</pre>	

Alternatively, an MGC may leave it up to the MG whether it wants to indicate that it supports VBD, as per this Recommendation, and to select its dynamic payload type for VBD mode of operation by including CHOOSE (i.e., $) in the payload type list field as example 3b illustrates:

Table 6 – Example ITU-T H.248 encoding (example 3b)

H.248 text encoding	Comments

1) MGC request (MGC to MG):
`MEGACO/3 [123.123.123.4]:55555`
`Transaction = 11 {`
  `Context = $ {`
   `Add = $ {`
    `Media {`
     `Stream = 1 {`
      `LocalControl {`
      `Mode = RecOnly,`
      `ReserveGroup = ON,`
      `ReserveValue = ON`
      `},`
      `Local {`
      `v=0`
      `c=IN IP4 $`
      `m=audio $ RTP/AVP 18 0 $ ; Wildcard`
                 `; in format list`
     `}; IP termination for audio and VBD`
    `}`
    `}`
   `}`
  `}`
`}`
	The MGC applies the wildcard CHOOSE ('$') in the media description:
	• the wildcard is in the format list (see also clause 6.11 of [ITU-T H.248.39]);
	• such an underspecification implies "*a priori* knowledge" at the MG side, in order to choose an "appropriate" media format;
	• e.g., it is expected that the MG selects a VBDoIP format, but not anything else (like, e.g., another audio codec type);
	• such kind of behaviour is either under the responsibility of the network management or could be defined in an ITU-T H.248 profile specification.

2) MG reply (MG to MGC response):
`MEGACO/3 [123.123.123.4]:55555`
`Transaction = 11 {`
  `Context = 34444 {`
   `Add = Te/1 {`
    `Media {`
     `Stream = 1 {`
      `LocalControl {`
      `Mode = RecOnly,`
      `ReserveGroup = ON,`
      `ReserveValue = ON`
      `},`
      `Local {`
      `v=0`
      `c=IN IP4 11.09.19.65`
      `m=audio 1970 RTP/AVP 18 0 98 99`
      `a=rtpmap:98 PCMU/8000`
      `a=gpmd:98 vbd=yes`
      `a=rtpmap:99 G726-32/8000`
      `a=gpmd:99 vbd=yes ; The  MGC should`
        `; supply sufficient information`
        `; for unambiguous resource`
        `; selections by the MG,`
        `; see clause 7.1.8 of`
        `; [ITU-T Rec. H.248.1] Version 3`
     `}; IP termination for audio and VBD`
	The MG selects two VBDoIP configurations (besides the two audio configurations 'G.729' and 'PCMU'):
	• the selection of multiple media configurations is due to *ReserveValue* 'True';
	• the MGC knows now by this response that the MG supports the two requested audio formats and furthermore two VBDoIP configurations with VBD codecs 'PCMU' and 'G.726-32';
	• it is expected that there will be a further MGC/MG request/response cycle in order to enable (at least) just a single VBDoIP configuration (because multiple VBD configurations (but also multiple audio configurations) are typically meaningless from service perspective);
	• thus, the MGC would use *ReserveValue* 'False' in the subsequent request.

Table 6 – Example ITU-T H.248 encoding (example 3b)

H.248 text encoding	Comments
} } } } }	

Once an MG has acknowledged a set of call capability alternatives, the MG is requested to reserve resources so that it can decode or encode the media stream according to any of the alternatives. Thus, in the above example 3a, if the MG supports [ITU-T G.729] and [ITU-T G.711] for audio and [ITU-T G.711] for VBD, (as per this Recommendation), then, in accordance with [ITU-T H.248.1], the MG must reserve resources such that it can decode one RTP stream in any of the formats in its response at any time during the call, i.e., ITU-T G.711 audio format, ITU-T G.729 audio format or ITU-T G.711 VBD format.

If a specific relay mechanism (e.g., ITU-T T.38, ITU-T V.150.1, etc.) is indicated as the preferred mechanism above that of VBD, then, for the applicable devices, the relay mechanisms shall be used instead of VBD. For example, if a remote descriptor indicates ITU-T T.38 as preferred over VBD, then an MG shall use ITU-T T.38 for all G3FE equipment instead of VBD.

This is a request for multiple media groups. There must be an order of preference. The preference may be either the "descending order rule" of ITU-T H.248 in case of a complete parameter specification by the MGC concerning the preferred media names, or a corresponding rule shall be provisioned in both ITU-T H.248 MG peers in case of "parameter under specification" (in ITU-T H.248 descriptors) by MGC.

If an MG cannot guarantee that it can commit and reserve the resources for VBD for the call being set up, then, in accordance with [ITU-T H.248.1], it shall not include the 'gpmd' attribute (that indicates support of ITU-T V.152) in its response SDP session description.

Note that this mechanism does not preclude an ITU-T H.248 MG implementation from sending to the MGC observedEvents indicating signals detected, as described in [ITU-T H.248.2] (e.g., call type discrimination package).

7.1.2   Mechanism for indicating support of ITU-T V.152 using SIP

A gateway answering an SDP session description offer that shows the capability to perform the relay model described in [IETF RFC 3264] shall be used to earmark one or more RTP payload types for VBD operation as defined in this Recommendation.

Just as a SIP-compliant terminal would indicate support of more than one audio codec payload or support of other payload types (e.g., [IETF RFC 4733] for DTMF relay) within a media stream, a SIP-compliant implementation shall indicate support of [ITU-T V.152] by including the payload types as described in clause 7.1.

If multiple media descriptions are being offered and if the implementations cannot support simultaneous reception and transmission of the various media types, then the 'group', 'mid' and 'FID' attributes described in [IETF RFC 3388] shall be used to indicate alternative support of each of the offered media types (as illustrated below in example 5).

Once a gateway has indicated support of [ITU-T V.152] in addition to other mechanisms within an SDP session description (such as, but not limited to, audio, facsimile relay via ITU-T T.38 DTMF relay as per [IETF RFC 4733], etc.), the gateway shall be capable of switching between any of the supported, and mutually negotiated, RTP payload types, at any time during a call.

7.1.2.1 Mechanism for indicating preference of VoIP relay mechanisms above VBD

SIP currently does not have a mechanism for indicating in a clear manner that a gateway would like to use a specific relay mechanism (e.g., ITU-T T.38, ITU-T V.150.1, text relay) instead of VBD. Hence, this clause defines the syntax and use of an attribute that indicates a preference list of modem and facsimile transport methods in an ITU-T V.152 implementation that supports any of the following alternative transport methods:

– facsimile relay of IP via [ITU-T T.38];

– modem relay over IP via [ITU-T V.150.1];

– text relay.

7.1.2.1.1 SDP attribute for transport method indication (a=pmft:)

The 'pmft' attribute and its formatting in the SDP session description is described by the following ABNF syntax:

```
pmft-attribute          =   "a=pmft:" *(SPACE modem-fax-transport)
modem-fax-transport     =   1* ("V1501" / "T38" / "V151")
```

This attribute allows an ITU-T V.152 implementation to indicate whether it prefers any of the listed relay transport mechanism above the VBD mode. Omission of this attribute in an SDP session description means that VBD mode is the preferred transport mechanism of voice-band data.

When included in an SDP session description, this attribute shall always be placed at the session level.

For example, an ITU-T V.152 implementation that also supports [ITU-T V.150.1] for modems and [ITU-T T.38] for facsimile, and prefers to use these relay mechanisms whenever possible instead of VBD, shall include, at the session level of the SDP session description, the following 'pmft' attribute:

```
a=pmft: T38 V1501
```

An ITU-T V.152 implementation that receives the above 'pmft' attribute, and is able to support both the relay mechanisms specified in this example, shall include the same 'pmft' attribute in its response. Thus, when the call is set up, all G3FE shall be transported via [ITU-T T.38]; voice-band modems that are supported by [ITU-T V.150.1] shall be transported via [ITU-T V.150.1] and all other modems (e.g., text telephones) shall be transported by [ITU-T V.152].

A gateway answering an SDP session description offer that includes the 'pmft' attribute, if indicating preference of the supported relay mechanism over VBD, shall include in the response SDP session description the 'pmft' attribute with the relay mechanism specified. If a relay mechanism is not supported, then that relay mechanism shall be removed from the list of the pmft attribute.

Once a specific relay mechanism (e.g., ITU-T T.38, ITU-T V.150.1, etc.) is indicated as the preferred mechanism above that of VBD, such relay mechanisms shall be used instead of VBD.

A gateway answering an SDP session description offer that shows the capability to perform relay mechanisms but does not include the 'pmft' attribute may include in the response SDP session description the 'pmft' attribute with the relay mechanisms specified to show a preference to use these.

7.1.2.1.2 Examples – Preferences VBDoIP (ITU-T V.152) versus FoIP (ITU-T T.38)

For example: if the initial SDP session description offer from a gateway that only supports [ITU-T V.152] and [ITU-T T.38] does not include the 'pmft' attribute because it prefers to use VBD above ITU-T T.38, then it would include an SDP session description such as the one described in Table 7 for in Legacy SDP Offer/Answer syntax.

Table 7 – Example SDP encoding – O/A cycle in Legacy SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments
1) OFFER: v=0 o=Offerer 0 0 IN IPV4 <IPAddressA> s=- t=0 0 p=+1 965 1109-1965 c=IN IP4 <IPAddressA> a=group:FID 1 2 m=audio <udpPort x> RTP/AVP 18 0 13 96 a=mid:1 a=ptime:10 a=rtpmap:96 PCMU/8000 a=gpmd: 96 vbd=yes m=image <udpPort y> udptl t38 a=mid:2 a=T38version:0 a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy	[IETF RFC 3388] is required in Legacy SDP Offer/Answer in order to indicate two media configurations in parallel: 1) audio VoIP or ITU-T V.152 VBDoIP, and 2) ITU-T T.38 FoIP. The example offer provides two audio codecs for VoIP.
2) ANSWER: v=0 o=Answerer 0 0 IN IPV4 <IPAddressB> s=- t=0 0 p=+1 965 0203-1970 c=IN IP4 <IPAddressB> a=group:FID 1 2 a=pmft: T38 m=audio <udpPort x> RTP/AVP 18 0 13 96 a=mid:1 a=ptime:10 a=rtpmap:96 PCMU/8000 a=gpmd: 96 vbd=yes m=image <udpPort y> udptl t38 a=mid:2 a=T38version:0 a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy	However, the answerer gateway (that may not be in as reliable a network) may always prefer ITU-T T.38 over VBD for facsimile transmission. Thus, it would include in its response, e.g., such an answer.

Figure 3 illustrates the four possible media configurations, offered in this example, and positively acknowledged by the answerer.
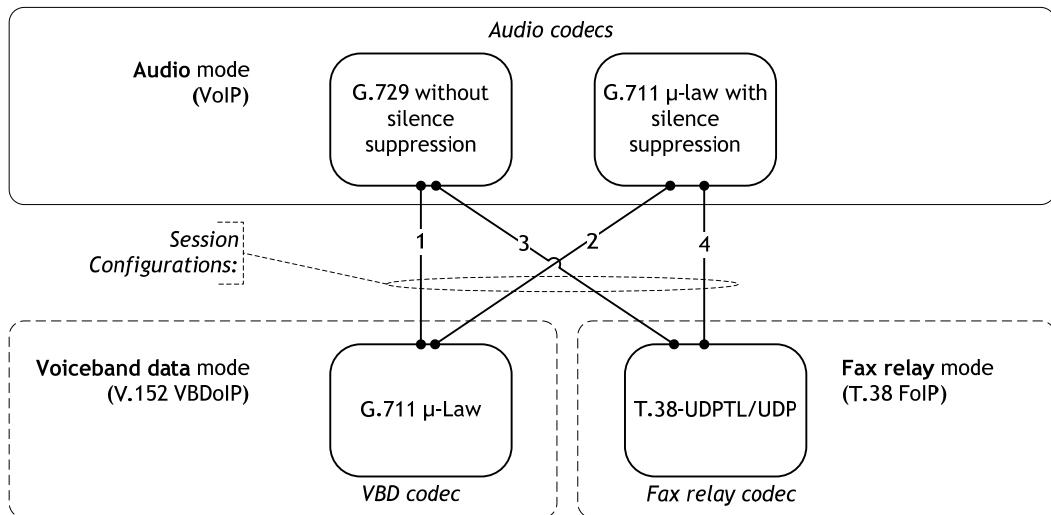

Figure 3 – Potential session configurations offered in this example

The offerer gateway, on receipt of such an answer, shall transport G3FE data using ITU-T T.38, i.e., potential configuration 3 or 4) and all other modems, or non-G3FE must be transported via ITU-T V.152 (i.e., potential configuration 1 or 2). There are still two audio options (due to two audio codecs). Such a situation is unfortunate (because it is not required for VoIP applications) and may be resolved by a subsequent negotiation cycle. Alternatively, a single cycle negotiation, using Revised SDP Offer/Answer, may be applied (Table 8).

Table 8 – Example SDP encoding: O/A cycle in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments

```1) OFFER:```
```; SESSION CONFIGURATIONs```
```a=sescap:1 1   ; VoIP = G.729, VBDoIP =```
```V.152 (G.711)```
```a=sescap:2 2   ; VoIP = G.711, VBDoIP =```
```V.152 (G.711)```
```a=sescap:3 3,5 ; VoIP = G.729, FoIP = T.38```
```UDPTL/UDP```
```a=sescap:4 4,5 ; VoIP = G.711, FoIP = T.38```
```UDPTL/UDP```
 	
```; LATENT CONFIGURATION for T.38```
```a=tcap:2 udptl   ; T.38 FoUDPTL/UDP```
```transport variant```
```a=mcap:5 t38   ; T.38 FoIP codec (subtype```
```= 't38')```
```a=acap:11 T38FaxVersion:0```
```a=acap:12```
```T38FaxRateManagement:transferredTCF```
```a=acap:13 T38FaxUdpEC:t38UDPRedundancy```
```a=acap:14 (… additional T.38 attributes may```
```be included)```
```a=lcfg:5 mt=image t=2 m=5 a=11,12,13,14,…```
	[IETF RFC 3388] is not required! Potential parallel media configurations (for ITU-T T.38 if G3FE stimuli are detected) are specified via the concept of session configurations in Revised SDP Offer/Answer. The ITU-T T.38 media configuration relates to a latent configuration because call may start with speech phase first:
	Offered (2) potential session configurations: see Figure 3.

Table 8 – Example SDP encoding: O/A cycle in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments
; ACTUAL CONFIGURATION (due to backward compatibility) … *ommitted, see Table 7* … ; ; POTENTIAL CONFIGURATIONs a=tcap:1 RTP/AVP ; transport for VoIP and VBDoIP a=acap:1 ptime:10 a=mcap:1 G729/8000 ; audio codec 1 a=mcap:2,4 PCMU/8000 ; audio codec 2 and VBD codec a=mcap:3 CN/8000 ; comfort noise for audio c. 2 a=mscap:4 gpmd vbd=yes ; for V.152 PCMU a=pcfg:1 t=1 a=-ms:1 m=1,4 pt=1:18,4:96 a=pcfg:2 t=1 a=-ms:1 m=2,3,4 pt=1:0,3:13,4:96 a=pcfg:3 t=1 a=-ms:1 m=1 pt=1:18 a=pcfg:4 t=1 a=-ms:1 m=2,3 pt=1:0,3:13	
2) ANSWER: …	Answerer would, e.g., just acknowledge session configuration 1 (for ITU-T V.152, non-G3FE modem) and 3 (for ITU-T T.38 G3FE emulation) if [ITU-T G.729] is supported (and thus the selected audio codec).

The following main advantages may be noted concerning Revised versus Legacy SDP Offer/ Answer:

– the ITU-T V.152 introduced SDP attribute "*a=pmft:*" may be replaced by IETF SDP syntax ("order principle within session and potential configurations in Revised SDP Offer/Answer");

– [IETF RFC 3388] is not required because parallel media configurations are specified via the concept of session configurations in Revised SDP Offer/Answer; and

– single cycle negotiation (because Answerer would already select 1-out-of-2 audio codecs).

7.1.3 Examples of indicating support for ITU-T V.152 using the session description protocol

This clause shall provide a few examples of SDP session descriptions sent by implementations that support ITU-T V.152 in addition to other Recommendations (such as, but not limited to, voice, [ITU-T T.38], ToIP, and [ITU-T V.150.1]).

7.1.3.1 Example 4 – Two audio codecs and one VBD codec (overlapping with audio)

An implementation that supports ITU-T V.152 (using the dynamic payload type 96 and ITU-T G.711 μ-law as the VBD codec) and the voice codecs ITU-T G.711 μ-law, silence suppression and ITU-T G.729, shall transmit the following SDP session description, only those lines that are relevant to this Recommendation are highlighted in bold, see Table 9.

Figure 4 illustrates the two possible media configurations, offered in this example.

Table 9 – Example SDP encoding – "Example 4" offer in Legacy SDP Offer/Answer syntax

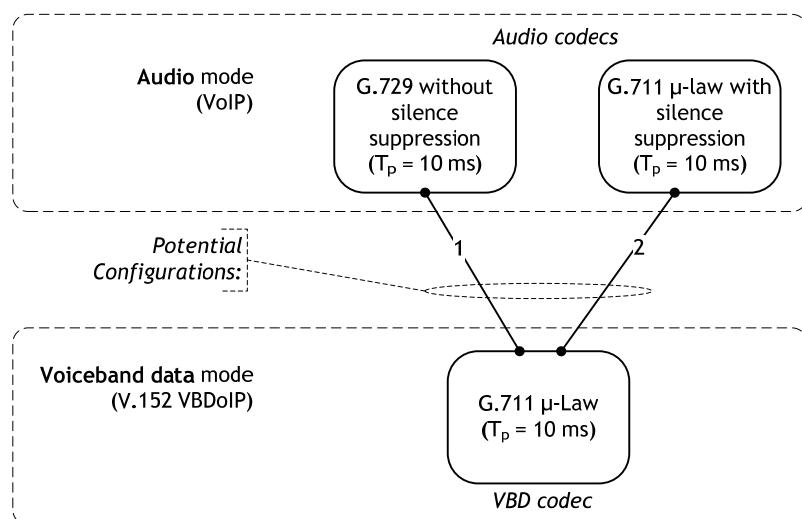| SDP encoding (shortened SDP description)	Comments

<pre>1) OFFER:
v=0
o=- 0 0 IN IPV4 <IPAddressA>
s=-
t=0 0
p=+1
c=IN IP4 <IPAddressA
m=audio <udpPort A> RTP/AVP 18 0 13 96
a=ptime:10
a=rtpmap:96 PCMU/8000
a=gpmd: 96 vbd=yes</pre>	

Figure 4 – Potential media configurations offered in this example

An ITU-T V.152 implementation that receives an SDP session description, as in the above example, shall interpret it as the remote gateway's capability to support ITU-T V.152 and that the payload type that shall be used for VBD packets is 96.

The Legacy SDP Offer is unclear whether one or two audio codecs may be selected, e.g., the answerer may agree to both audio codecs ("which is for the emulation of PSTN modem calls not required"). This ambiguity may be avoided by the concept of potential configurations in Revised SDP Offer/Answer (see Table 10).

Table 10 – Example SDP encoding – "Example 4" offer in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments
```; ACTUAL CONFIGURATION (due to backward compatibility) m=audio <udpPort A> RTP/AVP 18 0 13 96 a=ptime:10 a=rtpmap:96 PCMU/8000 a=gpmd: 96 vbd=yes ; ; POTENTIAL CONFIGURATIONs a=tcap:1 RTP/AVP ; transport for VoIP and VBDoIP a=acap:1 ptime:10 ; common packetization time a=mcap:1 G729/8000 ; audio codec 1 a=mcap:2,4 PCMU/8000 ; audio codec 2 and VBD codec 1 a=mcap:3 CN/8000 ; comfort noise for audio c. 2 a=mscap:4 gpmd vbd=yes ; for V.152 PCMU a=pcfg:1 t=1 a=-ms:1 m=1,4 pt=1:18,4:96 a=pcfg:2 t=1 a=-ms:1 m=2,3,4 pt=1:0,3:13,4:96```	Offered (2) potential configurations: • Preference 1: Audio (ITU-T G.729), VBD (ITU-T V.152 PCMU) • Preference 2: Audio (PCMU incl. silence suppression), VBD (ITU-T V.152 PCMU)

7.1.3.2 Example 5 – ITU-T V.152 to non-ITU-T V.152 (OCL with ITU-T V.152 and [ITU-T T.38], ACL without ITU-T V.152)

A call is set up between gateway A that supports, the Supported Configuration (Codec) List SCL_A, ITU-T V.152 with PCMU as VBD codec, ITU-T T.38 UDPTL/UDP, and voice codecs ITU-T G.729 and PCMU with [IETF RFC 3389] silence suppression and voice codec ITU-T G.729 and PCMU, and a Gateway B that supports (SCL_B) ITU-T T.38 UDPTL/UDP, silence suppression and voice codecs ITU-T G.729 and PCMU inclusive silence suppression, but does not support ITU-T V.152.

NOTE – Voice codec ITU-T G.729 is used here without silence suppression. This would require support of ITU-T G.729B and the correspondent SDP indication via attribute line "*a=fmtp:18 annexb=yes*".

The SDP transmitted by gateway A, the Offered Configuration (Codec) List OCL_A (= SCL_A), will be of the form described in the "Offer" in Table 11.

Gateway B, which does not support VBD shall respond with an SDP, the Answered Configuration (Codec) List ACL_B (= SCL_B), that omits all reference to ITU-T V.152 (see Answer in Table 11).

Table 11 – Example SDP encoding – O/A cycle in Legacy SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments
1) OFFER: v=0 o=GatewayA 0 0 IN IPV4 \<IPAddressA> s=- t=0 0 p=+1 c=IN IP4 \<IPAddressA> a=group:FID 1 2 m=audio \<udpPort x> RTP/AVP 18 0 13 96 a=mid:1 a=ptime:10 a=rtpmap:96 PCMU/8000 a=gpmd: 96 vbd=yes m=image \<udpPort y> udptl t38 a=mid:2 a=T38version:0 a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy (…….additional T.38 attributes may follow…..)	
2) ANSWER: v=0 o=GatewayB 0 0 IN IPV4 \<IPAddressB> s=- t=0 0 p=+1 c=IN IP4 \<IPAddressB> a=group:FID 1 2 m=audio \<udpPort w> RTP/AVP 18 0 13 a=mid:1 a=ptime:10 m=image \<udpPort z> udptl t38 a=mid:2 a=T38version:0 a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy (…….additional T.38 attributes may follow…..)	

On receipt of the above SDP, gateway A shall understand that gateway B does not perform ITU-T V.152. Thus, gateway A shall not transition into a VBD mode. The agreed Negotiated Configuration (Codec) List NCL is thus equal to ACL_B (= SCL_B). The NCL contains both audio codecs, which might be a problem, at least an unfortunate and not desired situation for PSTN modem call types.

Such an (single cycle O/A) NCL may be avoided by the usage of Revised SDP Offer/Answer (see Table 12), which allows to specify an OCL as Preferred Configuration (Codec) List PCL$_A$. Figure 5 illustrates the four possible session configurations offered in this example. The Answerer is only supporting session configurations 3 and 4 (due to SCL$_B$), and shall just acknowledge session configuration 3 due to its preference by the Offerer.
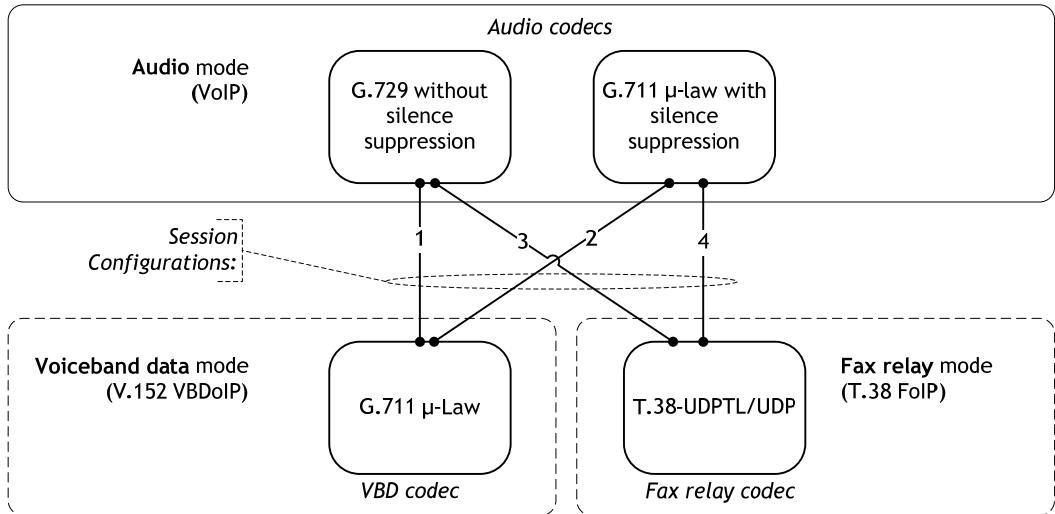

Figure 5 – Potential session configurations offered in this example

Table 12 – Example SDP encoding – O/A cycle in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments

<pre>1) OFFER:
; SESSION CONFIGURATIONs
a=sescap:1 1   ; VoIP = G.729, VBDoIP =
V.152 (G.711)
a=sescap:2 2   ; VoIP = G.711, VBDoIP =
V.152 (G.711)
a=sescap:3 3,5 ; VoIP = G.729, FoIP = T.38
UDPTL/UDP
a=sescap:4 4,5 ; VoIP = G.711, FoIP = T.38
UDPTL/UDP
 	
; LATENT CONFIGURATION for T.38
a=tcap:2 udptl   ; T.38 FoUDPTL/UDP
transport variant
a=mcap:5 t38   ; T.38 FoIP codec (subtype
= 't38')
a=acap:11 T38FaxVersion:0
a=acap:12
T38FaxRateManagement:transferredTCF
a=acap:13 T38FaxUdpEC:t38UDPRedundancy
a=acap:14 (… additional T.38 attributes may
be included)
a=lcfg:5 mt=image t=2 m=5 a=11,12,13,14,…
 	
; ACTUAL CONFIGURATION (due to backward
compatibility)
… ommitted …
;</pre>
	Offered (4) potential configurations (as session configurations due to 'voice' and 'facsimile'):
	• Preference 1: Audio (ITU-T G.729), VBD (ITU-T V.152 PCMU)
	• Preference 2: Audio (PCMU with silence suppression), VBD (ITU-T V.152 PCMU)
	• Preference 3: Audio (ITU-T G.729) and fax relay (ITU-T T.38 FoUDPTL/UDP)
	• Preference 4: Audio (PCMU) and fax relay (ITU-T T.38 FoUDPTL/UDP)

Table 12 – Example SDP encoding – O/A cycle in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments

`; POTENTIAL CONFIGURATIONs`
`a=tcap:1 RTP/AVP  ; transport for VoIP and`
`VBDoIP`
`a=acap:1 ptime:10`
`a=mcap:1 G729/8000    ; audio codec 1`
`a=mcap:2,4 PCMU/8000  ; audio codec 2 and`
`VBD codec`
`a=mcap:3 CN/8000      ; comfort noise for`
`audio c. 2`
`a=mscap:4 gpmd vbd=yes  ; for V.152 PCMU`
`a=pcfg:1 t=1 a=-ms:1 m=1,4 pt=1:18,4:96`
`a=pcfg:2 t=1 a=-ms:1 m=2,3,4`
`pt=1:0,3:13,4:96`
`a=pcfg:3 t=1 a=-ms:1 m=1 pt=1:18`
`a=pcfg:4 t=1 a=-ms:1 m=2,3 pt=1:0,3:13`	

`2) ANSWER:`
`…`
`a=sescap:3`
	The Answerer selects the 3rd session configuration.

7.1.3.3 Example 6 – ITU-T V.152 to non-ITU-T V.152 (OCL with ITU-T V.152 and ITU-T V.150.1, ACL without ITU-T V.152)

Gateway A supports voice codecs ITU-T G.729, ITU-T V.152 and ITU-T V.150.1. Gateway B also supports ITU-T V.150.1 and voice codec ITU-T G.729 but does not support ITU-T V.152.

NOTE – This example 6 shows the minimum number of lines needed to construct an SDP-compliant session descriptor that includes all attributes that are mandatory for the representation of SPRT modem relay and ITU-T V.152.

In this example, ports 49230 and 49232 (see Offer in Table 13) are used for the RTP/AVP and SPRT media streams, respectively. Within the RTP/AVP media stream, the static payload types of 0 (PCMU) and 8 (PCMA) are marked for VBD treatment via the 'gpmd' attribute, and thus cannot be used for voice.

Also note, in accordance with SIP, the SDP in Table 13 implies simultaneous support of audio 'rtp/avp' and audio 'udpsprt'. To indicate that only one media type can be supported at a time, the 'group' attribute with the FID semantics, together with the 'mid' attribute, should be used as specified in [IETF RFC 3388] (see example 5, clause 7.1.3.2).

Gateway B does not support ITU-T V.150.1 (implementations which also must support VBD). Gateway B shall respond with an SDP as follows (see Answer in Table 13).

Table 13 – Example SDP encoding – O/A cycle in Legacy SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments
1) OFFER: SDP from Gateway A: v=0 o=Gateway A 25678 753849 IN IP4 128.96.41.1 s= c=IN IP4 128.96.41.1 t=0 0 m=audio 49230 RTP/AVP 0 8 18 97 98 a=gpmd:0 vbd=yes a=gpmd:8 vbd=yes a=rtpmap:97 telephone-event/8000 a=fmtp:97 0-15,32,33,34,35,66,70 a=rtpmap:98 v150fw/8000 m=audio 49232 udpsprt 100 a=sprtmap:100 v150mr/8000 a=fmtp:100 mr=0; mg=1;DSCselect=3;mrmods=1,2;jmdelay=no;versn =1.1	
2) ANSWER: v=0 o=GatewayB 25678 753849 IN IP4 128.96.41.1 s= c=IN IP4 128.96.41.1 t=0 0 m=audio 49230 RTP/AVP 0 8 18 97 98 a=gpmd:0 vbd=yes a=gpmd:8 vbd=yes a=rtpmap:97 telephone-event/8000 a=fmtp:97 0-15,32,33,34,35,66,70 a=rtpmap:98 v150fw/8000 m=audio 49232 udpsprt 100 a=sprtmap:100 v150mr/8000 a=fmtp:100 mr=0; mg=1;DSCselect=3;mrmods=1,2;jmdelay=no;versn =1.1	

Because both gateways have negotiated support of SSE, they shall use SSEs to indicate transition between voice and VBD.

Figure 6 illustrates the three possible session configurations, offered in this example. The Answerer is not supporting session configuration 3 (due to lack of support of [ITU-T V.150.1]), and is acknowledging session configuration 1 (due to support of PCMU as VBD codec), see Table 14.
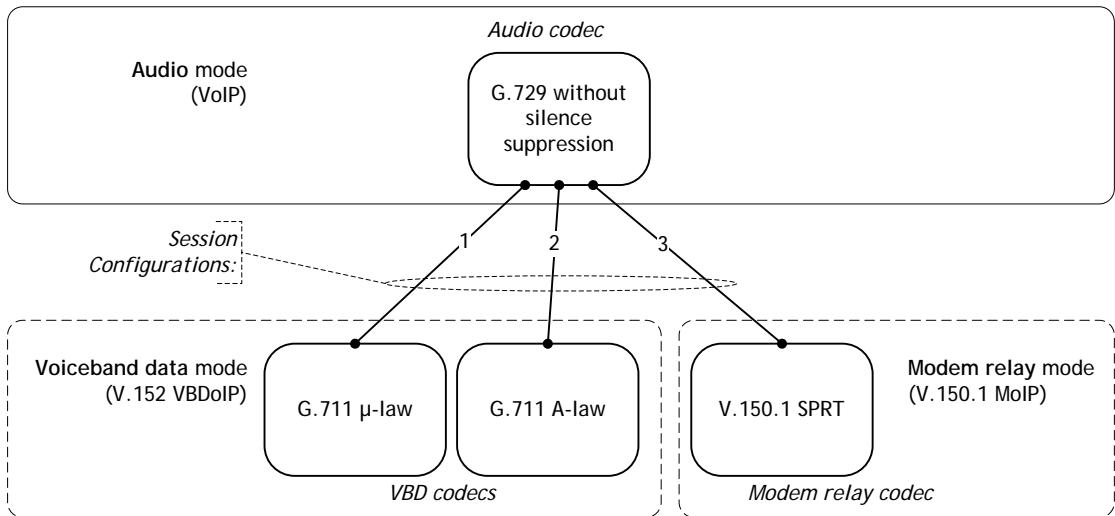
Figure 6 – Potential session configurations offered in this example

Table 14 – Example SDP encoding – O/A cycle in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments

<pre>1) OFFER:
; SESSION CONFIGURATIONs
a=sescap:1 1   ; VoIP = G.729, VBDoIP =
V.152 (PCMU)
a=sescap:2 2   ; VoIP = G.729, VBDoIP =
V.152 (PCMA)
a=sescap:3 3,4 ; VoIP = G.729, MoIP =
V.150.1 SPRT
 	
; LATENT CONFIGURATION for V.150.1
a=tcap:2 udpsprt ; V.150.1 SPRT/UDP
transport variant
a=mcap:6 v150mr/8000 ; V.150.1 MoIP codec
a=mfcap:6 mr=0; mg=1; CDSCselect=3;
mrmods=1,2;
          jmdelay=no; versn=1.1
a=lcfg:4 mt=audio t=2 m=6 pt=6:100
 	
; ACTUAL CONFIGURATION (due to backward
compatibility)
… ommitted …
;
; POTENTIAL CONFIGURATIONs
a=tcap:1 RTP/AVP ; transport for VoIP,
                  ; VBDoIP, SSEoIP & NTEoIP
a=acap:1 ptime:10
a=mcap:1 G729/8000 ; audio codec
a=mcap:2 PCMU/8000 ; VBD codec 1
a=mcap:3 PCMA/8000 ; VBD codec 2
a=mcap:4 telephone-event/8000 ; NTE codec
a=mcap:5 v150fw/8000   ; SSE protocol</pre>
	Offered (4) potential configurations (as session configurations due to 'voice' and 'facsimile'):
	
	• Preference 1: Audio (ITU-T G.729), VBD (ITU-T V.152 PCMU)
	• Preference 2: Audio (ITU-T G.729), VBD (ITU-T V.152 PCMA)
	• Preference 3: Audio (ITU-T G.729) and modem relay (ITU-T V.150.1 using SPRT (simple packet relay transport over UDP))
	
	Some MoIP configuration details:
	– ITU-T V.8-compatible modem relay (mr = 0);
	– media gateway of type "No Trans-compression" (mg = 1);
	– Call discrimination mode selection "Mixed" (CDSCselect = 3);
	– supported modulation types "1" (ITU-T V.34 duplex), "2" (ITU-T V.34 Half-duplex) (mrmods = 1, 2);
	– JM delay procedure "No" (jmdelay=no);
	– state signalling event (SSE) protocol supports "Yes" (see v150fw declaration for SSE-over-RTP packets with PT = 98);
	– SSE event types: "1" (initial audio), "3" (modem relay), "4" (text relay), "5" (fax relay);

Table 14 – Example SDP encoding – O/A cycle in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments

```a=mscap:2,3 gpmd vbd=yes ; for V.152 PCMU & PCMA```
```a=mfcap:4 0-15,32-35,66,70 ; value range of NTE```
```a=mfcap:5 1,3-5   ; SSE event types```
```a=mfcap:5 expack=yes ; SSE reliability```
 	
```a=pcfg:1 t=1 a=-ms:1 m=1,2,4```
```   pt=1:18,2:96,4:97```
```a=pcfg:2 t=1 a=-ms:1 m=1,3,4```
```   pt=1:18,3:99,4:97```
```a=pcfg:3 t=1 a=-ms:1 m=1,4,5```
```   pt=1:18,4:97,5:98```
	– SSE assured transport (using 'Explicit Acknowledgement' method).

```2) ANSWER:```
…
```a=sescap:1```
	The Answerer selects the 1st session configuration.

7.1.4   Optional ITU-T V.152 capabilities

This clause describes the SDP representation of information that may be optionally declared at session establishment time. The absence of their declaration shall be construed by an ITU-T V.152 implementation as an indication that the remote ITU-T V.152 implementation does not support them.

7.1.4.1   Declaration of redundancy and forward error correction

The declaration, in SDP, of [IETF RFC 2198] redundancy and [IETF RFC 5109] FEC, shall conform to the rules in the applicable IETF source documents. When supporting text telephones, on networks where the error character service requirements as specified in clause A.3 of [b-ITU-T F.700] is exceeded due to packet loss, then this Recommendation strongly encourages the appropriate use of [IETF RFC 2198] redundancy and [IETF RFC 5109] FEC for the IP network to which it is attached. However, on some networks, the application of redundancy/FEC may contribute to the character error rate and should not be used.

7.1.4.1.1 Example 7 – Packet redundancy for ITU-T V.152 with PCMU as VBD codec

Although the [IETF RFC 2198] rules are not repeated here, declaring [IETF RFC 2198] support with a redundancy level of three for a VBD codec is illustrated with an example (see Tables 15 and 16).

Figure 7 illustrates the offered media configuration in example 7.

Table 15 – Example SDP encoding – "Example 7" offer in Legacy SDP Offer/Answer syntax

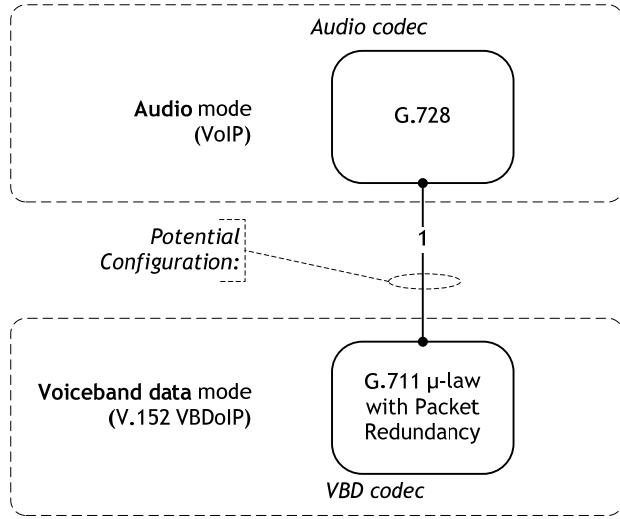| SDP encoding (shortened SDP description)	Comments

```1) OFFER:```
```m=audio 3456 RTP/AVP 0 15 102```
```a=gpmd:0 vbd=yes```
```a=rtpmap:102 red/8000```
```a=fmtp:102 0/0/0/0```	

Figure 7 – Potential media configuration offered in this example

Table 16 – Example SDP encoding – "Example 7" offer in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments

```; ACTUAL CONFIGURATION (due to backward compatibility)```
 ```m=audio 3456 RTP/AVP 0 15 102```
 ```a=gpmd:0 vbd=yes```
 ```a=rtpmap:102 red/8000```
 ```a=fmtp:102 0/0/0/0```
 ```;```
 ```; POTENTIAL CONFIGURATIONs```
 ```a=tcap:1 RTP/AVP ; transport for VoIP & VBDoIP```
 ```a=mcap:1 G728/8000   ; audio codec```
 ```a=mcap:2 PCMU/8000   ; VBD codec```
 ```a=mcap:3 RED/8000   ; RTP RFC 2198 redundancy```
 ```a=mfcap:3 %2%/%2%/%2%/%2% ; RFC 2198 redundancy format```
 ```a=mscap:2 gpmd vbd=yes ; for V.152 PCMU```
 ```a=pcfg:1 t=1 a=-ms m=2,1,3 pt=2:0,1:15,3:102```
	Offered potential configuration:
	 • Preference 1: Audio (ITU-T G.728), VBD (ITU-T V.152 PCMU with packet redundancy)

7.1.4.1.2 Example 8 – Forward error correction for ITU-T V.152 with PCMU as VBD codec

Examples of the declaration of FEC support are found in [IETF RFC 5109]. This includes the use of a separate FEC stream and the combination of the FEC stream with the primary stream via IETF RFC 2198 encapsulation. In the case when FEC is a separate stream, [IETF RFC 5109] uses an 'fmtp' line to associate this stream with an IP address and port. When FEC packets are sent to the same IP address and port (albeit a different SSRC) as the media packets they qualify, there is no need for the 'fmtp' line to associate the 'parityfec' payload type with an IP address and port. Thus, in the following SDP segment (see Table 17), the last line is superfluous and may be omitted. Likewise, the absence of an 'fmtp' line associating an IP address and port with a FEC payload type will be construed to mean that the FEC packets are to be sent to the same IP address and port as the media packets they qualify.

Table 17 – Example SDP encoding – "Example 8" offer in Legacy SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments
1) OFFER: c=IN IP4 224.2.17.12 t=0 0 m=audio 49170 RTP/AVP 0 15 78 a=gpmd:0 vbd=yes a=rtpmap:78 parityfec/8000 a=fmtp:78 49170 IN IP4 224.2.17.12	

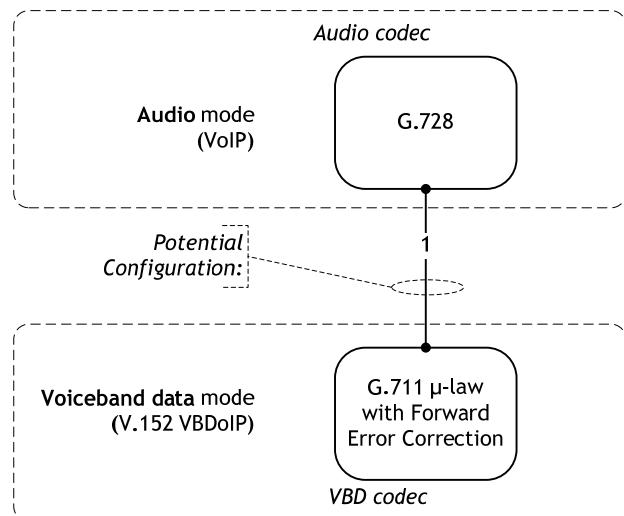Figure 8 illustrates the offered media configuration in example 8.

Figure 8 – Potential media configuration offered in this example

Table 18 – Example SDP encoding – "Example 8" offer in Revised SDP Offer/Answer syntax

SDP encoding (shortened SDP description)	Comments
; ACTUAL CONFIGURATION (due to backward compatibility) c=IN IP4 224.2.17.12 t=0 0 m=audio 49170 RTP/AVP 0 15 78 a=gpmd:0 vbd=yes a=rtpmap:78 parityfec/8000 a=fmtp:78 49170 IN IP4 224.2.17.12 ; ; POTENTIAL CONFIGURATIONs a=tcap:1 RTP/AVP ; transport for VoIP & VBDoIP a=mcap:1 G728/8000 ; audio codec a=mcap:2 PCMU/8000 ; VBD codec a=mcap:3 parityfec/8000 ; RTP RFC 5109 FEC a=mscap:2 gpmd vbd=yes ; for V.152 PCMU a=pcfg:1 t=1 a=-ms m=2,1,3 pt=2:0,1:15,3:78	Offered potential configuration: • Preference 1: Audio (ITU-T G.728), VBD (ITU-T V.152 PCMU forward error correction)

7.1.4.2 Optional vendor-specific parameters

The 'vndpar' (vendor parameters) attribute may be used to declare vendor codes for coordinating enhanced operation over and above those indicated in this Recommendation. It shall be possible to safely ignore vendor-specific parameters and still maintain interoperability with equipment conforming to this Recommendation. Hence, proprietary enhancements cannot be a substitute for the basic features required for compliance with this Recommendation.

The format of the 'vndpar' attribute line is as follows:

```
a=vndpar:<vendorIDformat> <vendorID> <vendorSpecificDataTag>
            [<vendorSpecificData>]
```

The `<vendorIDformat>`, a decimal, indicates the format of the following `<vendorID>` field. The following values are defined:

Integer representation	Vendor ID format
1	[ITU-T T.35]
2	IANA Private enterprise number

The `<vendorID>` may be represented in hex or in decimal format. If represented in hex, it has a '0x' prefix. Generally, if the vendor ID format is ITU-T T.35, the hexadecimal format is preferred. If it is the IANA private enterprise number (http://www.iana.org/assignments/enterprise-numbers), the decimal format is preferred.

When the vendor ID format is ITU-T T.35, the vendor ID consists of a country code followed by a vendor code. The country code consists of four octets and the vendor ID consists of two octets. If the representation of the vendor ID is hexadecimal, leading zeros in the country code may be omitted, while leading zeros in the vendor code may not be omitted.

When the `<vendorID>` is the vendor's private enterprise number, leading zeros may be omitted.

The `<vendorSpecificDataTag>` is a decimal integer between 0-255. If used, values in the range 1-255 are uniquely mapped, via the 'vndpar' attribute, to the combination of the vendor specified in the `<vendorID>` and the proprietary capabilities indicated by `<vendorSpecificData>`. This mapping, which exists for the duration of a session, does not persist across sessions. Further, each side may choose this integer independently of the other end. Due to the compactness of this index, a gateway or endpoint may use it in a number of places. A value of 0 is a null value. When present, it is equivalent to omitting the `<vendorSpecificDataTag>`. A null value of the `<vendorSpecificDataTag>` is not associated with any vendor ID.

It shall be possible for an endpoint or gateway to declare multiple (1-255) 'vndpar' attribute lines in an SDP session description. Each of these lines may indicate a different vendor. In addition, multiple 'vndpar' lines may indicate the same vendor. When multiple 'vndpar' lines are declared in an SDP session descriptor, each value of `<vendorSpecificDataTag>` must either be unique within all 'vndpar' lines in the session descriptor or null (0). If non-null, the `<vendorSpecificDataTag>` may serve as a dynamically assigned feature identifier for the vendor.

Inclusion of the parameter `<vendorSpecificData>` is optional. When included, this is a vendor-defined octet string consisting of one or more octets. Since it consists of an integer number of octets, it is represented by an even number of hex characters. No '0x' prefix is needed. No size limitation is specified since SDP parsers can ignore another vendor's string without checking its length. A vendor is permitted to add additional structure to the `<vendorSpecificData>` field such that features are identified by their position in this field. A vendor may also elect to add explicit

feature identification within the `<vendorSpecificData>` field. When present, these supplement the `<vendorSpecificDataTag>`.

Note that the vendor is not precluded from using the `<vendorSpecificData>` field to communicate parameters that are not related to ITU-T V.152.

7.2 Use of VBD in ITU-T H.323 systems

ITU-T H.323 systems support ITU-T V.152 through the use of the `VBDCapability` capability defined in [ITU-T H.245]. This capability, which is a type of `AudioCapability`, is used during capability exchange and in the open logical channel (OLC) signalling to indicate support for VBD channels and to signal the opening of those channels. Since VBD media flows are generally switched within a single RTP session with normal voice audio and other audio-related media (e.g., [IETF RFC 4733]), OLC proposals and terminal capability set messages generally utilize the multiple payload stream (MPS) constructs in [ITU-T H.245].

7.2.1 Fast connect procedures

ITU-T H.323 systems may offer one or more logical channel proposals in the SETUP message transmitted to the called party. The ITU-T H.323 device orders those logical channel proposals in order of preference. This allows an endpoint to indicate its preferred mode of operation and allows the called device to understand what is preferred, but to also accept any alternative modes offered by the calling device.

If the calling device prefers to use VBD for the transport of all voice-band data, including facsimile, text, and modem signalling, the first proposal in the OLC would consist of a non-VBD audio codec and a VBD codec. If the calling endpoint also supports ITU-T T.38 relay over RTP, for example, it might offer as a second proposal a non-VBD audio codec, a VBD codec, and ITU-T T.38. In this way, the called device knows that the calling device prefers to use VBD for all voice-band data, but is also willing to do ITU-T T.38 relay over RTP in the case that the called device has this preference. As with normal Fast Connect procedures, the called device is free to accept any of the alternative proposals or refuse all of them and utilize normal ITU-T H.245 signalling to open logical channels.

Generally, ITU-T H.323 devices would also signal as additional OLC proposals, ones that contain different audio codecs in combination with VBD codecs. Further, devices would also signal alternatives that offered only a non-VBD codec as the choice for media, in the case that the called device does not support this Recommendation. The choice of OLC proposals, the order of proposals, and the selection is a matter of implementation.

ITU-T H.323 devices compliant with this Recommendation may also utilize Extended Fast Connect, which allows devices to renegotiate media streams and to make counter-proposals to those OLCs offered by the remote endpoint. Refer to [b-ITU-T H.460.6] for the procedures related to Extended Fast Connect.

By no means does this Recommendation override the rules defined in [ITU-T H.323] related to Fast Connect procedures or the rules defined in [b-ITU-T H.460.6] for Extended Fast Connect.

7.2.2 Exchanging VBD capabilities

Devices specify support for VBD by including capabilities of the type `VBDCapability` in the ITU-T H.245 `TerminalCapabilitySet` message. As with other types of media, these capabilities may be grouped into capability descriptors to signal sets of simultaneous capabilities. Further, since VBD is generally one type of audio that is switched within the same RTP session as other media, `VBDCapability` capabilities are generally only defined as part of a multiple payload stream. However, since a device may wish to open a VBD stream that does nothing more than transmit media as VBD, capabilities may be defined and used outside of an MPS.

7.2.3   ITU-T H.245 logical channel signalling procedures

Once ITU-T H.323 devices have exchanged capabilities, they may open logical channels by transmitting open logical channel messages. The procedures for logical channel signalling are defined in [ITU-T H.323] and no additional procedures are defined in this Recommendation.

Since ITU-T H.323 devices operate asynchronously, it is possible that one device may transmit an OLC message offering one set of capabilities, while the peer device transmits an OLC with an incompatible set of capabilities. For example, one device may propose an OLC that suggests the use of {G.729, VBD/G.711, T.38} while the peer device sends an OLC which suggests the use of {G.723.1, VBD/G.726}. Of course, both messages should be independently legal based on the exchanged capabilities. While [ITU-T H.323] permits devices to use different audio codecs in each direction, it may not be preferred. In this case, the fact that one side proposes to use T.38 over RTP and the other side does not is a problem. In all such cases, [ITU-T H.323] specifies that the master shall resolve such conflicts by rejecting the OLC with a reason of `masterSlaveConflict` or other appropriate reason. Devices should not fail as a result, but should converge to a common mode.

Devices should utilize the request mode message in [ITU-T H.245] in order to suggest a compatible mode of operation. Either the master or the slave device may transmit a request mode message. Note, however, that the request may be rejected. Ultimately, the slave device may have no choice, except to use the preferred mode of the master. Even so, the master should honour requests from the slave device when possible.

As an example to illustrate the opening of a media channel in accordance with this Recommendation, consider an OLC that has an ITU-T G.729 voice stream, an ITU-T G.711 A-law VBD stream that is protected with redundancy encoding, an [IETF RFC 4733] stream, and an ITU-T T.38 over RTP stream. The `OpenLogicalChannel` would essentially have a composition similar to what is shown here:

```
{
    forwardLogicalChannelNumber 1,
    forwardLogicalChannelParameters {
        dataType : multiplePayloadStream {
            element {
                dataType : audioData : g729 2
            },
            element {
                dataType : redundancyEncoding {
                    primary {
                        dataType : audioData : vbd : g711Alaw64k 160
                    },
                    secondary {
                        {
                                            dataType : audioData : vbd
                          : g711Alaw64k 160,
                        }
                    }
                },
                payloadType 101      -- The PT for the RFC 2198 packet
            },
            element {
                dataType : audioData : audioTelephonyEvent {
                        audioTelephoneEvent : "0-15,32,33"
                },
                payloadType 102
            },
            element {
                dataType : audioData : genericDataCapability {
                        capabilityIdentifier : standard {
                                itu-t(0) recommendation(0) t(20) 38
```

```
                                        h245-audio-capability(0)
                            },
                    nonCollapsing {
                            {
                                    parameterIdentifier : standard : 0,
                                    parameterValue : booleanArray : 0
                            },
                            {
                                    parameterIdentifier : standard : 1,
                                    parameterValue : unsignedMin : 0
                            },
                            {
                                    parameterIdentifier : standard : 2,
                                    parameterValue : genericParameter
                                    {
                            {
                                    parameterIdentifier : standard : 1,
                                    parameterValue : logical
                            }
                                    }
                            },
                            {
                                    parameterIdentifier : standard : 3,
                                    parameterValue : unsigned32Max : 200
                            },
                            {
                                    parameterIdentifier : standard : 4,
                                    parameterValue : unsigned32Max : 72
                            },
                            }
                    }
            },
                    payloadType 103
            }
        }
    },
    multiplexParameters : h2250LogicalChannelParameters {
        sessionID 1
        }
    }
  }
```

7.3 Mid-call negotiation using session description protocol

There might be the need for mid-call ITU-T V.152 negotiations, if ITU-T V.152 was not negotiated during the call establishment phase and if VBD stimuli were detected in a "mid-call phase".

7.3.1 Possible use cases

Example use cases:

1) A call is started with negotiated audio VoIP only (neither ITU-T V.152 VBDoIP nor any packet relay mechanism like ITU-T T.38 FoIP was indicated and negotiated);

2) There might be mid-call VBD stimuli detected.

   NOTE – The VBD stimuli detection logic in VoIP gateways may be always enabled, independent of the call-individual negotiation;

3) Any mid-call VBD stimuli detection shall trigger a capability renegotiation (e.g., via SIP re-INVITE);

4) Preferred media configuration may be ITU-T T.38 FoIP (thus, SDP Offer in SIP re-INVITE only contains ITU-T T.38 and no other alternative);

5) However, if ITU-T T.38 fails, then ITU-T V.152 VBDoIP may be negotiated in a subsequent SDP Offer/Answer cycle.

7.3.2 SIP/SDP mid-call negotiations

The possibility of the above-mentioned mid-call scenarios should be superseded by the application of Revised SDP Offer/Answer during the call establishment phase. See the advantages listed in clause 7.3.2.2.

7.3.2.1 SIP/SDP mid-call negotiations using Legacy SDP Offer/Answer

Step 4 in clause 7.3.1 implies at least one negotiation cycle.

Step 5 in clause 7.3.1 implies at least one further negotiation cycle.

7.3.2.2 SIP/SDP mid-call negotiations using Revised SDP Offer/Answer

The problem of possible mid-call negotiations may be simply addressed by Revised SDP Offer/Answer, because:

– supported media and transport capabilities (here: ITU-T T.38, ITU-T V.152) may be already checked out during the call establishment phase;

– preferences and alternatives may be indicated and agreed already during the call establishment phase (here: ITU-T T.38 would fail, ITU-T V.152 would be selected for VBD);

– concept of latent configuration allows to identify a possible, later (i.e., "mid-call") media configuration (here: start with audio, but later VBD stimuli detection will lead to an operational mode change).

Consequently, a possible mid-call change shall be already clarified during the call establishment phase when both gateway entities support Revised SDP Offer/Answer. This obsoletes any SIP signalling (see steps 3 to 5 in clause 7.3.1) due to the autonomous state transitioning capability of ITU-T V.152 gateways.

8 The use of IETF RFC 4733 modem/facsimile/text telephone events

IETF RFC 4733 telephone-events ANS (32), /ANS (33), ANSam (34) and /ANSam (35) can optionally be used as an alternative method for the transport of these signals in audio or VBD packets. If these events are declared by a media gateway, the remote media gateway may use [IETF RFC 4733] to transmit these signals. If both media gateways indicate the support of IETF RFC 4733 telephone-events ANS (32), /ANS (33), ANSam (34) and /ANSam (35), the gateway generating the events shall use [IETF RFC 4733] to transmit these signals. In either case, when [IETF RFC 4733] is used for transporting these signals, the gateways shall suppress the transport of these signals in audio or VBD packets. The amount of in-band signal leakage into the IP network using audio encoding for ANS, ANSam, /ANS, and /ANSam signals shall be less than 50 ms.

IETF RFC 4733 telephone-events ANS (32), /ANS (33), ANSam (34) and /ANSam (35) could be used by the media gateways for the tone disabling ([ITU-T G.168]) of the echo canceller function if provided and enabled in the media gateway and shall be used for the generation of the appropriate signal on the TDM interface. If either end does not indicate this support, then the media gateways shall detect the 2100 Hz tone with phase reversals signal for echo canceller tone disabling on their incoming VBD packet stream.

9 VBD stimuli

This clause lists the stimuli that should be detected, per type of application, by a VBD gateway to initiate a transition to the VBD mode of operation, as described in clause 10.

The list of stimuli below is not exhaustive and there may be other tones that can be used to initiate a transition to VBD for the listed applications:

- *For facsimile applications*:
    - CED as per [ITU-T T.30];
    - ANSam as per [ITU-T V.8];
    - Preamble as per clause 5.3.1 of [ITU-T T.30];
    - CNG as per [ITU-T T.30].
- *For modem applications*:
    - ANS as per [ITU-T V.8];
    - ANSam as per [ITU-T V.8];
    - 2225 Hz answer tone as per Appendix VI of [ITU-T V.150.1];
    - Unscrambled binary ones signal as per [b-ITU-T V.22];
    - CI signals that precede ANSam, as per [ITU-T V.8];
    - Initiating Segment 1 dual tones (1375 Hz and 2002 Hz) as per [b-ITU-T V.8*bis*].
- *For text telephony applications*:
    - ANS as per [ITU-T V.8];
    - ANSam as per [ITU-T V.8];
    - Text telephone signals as defined by clause 5.1.1 of [ITU-T V.18];
    - DTMF signals only if [IETF RFC 4733] telephone-events are not supported;
    - CI signals that precede ANSam, as per [ITU-T V.8];
    - Calling tone (CT) signals that precede ANS, as per [b-ITU-T V.25];
    - Initiating Segment 1 dual tones (1375 Hz and 2002 Hz) as per [b-ITU-T V.8*bis*].

In addition to the above list, if any other unrecognized tonal non-voice signal is detected, this may be used to transition into VBD mode.

VBD gateways should keep signal leakage to a minimum to prevent erroneous behaviour of end terminals.

10 Procedures for transitioning between audio mode and VBD mode

This clause describes the transitioning mechanism for an implementation that only supports VBD as per this Recommendation and voice, but does not support any relay mechanisms such as ITU-T T.38 or ITU-T V.150.1, nor VBD as per [ITU-T V.150.1].

The mechanism described in this clause shall be the default mandatory mechanism used by ITU-T V.152-compliant gateways if no other mechanisms have been successfully negotiated between the gateways, otherwise the mutually negotiated mechanism (such as those described in clause 11) shall be used in preference to this method.

10.1 Procedures for state transitioning

10.1.1 Audio-to-VBD transition

The transition from audio mode to VBD mode is performed when the VBD detectors classify an input signal as VBD.

Detection of the stimuli described in clause 9 shall be carried out at least in the direction from the GSTN to the IP network; however, detection in the direction from the IP network to the GSTN network is not precluded. The bidirectional VBD stimuli detection capability of a VBD gateway improves the probability of correct and timely gateway local state transition, which offers a more robust VBD service and state transition.

On detection of any of the stimuli described in clause 9, if the corresponding IETF RFC 4733 telephone-event has not been mutually negotiated, a ITU-T V.152 implementation must transmit them as in-band VBD packets.

If the ITU-T V.8 CI and the ITU-T V.8*bis* signals are transmitted in-band rather than as IETF RFC 4733 events, a shift to VBD must not lose any part of the signals. The choice of using in-band indication or IETF RFC 4733 for indicating these signals is dependent upon capability declaration, whether a VBD channel is available, and the preference of the transmitter.

When in the VBD media state, a media gateway may use [IETF RFC 4733] in lieu of voice-band transmission to communicate to the remote gateway any of the voice-band data stimuli indicated in clause 9.

The use of [IETF RFC 4733] in this case is contingent on the capabilities declared by the remote gateway.

When in the VBD media state, the unscrambled binary ones signal is communicated in-band. There is no [IETF RFC 4733] support for this signal.

When in the VBD media state, the text telephone signals are communicated in-band. The media gateway shall not lose any characters at the onset of in-band VBD transmission.

The gateway shall suppress a voice-band data stimulus from the bearer path if it intends to convey the stimulus as an IETF RFC 4733 telephone event. This shall be done immediately on detection of the stimulus. A media gateway knows, prior to the detection of a voice-band data stimulus, whether it will transmit the stimulus in-band or via an IETF RFC 4733 telephone event. This knowledge is based on the remote gateway's capabilities (whether it can receive an IETF RFC 4733 encoding of that stimulus) and the local gateway's own choice (since it may use in-band transmission regardless of the remote gateway's capability declaration).

Once VBD has been mutually negotiated by the two gateways, using the procedures described in clause 7, a gateway that complies with this Recommendation shall be able to receive and appropriately decode, from the IP network, RTP packets with any of the supported negotiated payload types for a particular call. Hence, an ITU-T V.152 implementation shall transition from voice to VBD on receipt of an RTP packet that has the negotiated VBD payload type.

Additionally, gateways may optimize operation by doing one of the following:

• Loading both the audio and VBD codecs to streamline quick, on-the-fly transitions between talkspurts and textspurts.

• Staying in the VBD mode across talkspurts and textspurts.

Thus, on detection, in the direction from the GSTN to IP network of the appropriate VBD signals, a VBD gateway shall transition to VBD and transmit as soon as possible RTP packets with the corresponding negotiated VBD payload type. Reception of an RTP packet that has the pre-negotiated VBD payload type at the remote end shall cause a VBD gateway to transition to VBD mode, but only if, prior to receiving the VBD RTP packet, it received RTP packets that correspond to the state it was previously in (e.g., voice packets). The reason for the latter rule is explained below:

Consider two VBD gateways, termed A and B, connected via an IP network and each having a GSTN network on their other sides. There will be a period during a call when both VBD gateways are in VBD mode. Gateway A transitions into audio mode due to detection of voice signals in the direction from the GSTN to the IP network, which will cause it to transition into voice and transmit voice RTP packets. While the first transmitted voice RTP packet is traversing the IP network, the remote end (gateway B) is still transmitting VBD RTP packets, because it has not detected anything on its GSTN side, nor has it yet received the voice RTP packets. To avoid gateway A from transitioning erroneously back into VBD mode, it must not transition back to VBD until it has first received (the pre-negotiated) voice RTP packets, which it should expect to receive due to it (i.e., gateway A) transitioning into voice.

NOTE – An implementation shall be able to handle out-of-order RTP packets (e.g., a voice packet followed by a VBD packet that was actually sent before the voice packet).

10.1.2 VBD-to-audio transition

Transition from VBD to voice may be carried out by detection:

* In the direction from the GSTN to IP network of any of the following stimuli:
  - end of modem or facsimile signals;
  - voice signals;
  - detection in both directions, GSTN to IP and IP to GSTN, of silence. With the following caveats:
    * For text telephones, the appropriate detection of silence must be considered because text telephone conversations may have long periods of silence.
    * For the case of facsimile calls, the silence period should be greater than the T2 timer defined in [ITU-T T.30].
  - MGC signalling or other out-of-band signalling methods.
* In the direction from IP to GSTN network due to receipt of RTP packets that have non-VBD payload types, only after the first VBD RTP packet has been received. This will avoid the situation of an incorrect transition into audio mode when it has transitioned to VBD mode on detection of VBD signals on its TDM side and is still receiving voice RTP packets (because the remote end has not yet transitioned, based on reception of the VBD RTP packets).

10.2 State machine – Overview

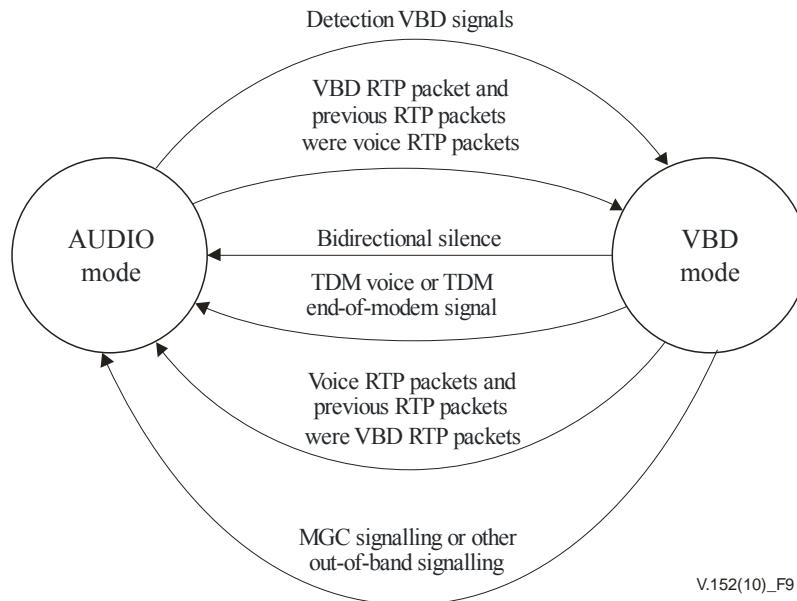The above described transition criteria are also summarized in Figure 9.

Figure 9 – Voice-VBD transitioning state diagram

10.3 Enforced state transition by call/session control

The call/session control may enforce state transitioning based on detected VBD stimuli in the bearer-path, which are reported to the call control, or the media capability negotiation during call establishment or modification, or other policies (like, e.g., preferred media formats of a service provider). This capability is indicated by the arrow "*MGC signalling or ...*" in Figure 9.

Procedures for such transitioning behaviour would be dependent on the specific applied signalling protocols (e.g., ITU-T H.323, SIP, ITU-T H.248, etc.), which are therefore beyond the scope of this Recommendation.

11 Optional procedures for indicating to a remote end transition into VBD using state signalling events

This clause describes the procedures that an ITU-T V.152 implementation must use when using the state signalling event protocol as defined in Annexes C, E and F of [ITU-T V.150.1].

Note that the use of SSE is optional for ITU-T V.152 implementation and subject to negotiation with the remote gateway. When one or both of the two gateways does not support SSE operation, then transitions to and from the VBD mode shall be governed by the procedures defined in clause 10.

11.1 Declaration of SSEs

The SSE capability shall be signalled as defined in clause F.6 of [ITU-T V.150.1]. The minimum set of state signalling events that shall be supported for VBD operation are events 0 through 3 which are basic to the SSE protocol. The null SSE event (0) shall never be sent and should be ignored if received.

11.2 Transition to the VBD mode for ITU-T V.150.1 gateways

When both gateways support [ITU-T V.150.1], then transitions to and from the VBD mode is governed by ITU-T V.150.1 procedures. These transitions are synchronized via the SSE protocol.

11.3   Transition to the VBD mode for non- ITU-T V.150 cases

When one or both of the two gateways does not support ITU-T V.150.1 operation, then transitions to and from the VBD mode shall be governed by the procedures in this clause. An attempt shall be made to make these procedures isomorphic to ITU-T V.150.1 procedures so that media gateways do not incur the burden of supporting and testing multiple VBD switching mechanisms.

An ITU-T V.152-compliant media gateway that has successfully negotiated support of SSE media gateway shall respond to a voice-band data stimulus by immediately transitioning the connection to the VBD media state and issuing an SSE indicating this state (see clause C.5.2 of [ITU-T V.150.1]). As with all other media state transitions, this is contingent on resource availability. On making this transition locally, the stimulus-detecting media gateway may start sending VBD packets immediately.

On receiving an SSE indicating the VBD media state (SSE:VBD), a media gateway shall immediately transition the connection to the VBD media state if it has the resources to do so. Before making this transition, it may ignore any in-band VBD packets it receives (see clause 20.4 of [ITU-T V.150.1]).

The transition to a VBD media state in response to detecting a voice-band data stimulus (such as an answer tone variant) is illustrated in Figure 10. In this example, the on-ramp (call-originating) gateway G1 and the off-ramp (call-terminating) gateway G2 support VBD operation.

On detecting a voice-band data stimulus, gateway G2 determines whether it has the resources to transition the session to the VBD media state. If it does, then it immediately makes the transition and sends SSE:VBD (event code 2) to gateway G1. While in the VBD media state, it uses an RTP payload type marked for VBD treatment.

On receiving SSE:VBD, gateway G1 determines whether it has the resources to transition the session to the VBD media state. If it does, then it immediately makes the transition and sends an SSE:VBD back to gateway G2, confirming that its media state has changed to VBD. If it does not, then it sends an SSE:audio (event code 1) on receiving a SSE:VBD from G2.
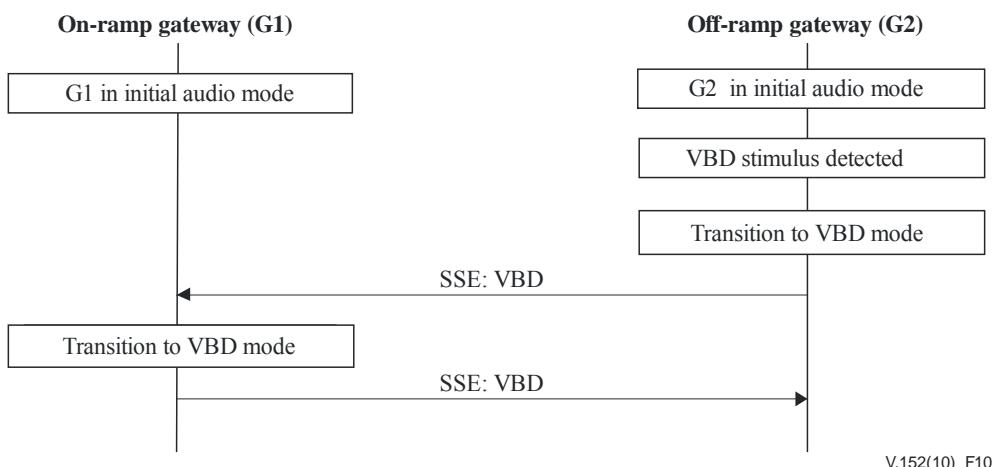

Figure 10 – Initiation of VBD operation in response to VBD stimulus detection

When sending SSE:VBD, gateways G1 and G2 may use suitable reason information codes (RICs) defined in [ITU-T V.150.1]. An example is the RIC indicating an answer tone. A null code, which conveys no information, may also be used. The SSE:VBD from G1 may indicate a p' state transition as the reason information code. Since p' is defined as a gateway's view of the other gateway's protocol state, this indicates that this SSE from G1 is a response to a received SSE.

Distinctions based on RIC codes (ITU-T V.8 versus non- ITU-T V.8, text vs. non-text) may be used to optimize play-out buffer settings and FEC levels for different applications of VBD. Additionally, when the RIC indicates text, gateways may optimize operation by doing one of the following:

– Loading both the audio and VBD codecs to streamline quick, on-the-fly transitions between talkspurts and textspurts.

– Staying in the VBD mode across talkspurts and textspurts.

11.4 Transition from the VBD media mode

On detecting a termination of data transmission, media gateways shall locally transition the connection to the audio mode and issue an SSE:audio (Initial audio SSE, event code 1) to the remote gateway. If it receives an SSE:audio, it shall change the media state to initial audio and respond back with an SSE:audio.

The criteria for determining the termination of data transmission are application-specific and are not defined here. Examples of such criteria are the detection of voice or of pre-determined intervals of silence. A transition to the modem, fax or text media states is not a termination of data transmission.

By declaring support of the SSE protocol, gateways implicitly declare support for events 1-3 which are basic to the protocol. In order to be used, support for other SSEs such as SSE:FR (Facsimile relay SSE, event code 4) and SSE:TR (Text relay SSE, event code 5) shall be explicitly declared.

Transitions from the VBD media state to the MR (modem relay), FR (facsimile relay) and TR (text relay) states are permitted. These media state transitions are contingent on:

1) Declaration of the applicable capabilities at call establishment.

2) Availability of resources at the time of the state transition.

In the modem relay case, these are synchronized using SSE:MR (Modem relay SSE, event code 3) per [ITU-T V.150.1].

SSE:TR (Text relay SSE, event code 5) is recommended to synchronize media shifts from VBD to the text relay mode, and vice versa. For example, when ITU-T V.21 signals are followed by Annex A of [ITU-T V.18] signals in end-to-end ITU-T V.18 automoding, there may be a shift to VBD based on the ANS preceding [ITU-T V.21], and possibly another shift to TR if the gateway does not support VBD for Annex A of [ITU-T V.18].

For autonomous media shifts from VBD to facsimile relay, two cases arise from [ITU-T T.38]:

1) For gateways that comply with [ITU-T V.150.1] and Annex F of [ITU-T T.38], SSE:FR is used. Both single-port and multi-port operation is supported.

2) For all other gateways, port activity monitoring is used. Note that single-port operation for audio RTP and ITU-T T.38 udptl packets is not supported; however, single-port operation may be used if audio RTP and the optional ITU-T T.38 RTP procedure are being used.

Since facsimile switchovers are tolerant in terms of signal timing, external signalling can be used in lieu of the autonomous VBD-to-FR media shifts described in the last paragraph. Examples of external signalling are SIP re-invites, ITU-T H.245 RequestMode/CLC/OLC and ITU-T H.248.1 context modification. End-to-end timing issues that often jeopardize the use of external signalling with modem traffic do not exist for facsimile traffic.

For a session that is in the VBD media state, a gateway may reject SSE:MR, SSE:FR or SSE:TR with an SSE:VBD or SSE:audio. If used, SSE:audio causes a transitioning of the session to an audio mode.

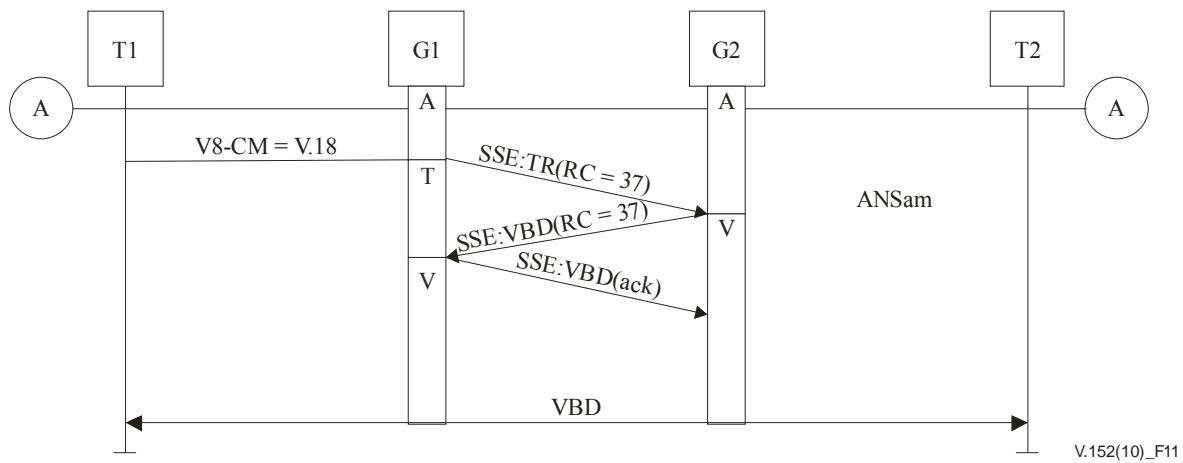The SSE reason identifier codes for VBD mode are defined in Table 12 of [ITU-T V.150.1].

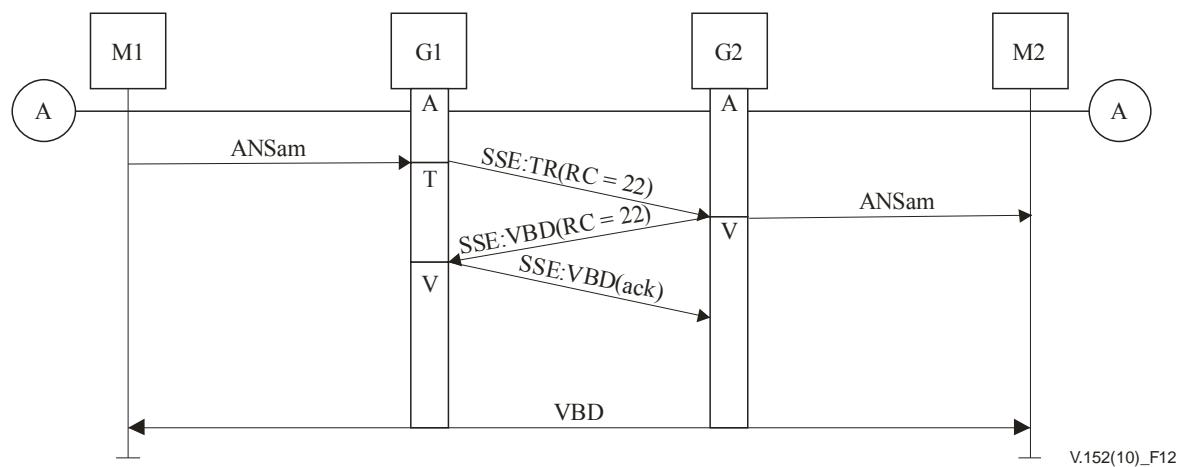**Figure 11 – ITU-T V.18 text telephone using VBD**

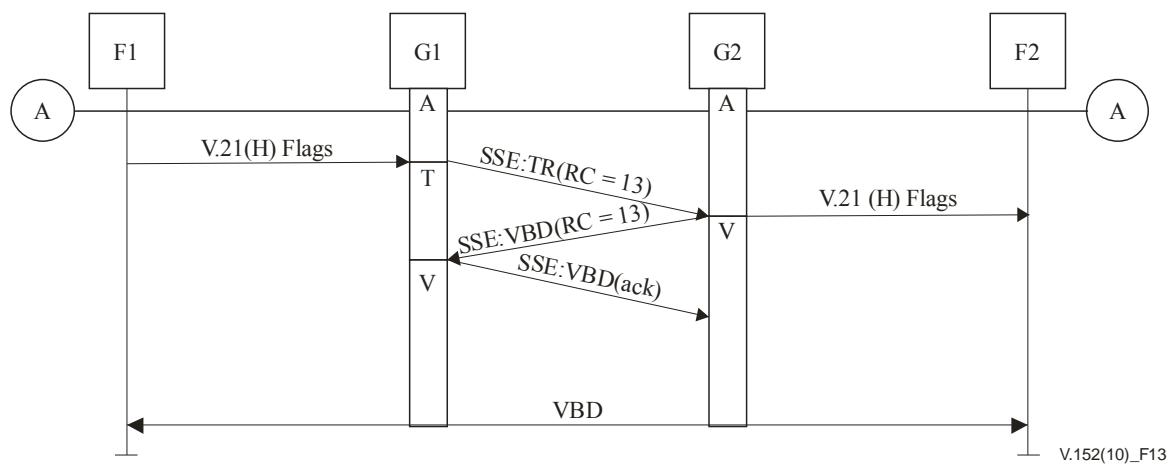**Figure 12 – ITU-T V.34 modem using VBD**

Figure 13 – Group 3 facsimile (without CNG/CED tones) using VBD

11.5 Security – Optional

When the VBD mode is used to transport data payloads, it lends itself easily to secure, encrypted operation based on SRTP (Secure RTP). Support of security features is not required by ITU-T V.152 compliant implementations and is negotiated at call establishment.

Based on declarations at the time of session establishment, it is possible to encrypt some RTP payload types (e.g., voice, VBD and IETF RFC 4733 events), while passing other RTP payload types (e.g., SSEs) unencrypted. Such selective encryption will allow fast response times to SSEs, without compromising the security of media sourced by the end-user. When one end proposes encrypted operation for a set of payload formats, and the other end does not support encryption, the preferred outcome is a rejection of the proposal and termination of the connection attempt. At this point, either the on-ramp or off-ramp media gateway or media gateway controller can counter-propose a non-encrypted connection through the call signalling protocol(s) in use.

Annex A

Vendor-defined messages

(This annex forms on integral part of this Recommendation.)

Vendor-specific messages may be supported within ITU-T V.152, subject to negotiation with the remote end. In general, an ITU-T V.152 implementation may support up to 255 vendor identifiers (vendor-ID) for a given call. Each vendor-ID may be unique or specific and tied to either a single or multiple sets of attributes. A unique vendor-Tag may also be assigned to each set of attributes associated with a vendor-ID to allow simpler use within ITU-T V.152.

Usually the vendor-ID is provided during the external signalling used during the call set-up (i.e., ITU-T H.245, ITU-T H.248 or SDP, etc.). The format used in signalling schemes may be compliant to either [ITU-T T.35] or the IANA private enterprise number. The choice is up to the vendor.

When the vendor-ID format is ITU-T T.35, the vendor-ID consists of a country code followed by a vendor code. The country code consists of four octets and the vendor-ID consists of two octets. If the representation of the vendor-ID is hexadecimal, leading zeros in the country code may be omitted, while leading zeros in the vendor code may not be omitted.

When the vendor-ID is the vendor's IANA private enterprise number, leading zeros may be omitted.

The vendor-Tag is a decimal integer with a value between 0 and 255. If used, values in the range of 1 to 255 are uniquely mapped to the combination of vendor-ID and vendor-specific information. The choice of this integer made by a gateway is independent of the choice made by its peer gateway. Due to the compactness of this index, a gateway or endpoint may use it in a number of places to simplify the messaging. A value of zero for the vendor-Tag is a null value. When present, it is equivalent to omitting the vendor-Tag. A null value of the vendor-Tag is not associated with any vendor-ID. If non-null, the vendor-Tag may serve as a dynamically assigned vendor-specific identifier.

The vendor-specific information is an octet string consisting of one or more octets as defined by the vendor. Since it consists of an integer number of octets, it is represented by an even number of hex characters. No "0x" prefix is needed. Limitation on size is context specific. Details where size is limited will be indicated appropriately.

Annex B

Use of data signal detection and silence insertion in voiceband data

(This annex forms on integral part of this Recommendation.)

B.1   Introduction

Data signal detector (DSD) and silence insertion in VBD mode is a technique that reduces the transmission bandwidth over IP networks by inserting silence indicator descriptors during the silence periods of a voiceband data modem connection. This is especially efficient for half-duplex data transmissions. This annex provides guidance on DSD and silence insertion for gateways compliant to this Recommendation.

B.2   Guideline for the use of DSD

When in the VBD state, a data signal detector monitors and analyses the voiceband analogue input signals. The data signal detector categorizes the analysed signal to be either a correct voiceband modem signal or silence. The means of determining this signal categorization is implementation specific and this Recommendation does not describe any particular method in detail. However, an example is: once in the VBD state, a gateway may monitor the signal energy level (this is the same signal as used for driving ITU-T V.24 Circuit 109) and use that for determining the presence of silence. In this instance, echo levels must be accounted for in the analysis process.

Most modem procedures include a short silence between different training and data mode phases. These vary from 70 ± 5 ms for [ITU-T T.30] to 75 ± 5 ms for high speed duplex data modems. In these instances, it is important to maintain the fidelity of these short silence periods. A DSD detector shall use a silence validation time greater than 80 ms.

Upon the qualified detection of silence, the gateway transmits a silence insertion description (SID) frame to inform the peer gateway of the beginning of the silence period. The SID frame format shall be that specified for use with the selected VBD codec. For example if ITU-T G.711 is the codec being used, then the SID format shall comply with Appendix II of [ITU-T G.711].

In the SID frame, the level field is set to zero and the N1~Nm fields are set to zero. Only an initial SID frame needs to be transmitted per silence period.

The receiving gateway upon receiving a SID frame or packet shall play out the SID frame zero level data as silence. The level of the transmitted analogue silence shall be less than –55 dBm. The receiving gateway shall continue to play out silence until it receives a valid VBD data packet.

To ensure that the signal can be detected at the receiver side, the maximum response time for a DSD detector to detect the silence to data transition should less than 1.5 ms. The transmitting gateway shall detect the presence of a voiceband signal and transmit a valid VBD packet without delay.

B.3   Negotiation of the DSD capability with SDP

Gateways shall mutually negotiate the capability of DSD. Clause 7.1 explains the detailed procedure for negotiation using SDP. The DSD negotiation shall comply with the procedures described in clause 7.1, and use two a-line attributes to indicate support of DSD capability.

If VBD MGW supports DSD capability, the SDP attribute should indicate:

```
m=audio 3456 RTP/AVP 0 18 98 99
a=rtpmap:98 PCMA/8000
a=gpmd:98 vbd=yes
a=rtpmap:99 CN/8000
a=gpmd:99 dsd=yes
```

"a=gpmd:99 dsd=yes" indicates that the device supports the DSD capability; "a=rtpmap:99 CN/8000" shows that the device sends SID frame with payload type 99. If mutually negotiated, both gateways can use the DSD procedures described in this annex.

If a VBD MGW does not support DSD capability, the SDP attribute should indicate:

```
m=audio 3456 RTP/AVP 0 18 98 99
a=rtpmap:98 PCMA/8000
a=gpmd:98 vbd=yes
a=rtpmap:99 CN/8000
a=gpmd:99 dsd=no
```

"a=gpmd:99 dsd=no" indicates that the device does not support or does not want to use DSD.

To maintain compatibility with gateways using a previous version of this Recommendation that would not support DSD capability, the absence of the gpmd attribute for DSD is interpreted to be the same as "dsd=no" capability.

B.4 Negotiation of the DSD capability with [ITU-T H.245]

Clause 7.2 explains the use of VBD in ITU-T H.323 systems. ITU-T H.323 systems support ITU-T V.152 through the use of the `VBDCapability` capability defined in [ITU-T H.245]. This capability, which is a type of `AudioCapability`, is used during capability exchange and in the open logical channel (OLC) signalling to indicate support for VBD channels and to signal the opening of those channels. An extension to the `VBDCapability` is needed to indicate the DSD capability.

```
VBDCapability                       ::=SEQUENCE
{
    type                            AudioCapability,    -- shall not be "vbd"
    dsd                             BOOLEAN
    ...
}
```

If VBD MGW supports DSD capability, the dsd attribute should indicate "TRUE". If a VBD MGW does not support the DSD capability, the dsd attribute should indicate "FALSE". The absence of the dsd attribute for DSD is interpreted as no support for the DSD capability.

Annex C

Use of ITU-T V.21 preamble for echo canceller control
in an ITU-T V.152 gateway

(This annex forms an integral part of this Recommendation.)

For the scenario where an ITU-T V.34 facsimile terminal is being called by a standard G3 facsimile terminal, the procedures defined in [ITU-T T.30] stipulate that the connection proceeds as a standard G3 facsimile. Also, [ITU-T T.30] recommends that the ITU-T V.34 facsimile terminal should transmit an ITU-T V.8 answer tone with phase reversals, thereby tone disabling any gateway echo cancellers in the connection. In this situation, the gateway echo cancellers will be left in the disabled state when proceeding into ITU-T T.30 standard G3 modes. This has a direct impact upon the performance of standard G3 facsimile terminals as described in [ITU-T G.168] and [b-ITU-T G.161], which require that echo cancellers be in their initial cancelling state in order to provide the best conditions for a facsimile transmission.

[ITU-T T.30] (2006) was updated to correct this situation and defines how the ITU-T T.30 procedures will allow for the re-enabling of echo cancellers to their initial cancelling state. However, this particular scenario remains an issue to be considered as there are many millions of facsimile terminals that do not support the 2006 version of [ITU-T T.30].

This annex describes a method that may be used to rectify this condition in gateways compliant to this Recommendation. This method consists in using the detected presence of the V.21High Channel HDLC encoded FLAG preamble as defined in [ITU-T T.30] to initiate a transition from the tone disabled state back to the initial cancelling state. Once the echo canceller has returned to its initial state, the standard G3 facsimile procedure will operate as intended. (See Figure C.1.)
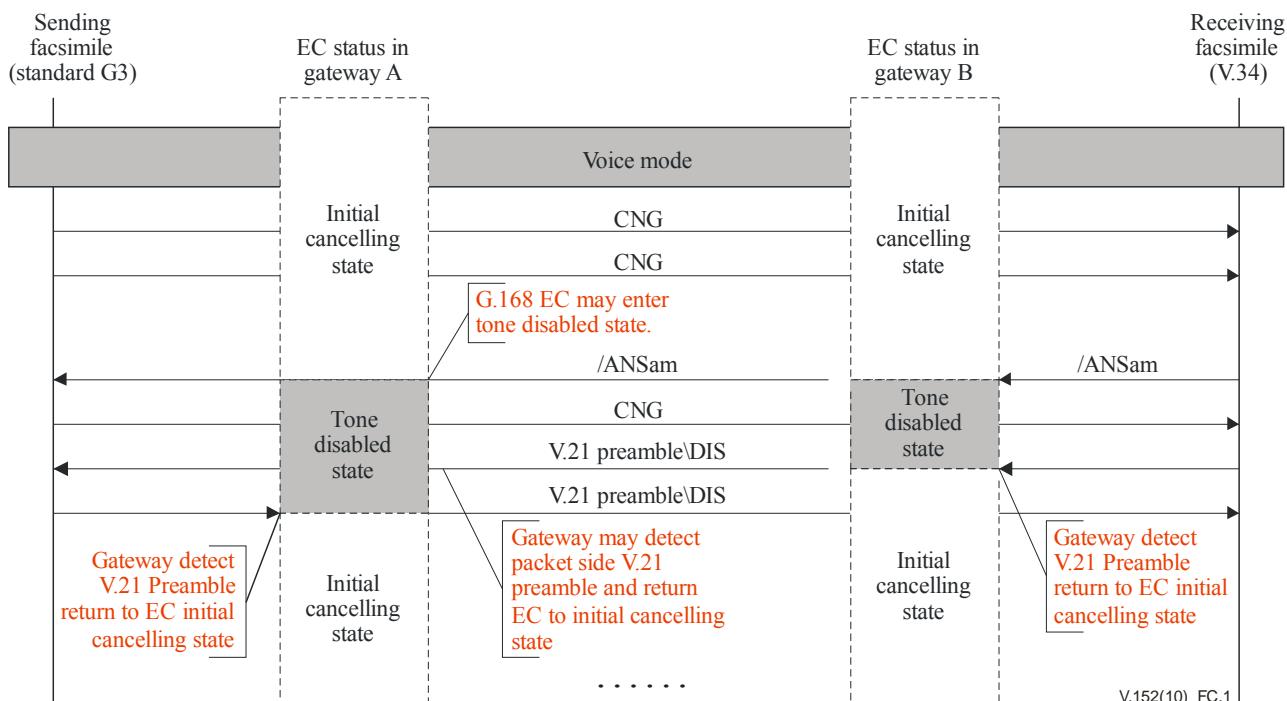

Figure C.1 – Fax procedure and echo canceller status after enabling by ITU-T V.21 preamble

Gateway to echo canceller signalling

This procedure is only valid for media gateways compliant to this Recommendation. Note that in this application, echo cancellers may be embedded within the media gateway or be external to it. The signalling of the gateway to the echo canceller can be via proprietary means if it is an embedded type in the gateway, or by some standardized means if connected to an external echo canceller. The means definition of external echo canceller control is for further study.

Bibliography

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[b-ITU-T V.8bis]   Recommendation ITU-T V.8*bis* (2000), *Procedures for the identification and selection of common modes of operation between data circuit-terminating equipments (DCEs) and between data terminal equipments (DTEs) over the public switched telephone network and on leased point-to-point telephone-type circuits*.

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[b-ITU-T V.25]   Recommendation ITU-T V.25 (1996), *Automatic answering equipment and general procedures for automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls*.

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