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SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

Digital terminal equipments – Other terminal equipment

Procedures for control of de-jitter buffers used in PSTN-IP gateways carrying voiceband data

Recommendation ITU-T G.799.4

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Recommendation ITU-T G.799.4

Procedures for control of de-jitter buffers used in PSTN-IP gateways carrying voiceband data

Summary

Recommendation ITU-T G.799.4 describes procedures for the control of de-jitter buffers used in PSTN-IP gateways carrying voiceband data.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T G.799.4	2014-10-14	16	11.1002/1000/12234

Keywords

De-jitter buffer, PSTN-IP gateways.

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Procedures for control of de-jitter buffers used in PSTN-IP gateways carrying voiceband data

1 Scope

De-jitter buffers have a major effect on voice and data transmission quality in telecommunication networks. They affect the three service categories of public switched telephone network/integrated service digital network (PSTN/ISDN) voice, voiceband data (due to PSTN modem calls) and ISDN circuit mode data. Since the requirements for the settings of de-jitter buffers differ for different services, this Recommendation describes the activation and mode switching procedures of de-jitter buffers, including the requirement for in-band tone activating and other control mechanisms.

It is assumed that the clock accuracy of all elements involved is sufficiently high for application of this Recommendation.

This Recommendation covers de-jitter buffer usage in circuit-to-IP media gateways, such as residential, access or trunking gateways in the context of the next generation network/internet protocol multimedia subsystem (NGN/IMS).

The notion of circuit relates to a PSTN analogue line or an ISDN 1×64 bearer channel.

1.1 Applicability statements

This Recommendation is

a) applicable:

  a1) to circuit-to-IP media gateways and communication services with a gateway interworking function operating at the level of a synchronous byte-stream, such as:

  – service "voice-over-IP" (VoIP) without or with silence suppression;

  – service "voiceband data-over-IP" (VBDoIP); and

  – service "circuit-mode data-over-IP" (CMDoIP);

  a2) to IP-to-IP media gateways for dedicated interworking services between two IP domains, such as:

  – service "IPDV reduction between two IP domains" with different grade of service (GoS) as e.g., described in [b-ITU-T G.799.3];

  – service "RTP PDV reduction between two RTP domains", which may be the subject of an "RTP transport translator" topology (see [IETF RFC 5117] and [ITU-T H.248.88]);

  but IP-to-IP media gateways are basically out of the scope of this Recommendation due to their focus on circuit-to-IP gateway types,

and is

b) not applicable, because de-jitter buffers are not required:

  b1) to circuit-to-IP media gateways and communication services with a gateway interworking function operating at the level of individual packets as atomic units (i.e., an asynchronous packet-stream), such as:

  – service "facsimile-over-IP" (FoIP) according to [ITU-T T.38];

  – service "text-over-IP" (ToIP) according to [ITU-T V.151]; and

  – service "data-over-IP" (MoIP) according to [b-ITU-T V.150.1];

b2) to IP-to-IP media gateways in general.

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.161.1] Recommendation ITU-T G.161.1 (2014), *Do-no-harm testing*.

[ITU-T G.168] Recommendation ITU-T G.168 (2012), *Digital network echo cancellers*.

[ITU-T H.248.88] Recommendation ITU-T H.248.88 (2014), *Gateway control protocol: RTP topology dependent RTCP handling by ITU-T H.248 media gateways with IP terminations*.

[ITU-T T.38] Recommendation ITU-T T.38 (2010), *Procedures for real-time Group 3 facsimile communication over IP networks*.

[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.8bis] Recommendation ITU-T V.8bis (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*.

[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.22] Recommendation ITU-T V.22 (1988), *1200 bits per second duplex modem standardized for use in the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits*.

[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*.

[ITU-T V.32] Recommendation ITU-T V.32 (1993), *A family of 2-wire, duplex modems operating at data signalling rates of up to 9600 bit/s for use on the general switched telephone network and on leased telephone-type circuits*.

[ITU-T V.32bis] Recommendation ITU-T V.32bis (1991), *A duplex modem operating at data signalling rates of up to 14 400 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits*.

[ITU-T V.151] Recommendation ITU-T V.151 (2006), *Procedures for the end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay*.

[ITU-T V.152] Recommendation ITU-T V.152 (2010), *Procedures for supporting voice-band data over IP Networks*.

[ITU-T P.501] Recommendation ITU-T P.501 (2012), *Test signals for use in telephonometry*.

[IETF RFC 5117] IETF RFC 5117 (2008), *RTP Topologies*.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

3.1.1 de-jitter buffer [b-ITU-T G.1020]: A buffer designed to remove the delay variation (i.e., jitter) in packet arrival times. Data is put into the de-jitter buffer at a variable rate (i.e., whenever they are received from the network), and taken out at a constant rate.

3.1.2 voice-band data mode [ITU-T V.152]: 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 [ITU-T V.152].

3.1.3 G3 facsimile equipment (G3FE) [ITU-T T.38]: G3FE refers to any entity which presents a communications interface conforming to [b-ITU-T T.30], [b-ITU-T T.4], and, optionally, [b-ITU-T T.6]. A G3FE may be a traditional G3 facsimile machine, an application with a [b-ITU T T.30] protocol engine, or any of the other possibilities mentioned in the network model for IP facsimile.

3.1.4 PSTN modem call [b-ETSI TR 183 072]: Voiceband data call originating/terminating in a PSTN domain.

NOTE – The term voiceband data (VBD) is an umbrella term for all kind of teleservices which using a "data-oriented transport" in the frequency band of the narrowband voice spectrum (which is a 3.1-kHz-band). The data-oriented transport is realized by modem protocols (definition as in clause 3.13 of [ITU-T V.152]), as defined e.g., within the ITU-T V.x-series of Recommendations. Teleservices may be categorized into three major applications areas: facsimile, text-based communication and general data services.

3.1.5 ITU-T T.38/G3 [ITU-T T.38]: ITU-T T.38/G3 refers to an ITU T T.38 endpoint that supports G3FE, but excludes the ITU-T T.30/ITU-T V.34 procedures.

3.1.6 ITU-T T.38/ITU-T V.34G3 [ITU-T T.38]: ITU-T T.38/ITU-T V.34G3 refers to an ITU-T T.38 endpoint that supports G3FE and includes the ITU-T T.30/ITU-T V.34 half-duplex procedures.

3.1.7 VBD gateway [ITU-T V.152]: A media gateway that is compliant with [ITU-T V.152].

3.1.8 XoIP emulation service (for PSTN modem calls) [b-ETSI TR 183 072]: Emulation service in IP networks, based on appropriated gateway technologies for interworking voiceband data information between the PSTN and IP networks.

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

– Facsimile/modem: Gateway technologies for PSTN-to-IP interworking see e.g., [ITU-T V.152] for pass-through mode and [ITU-T T.38] as packet-relay mode;

– Text/modem: Gateway technologies for PSTN-to-IP interworking see e.g., [ITU-T V.152] for pass-through mode and [ITU-T V.151] as packet-relay mode; and

– Data/modem: Gateway technologies for PSTN-to-IP interworking see e.g., [ITU-T V.152] for pass-through mode and [b-ITU-T V.150.1] as packet-relay mode.

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 de-jitter buffer delay (JBD): The de-jitter buffer delay is also called de-jitter delay, holding time or play-out delay. It corresponds to the average time packets stay in the buffer, which is less than the de-jitter buffer size (typically corresponding to 50% of the de-jitter buffer size). The time of departure of each packet is determined by reading out the timestamp information provided by RTP.

3.2.2 de-jitter buffer size (JBS): The maximum amount of time packets can stay in the buffer (typically specified in milliseconds).

3.2.3 pseudo-VBDoIP emulation service: An XoIP emulation service, trying to support voiceband data in audio mode (see clause 3.2.1 of [ITU-T V.152]), also known as non-V.152 VBDoIP service.

3.2.4 VBDoIP emulation service: A XoIP emulation service compliant with [ITU-T V.152].

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

ANS	Answer tone 2 100 Hz
ANSam	Answer tone 2 100 Hz amplitude-modulated by a sinewave at 15 Hz
/ANSam	Answer tone 2 100 Hz amplitude-modulated by a sinewave at 15 Hz with phase reversals at an interval of 450 ± 25 ms
ANM	Answer Message
ATA	Analogue Terminal Adapter
CED	Called tone which is physically identical to ITU-T V.25 ANS
CI	signals that precede ANSam, as per [ITU-T V.8]
CT	Calling Tone
DCE	Data Communication Equipment
DJB	De-Jitter Buffer
DTMF	Dual-Tone Multi-Frequency signalling
EC	Echo Canceller
ECM	Error Correction Mode
FoIP	Facsimile over IP (ITU-T T.38)
G3FE	Group 3 Facsimile Equipment
GSTN	General Switched Telephone Network
GW	Gateway
IAD	Integrated Access Device
IMS	Internet protocol Multimedia Subsystem
IP	Internet Protocol
IPDV	IP Delay Variation
ISDN	Integrated Service Digital Network
ISUP	ISDN User Part

JBD	de-Jitter Buffer Delay
JBS	de-Jitter Buffer Size
MGW	Media Gateway
MoIP	data Modem over IP
NGN	Next Generation Network
NNI	Network-Network Interface
PLC	Packet Loss Concealment
PSTN	Public Switched Telephone Network
RTP	Real Time Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
ToIP	Text over IP [ITU-T V.151]
UDI	Unrestricted Digital Information
UNI	User Network Interface
VBD	Voiceband Data
VBDoIP	Voiceband Data over IP
VGW	Voice Gateway (SIP)
XoIP	X over IP (with 'X' as placeholder for "IP application protocol 'X'")

5 Conventions

None.

6 Overview – XoIP emulation services versus applicable de-jitter buffer modes of operation

Clause 7 of [b-ETSI TR 183 072] summarizes all possible media configurations used in circuit-to-IP gateways (such as ITU-T H.248 media gateways, SIP voice gateway) for the emulation of PSTN/ISDN bearer services. The two major modes of a de-jitter buffer, adaptive and fixed, are not useful for all XoIP emulation services. Figure 1 provides a correspondent summary.

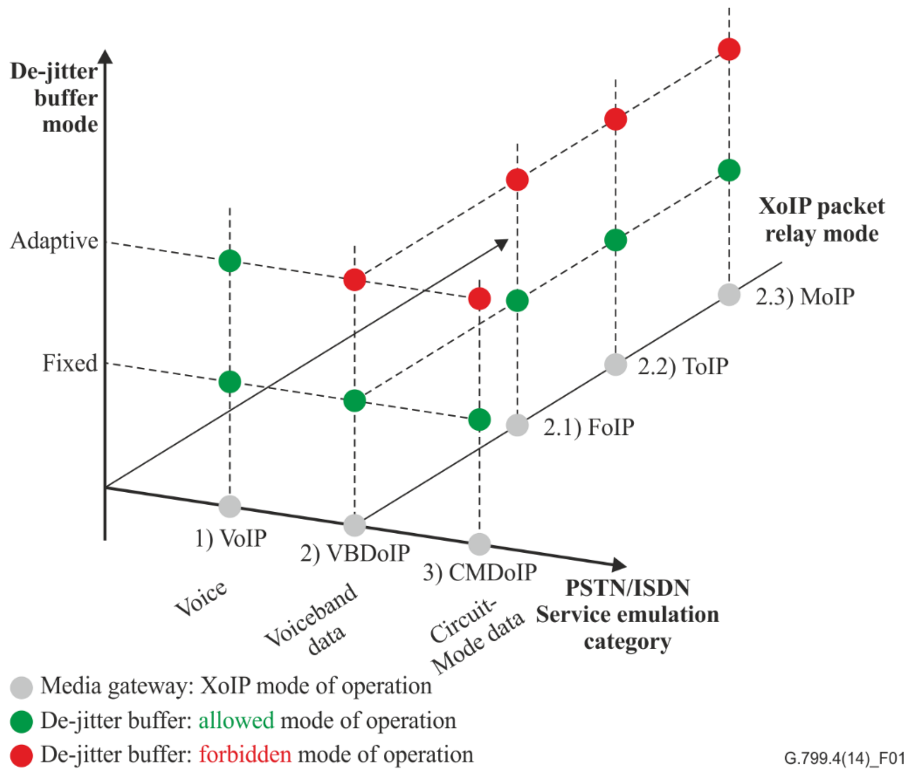


Figure 1 – Circuit-to-IP media gateways – XoIP emulation services versus applicable de-jitter buffer modes of operation

The selection of a de-jitter buffer mode follows the two basic targets of "minimization of end-to-end transfer delay" versus "minimization of information loss". The cost of adaptive de-jitter buffer mode has to be paid for by increased information loss in comparison to the fixed mode.

Minimization of information loss is a basic requirement for all data services. Hence, the adaptive mode should be avoided for XoIP emulation services related to voiceband data and circuit-mode data.

The adaptive de-jitter buffer mode is therefore only relevant for the VoIP service category.

NOTE – This fundamental relation of the applicability of DJB mode versus XoIP emulation services could and should be basic behaviour of circuit-to-IP media gateways, as e.g., underlined by clauses 6.2.2.1.2 and 6.2.2.2.2 of [b-ETSI TR 183 072].

7 Characteristics of de-jitter buffers

7.1 General

This Recommendation describes the activation procedures of a de-jitter buffer, including the requirement for in-band tone activating and other control mechanisms. The de-jitter buffers are assumed to be adaptive de-jitter buffers and fixed de-jitter buffers. Fix de-jitter buffers shall be provided for fax and voice band data and 64 kbit/s bit sequence (UDI).

7.1.1 De-jitter buffers

A de-jitter buffer is designed to remove the effects of jitter from the decoded packet voice stream, buffering each arriving packet for a short interval before playing it out synchronously. A fixed de-jitter buffer maintains a constant size whereas an adaptive de-jitter buffer has the capability of adjusting its size dynamically in order to optimize the delay/discard tradeoff. The disadvantage of adaptive de-jitter buffer is that a part of the jitter budget is transferred to the user. While the human perception of audio delay variation is low, modem and fax applications are extremely sensitive to delay variation in the audio path. For this reason, adaptive de-jitter buffer are not applicable for fax and modem transmission. Fixed de-jitter buffers try to maintain a constant end-to-end delay. See Figure 2.

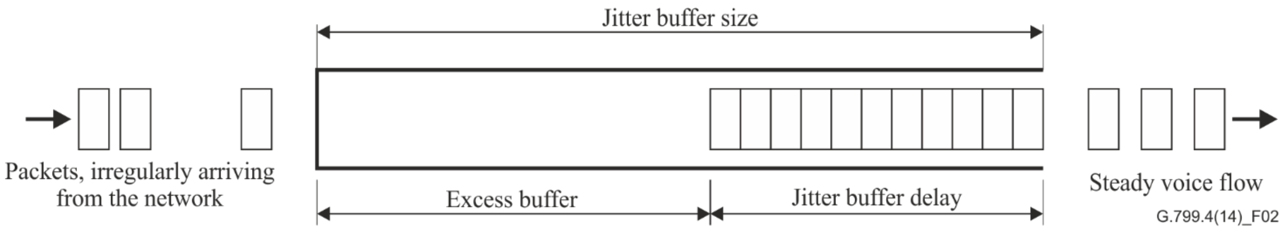


Figure 2 – De-jitter buffer size and delay

7.2 Purpose, operation and environment

The fundamental requirements (for the considered TDM/analogue-to-RTP media gateway types) are:

1. support of ITU-T V.152 VBDoIP due to:
   - inherent "end-to-end constant latency" support (clause 6 of [ITU-T V.152])
   - echo canceller (EC) ITU-T G.168 relation (clause 6 of [ITU-T V.152]).

NOTE – Pseudo-VBD is not sufficient or not adequate for ITU-T V.152 VBD. There is an explicit distinction and support of a 2-state model (audio, VBD) required.

For proper operation for VBD-services, de-jitter buffers have the following fundamental requirements:

1) fast and correct switching between adaptive and fixed de-jitter buffer mode;
2) proper operation during facsimile and data transmissions.

For proper operation of speech services of good quality, false detection of tones (e.g., from answering machines, call centres or speech) has to be minimized.

7.3 External enabling of fixed de-jitter buffers

The fixed mode of the de-jitter buffers shall be applied for the XoIP media configurations, VBDoIP and CMDoIP. CMDoIP emulation service is always enabled via the network control plane. VBDoIP emulation service is normally enabled by network user plane stimuli.

NOTE – Gateway control signalling for de-jitter buffer mode of operation:

– implicit, such as [ITU-T V.152]
– explicit, such as [ITU-T H.248.31] or clause E.11.1.1 of [ITU-T H.248.1].

8 Activation of de-jitter buffer for VBDoIP

During audio mode, the initiating station (in the V.x modem) sends the calling tone (for fax called CNG), 1 100 Hz, a series of interrupted bursts of binary 1 signal or the 1 300 Hz signal (ITU-T V.25) and while this takes place the user of the receiving station may be continuing to speak or send audio. The station on the left (Figures 3, 4 and 5) is the initiating station. The speech or audio signal from the station on the right has placed the de-jitter buffer in the adaptive mode. The following tones shall drive both de-jitter buffers DJB2 and DJB1 into the fixed mode:

– CED as per [ITU-T T.30]
– ANS as per [ITU-T V.8]
– ANSam as per [ITU-T V.8]
– /ANSam as per [ITU-T V.8]
– Preamble as per clause 5.3.1 of [ITU-T T.30]

– 2 225 Hz answer tone as per Appendix VI of [b-ITU-T V.150.1]
– Unscrambled binary 1 is detected for 155 ± 50 ms as per [ITU-T V.22]
– Segment 1 dual tones (1 529 + 2 225) as per [ITU-T V.8bis]

NOTE – While the operation of DJB2 does not constitute a problem, the change to fixed mode of DJB1 may need special attention. These scenarios are illustrated in Figures 3, 4 and 5.

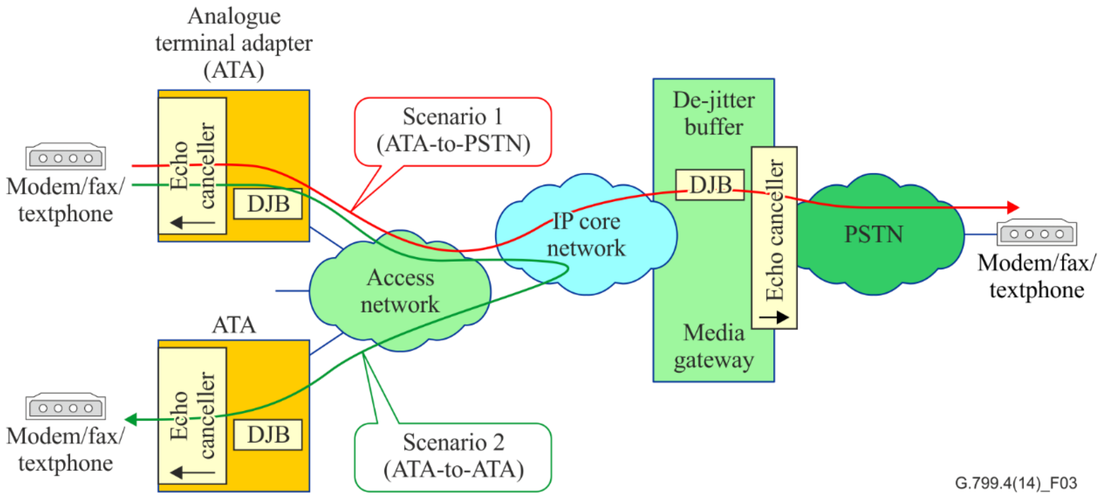


Figure 3 – Diagram showing voice service scenario [ITU-T G.161.1]

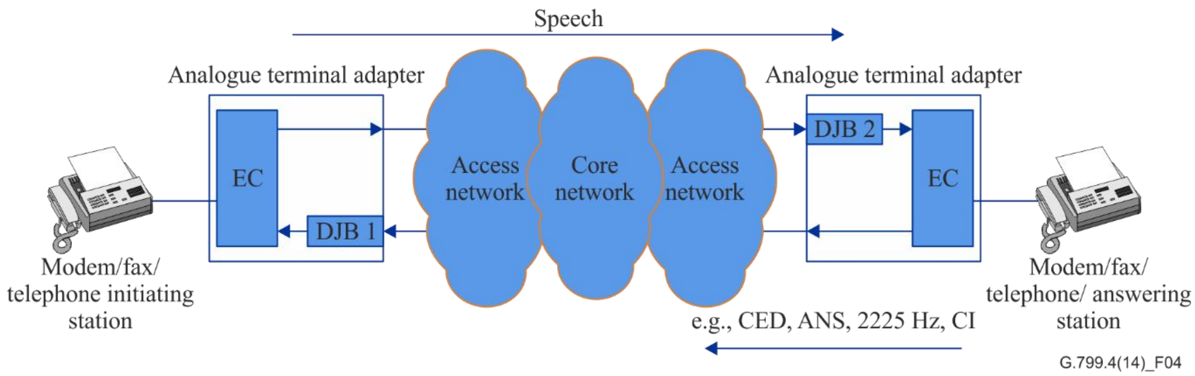


Figure 4 – Activation of de-jitter buffer for VBDoIP between two ATA

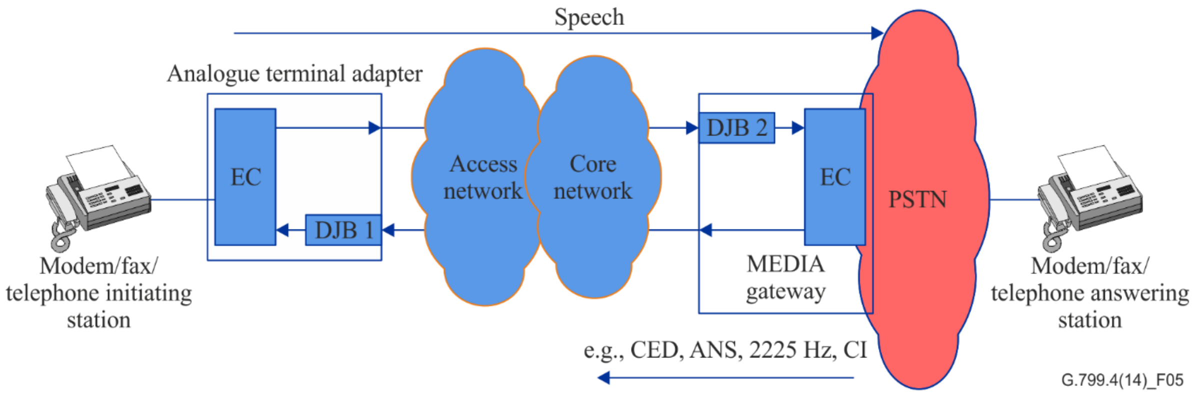


Figure 5 – Activation of de-jitter buffer for VBDoIP between ATA and MGW

Figure 6 recalls again the typical network configuration for PSTN emulation services in NGN. There are two PSTN-IP gateways involved in case of the considered two-party communication by this

Recommendation. The gateways have to provide a dedicated XoIP emulation service, dependent on the PSTN modem call type.

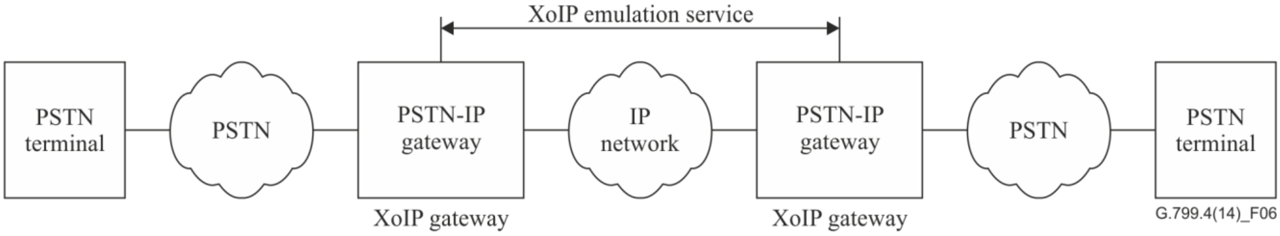


Figure 6 – Typical network configuration for PSTN emulation services

The typical activation and configuration of de-jitter buffers is illustrated by the example of a PSTN fax/modem call, see Figure 7.

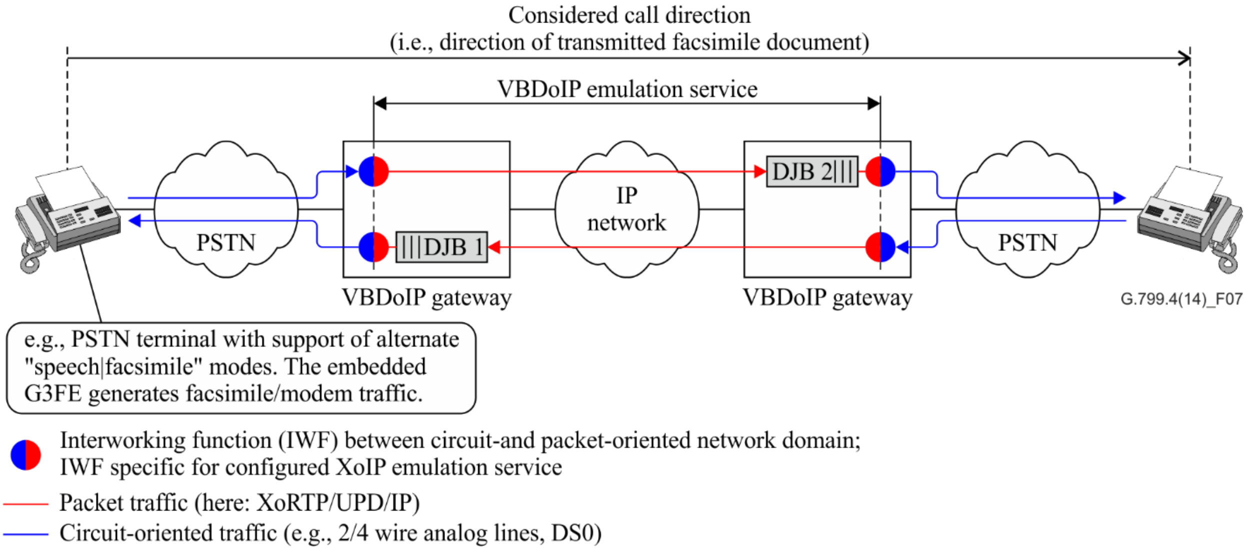


Figure 7 – Example of a VBDoIP emulation service

The VBDoIP gateway provides two main modes of operation at IP side: audio mode (VoIP) and voiceband data mode (VBDoIP), see [ITU-T V.152].

NOTE – It has to be noted that such a state model is undefined in case of pseudo-VBDoIP emulation service (see clause 3.2.1), leading to ambiguity, different interpretations and deployed implementations.

According to [ITU-T V.152], the early VBD detection procedure shall be always used. See Figure 8. In that case, the following initiating calling tones shall be recognized from the detector that shall drive the de-jitter buffer DJB1 and DJB2 into the fixed mode:

– CI (ITU-T V.21 bits) signal ITU-T V.8 (1 180 Hz, 980 Hz, 1 850 Hz, 1 650 Hz)
– CNG, 1 100 Hz, (T.30, ITU-T V.8)
– CT, 1 300 Hz signal (ITU-T V.25)
– a series of interrupted bursts of binary 1 signal
– unscrambled binary ones signal as per [ITU-T V.22] initiating segment 1 dual tones (1 375 Hz and 2 002 Hz) as per [ITU-T V.8bis].

If the calling tones needed to activate the early VBD detection are not detected and the de-jitter buffers are not activated, the signals generated from the called side shall be recognized by the detector that shall drive the de-jitter buffers DJB1 and DJB2 into the fixed mode as described in the previous paragraph. See Figure 9.

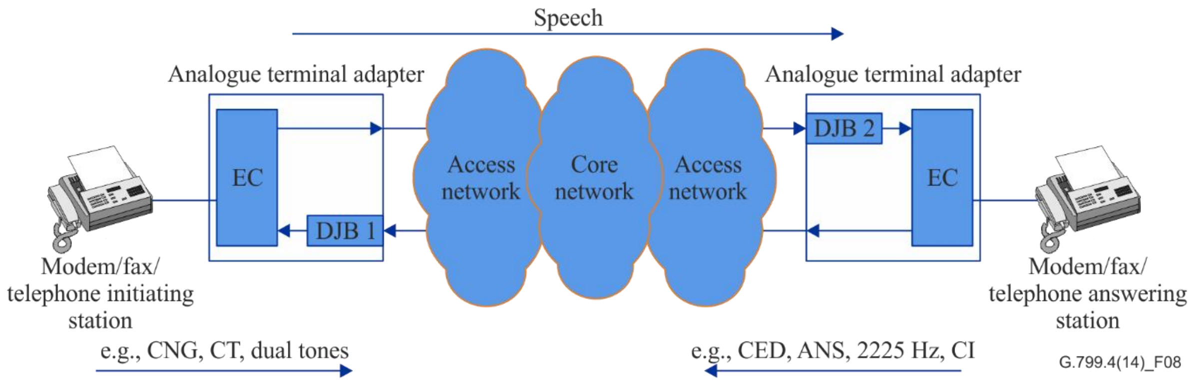


Figure 8 – Early VBD detection procedure

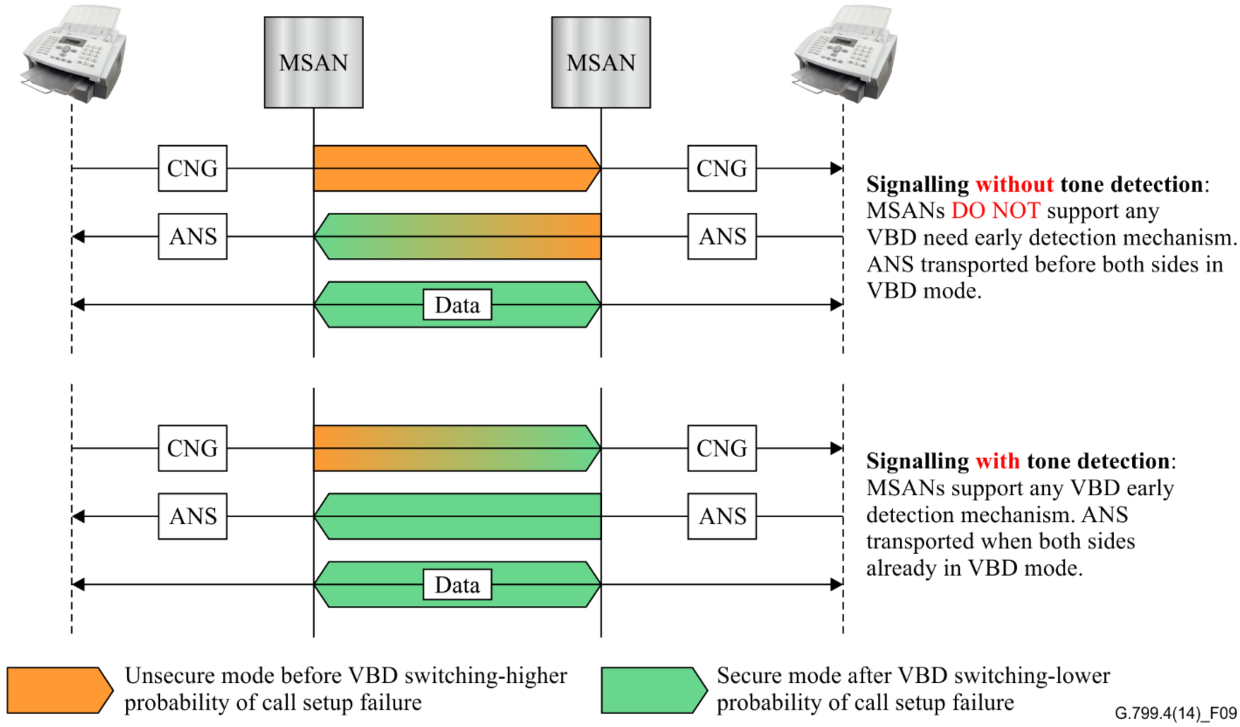


Figure 9 – Example of signalling with and without VBD early detection

To make sure that CNG/CT has been received, two CNG/CT signals must be evaluated (not just one, since the risk of erroneously switching is too high). After the evaluation, the adaptive de-jitter buffer has to switch within 0.9 s to a fixed jitter buffer.

To minimize the risk of modem carrier loss, packet loss concealment (PLC) for voice mode according to Appendix I of [ITU-T G.711] shall be supported, under the following conditions:

Usage of PLC is principally conditional, e.g.,:

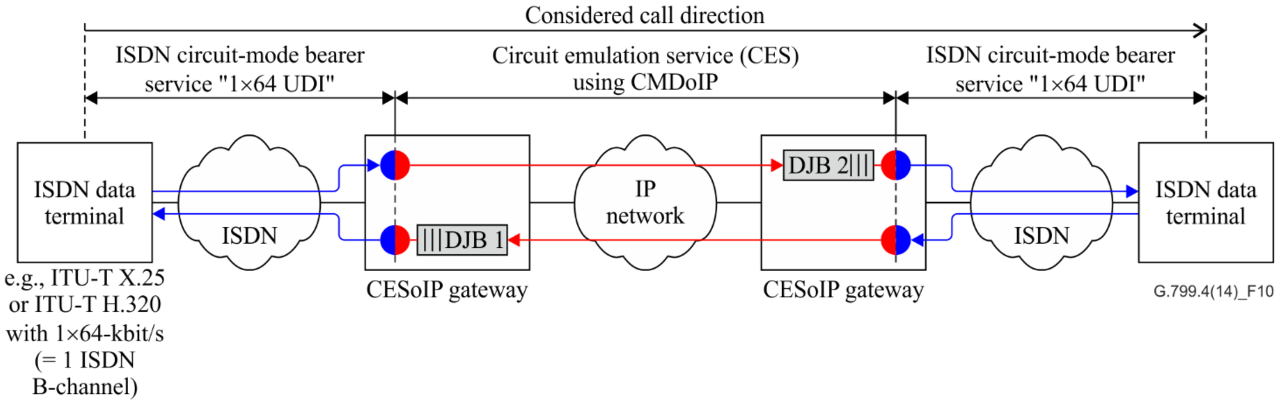
– If gateway configuration for VoIP emulation service, then condition "G.711 as audio codec" applies (but not non-G.711 audio);

– If gateway configuration for VBDoIP emulation service, then condition "G.711 as audio codec" applies.

Usage of such a PLC is also outside the scope for FoIP, ToIP or MoIP emulation services.

8.1 Activation of de-jitter buffer for ISDN bearer service

ISDN provides circuit-mode data services besides audio. The ISDN bearer service [b-ITU-T I.231.1] is used for unrestricted digital information (UDI) and consumes a single ISDN B-channel (64 kbit/s) in the ISDN network domain. IP networks need to support circuit emulation services (CESoIP) for the general category of ISDN circuit-mode data services. The so-called clearmode (CMD), according to [b-IETF RFC 4040], could be used as XoIP emulation service for [b-ITU-T I.231.1] traffic. The protocol stack of the CMDoIP emulation service relates to a "1×64/CMD/RTP/UDP/IP" transport. It has to be noted that general N×64 circuit-mode data services (with N greater than 1) are outside the scope of [b-IETF RFC 4040]. Figure 10 illustrates the example network configuration:



Interworking function (IWF) between circuit-and packet-oriented network domain; IWF specific for configured XoIP emulation service

Packet traffic (here: XoRTP/UPD/IP)

Circuit-oriented traffic (here TDM DS0)

Figure 10 – Example of a CMDoIP emulation service

PSTN voiceband data and ISDN circuit-mode data services share the same requirement of constant end-to-end transfer delay, leading to fixed DJB configurations in the gateways.

However, there is no in-band signalling in case of ISDN circuit-mode data service, in contrast to the modem-based signalling procedures in case of PSTN voiceband data. The particular ISDN bearer service (here [b-ITU-T I.231.1]) is indicated in ISDN call control signalling (such as [ITU-T Q.931] at ISDN UNI or ISUP/SS7 at ISDN NNI. Gateway to gateway signalling in the IP domain again uses no IP in-path signalling (such as in the case of [ITU-T V.152], [ITU-T T.38]), rather an explicit indication at IP call control signalling (e.g., in case of SIP the SDP attribute " `a=rtpmap:… CLEARMODE/8000`" within in the media configuration).

DJB handling is illustrated in Figure 11.

The fixed de-jitter buffer from the calling and called side for a 64 kbit/s bit sequence (UDI) shall be activated directly by call control signalling. The activation takes place at the latest with the reception of call control stimuli such as Connect/ANM (ISUP) in the ISDN domain or a 200 OK (SIP) message in the IP domain.

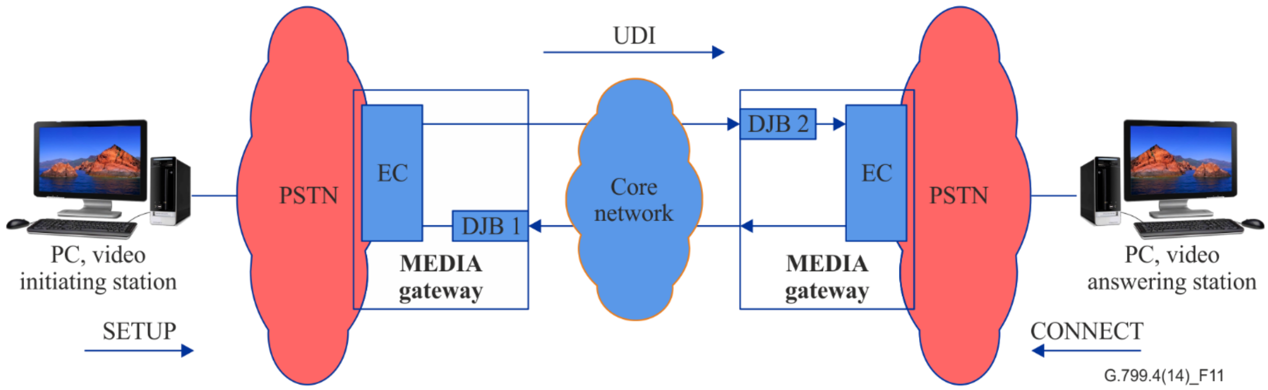


Figure 11 – Activation of de-jitter buffer for clear mode (CMDoIP) activated directly by call control signalling

9 Characteristics of VBD-mode switching of de-jitter buffers

9.1 General

The de-jitter buffer covered by this Recommendation should be equipped with a tone detector that conforms to this clause.

The change of the de-jitter buffer to VBD mode should be based on the following signals (mostly taken out of [ITU-T V.152]) 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]
– /ANS as per [ITU-T V.25]
– 2 225 Hz answer tone as per Appendix VI of [b-ITU-T V.150.1]
– Unscrambled binary ones signal as per [ITU-T V.22]
– CI signals that precede ANSam, as per [ITU-T V.8]/[ITU-T V.21]
– Dual-frequency tones (1 375 Hz + 2 002 Hz and 1 529 Hz + 2 225 Hz) as per [ITU-T V.8bis].

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 [b-ITU-T V.18]
– CI signals that precede ANSam, as per [ITU-T V.8]
– CT (calling tone) signals that precede ANS, as per [ITU-T V.25]
– Initiating Segment 1 dual tones (1 375 Hz and 2 002 Hz) as per [ITU-T V.8bis].

9.2 Detector characteristics

9.2.1 Detector characteristics for frequency range of 2 100 Hz ± 21 Hz

The tone detector shall detect a tone in the frequency range of 2 100 Hz ±21 Hz (see [ITU-T V.21]). The detection channel bandwidth should be chosen wide enough to encompass this tone (and possibly other tones used within national networks). At the same time, the detection channel bandwidth should be such that, in conjunction with guard action and timing, adequate protection is provided against false operation of the detector by speech signals. The detector channel sensitivity (threshold level) should be such that the detector operates on the lowest expected power of the tone. The band characteristics shown in Figure 12 permit the de-jitter buffer behaviour to be changed by the 2 100 Hz tone as well as others used in North America. Figure 12 indicates that in the frequency band 2 079 Hz to 2 121 Hz, detection is certainly possible whilst in the band 1 900 Hz to 2 350 Hz detection may be possible. If only the recommended 2 100 Hz tone is used internationally, interference with signalling equipment will be avoided. The dynamic range of the detector should be consistent with the input levels as specified in [ITU-T V.2] with allowances for variation introduced by the PSTN.

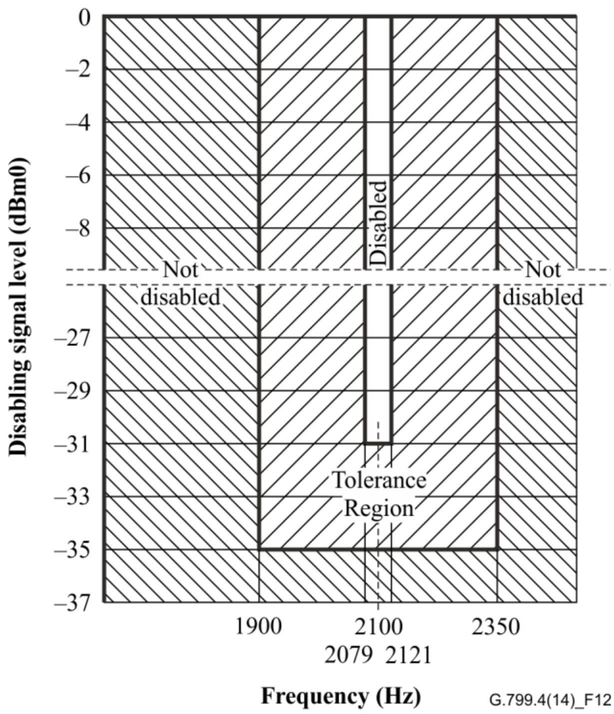


Figure 12 – Required band characteristics

9.2.2 Detector characteristics for dual-frequency tones 1 375 Hz + 2 002 Hz and 1 529 Hz + 2 225 Hz ([ITU-T V.8bis])

The tone detector shall detect two tone segments. The first segment consists of a dual-frequency tone held for 400 ms. The specific frequencies 1 375 Hz + 2 002 Hz are used from the initiator, the specific frequencies 1 529 Hz + 2 225 Hz from the responder in a transaction. When using the telephone-event payload, the V8bISeg and V8bRSeg events in Table 1 represent the first segment of any ITU-T V.8bis signal in the initiating and responding case, respectively.

Table 1 – Events for ITU-T V.8bis signals

Signal	Frequency
V8bISeg	1 375 Hz + 2 002 Hz
V8bRSeg	1 529 Hz + 2 225 Hz

The tolerance of the frequency of all tones shall be ±250 ppm of the nominal value.

The tolerance of the duration of the tone segments shall be ±2%.

The detection channel bandwidth should be chosen wide enough to encompass these tones (and possibly other tones used within national networks). At the same time, the detection channel bandwidth should be such that, in conjunction with guard action and timing, adequate protection is provided against false operation of the detector by speech signals. The detector channel sensitivity (threshold level) should be such that the detector will operate on the lowest expected power of the tone.

9.2.3 Detector characteristics for frequencies 980 Hz, 1 180 Hz, 1 650 Hz, 1 850 Hz [ITU-T V.21]

The tone detector shall detect the frequencies 980 Hz for '1' (mark) and 1 180 Hz for '0' (space) (low channel uses) and the frequencies 1 650 Hz for '1' and 1 850 Hz for '0' (high channel uses). The frequency deviation shall be ±100 Hz.

The detection channel bandwidth should be chosen wide enough to encompass these tones (and possibly other tones used within national networks). At the same time, the detection channel bandwidth should be such that, in conjunction with guard action and timing, adequate protection is provided against false operation of the detector by speech signals. The detector channel sensitivity (threshold level) should be such that the detector operates on the lowest expected power of the tone. See Table 2.

Table 2 – Events for ITU-T V.21 signals

Signal	Frequency (Hz)
ITU-T V.21 channel 1, '0' bit	1 180
ITU-T V.21 channel 1, '1' bit	980
ITU-T V.21 channel 2, '0' bit	1 850
ITU-T V.21 channel 2, '1' bit	1 650

9.2.4 Detector characteristics for 2 100 Hz amplitude-modulated by a sinewave at 15 Hz, 2 100 Hz amplitude-modulated by a sinewave at 15 Hz with phase reversals, 1 300 Hz and 1 100 Hz [ITU-T V.8]

To activate the de-jitter buffer at the calling end or called end for procedures according to [ITU-T V.8], the tone detector shall detect frequencies described in Table 3.

Table 3 – Events for ITU-T V.8 signals

Signal	Frequency
ANSam	2 100 Hz amplitude-modulated by a sinewave at 15 Hz
/ANSam	2 100 Hz amplitude-modulated by a sinewave at 15 Hz with phase reversals at an interval of 450 ± 25 ms
CI	(ITU-T V.21 bits) (see Note)
CT	1 300 Hz
CNG	1 100 Hz
NOTE – CI is transmitted from the calling DCE with a regular ON/OFF cadence. The ON periods shall be not less than three periods of the CI sequence, and not greater than 2 s in duration; the OFF periods shall be not less than 0.4 s and not greater than 2 s in duration. A CI sequence consists of 10 ONEs followed by 10 synchronization bits and the call function octet.	

To initiate a session of data transmission on the PSTN according to [ITU-T V.8], a DCE transmits a CI, CT, CNG or no signal. Signal CI is an ITU-T V.8 alternative to call tone CT, and is coded to indicate a call function. The term "call signal" is used hereinafter to refer to CI, CT or CNG.

Modified answer tone ANSam consists of a sinewave signal at 2 100 ± 1 Hz with phase reversals at an interval of 450 ± 25 ms, amplitude-modulated by a sinewave at 15 ± 0,1 Hz. The modulated envelope shall range in amplitude between (0.8 ± 0.01) and (1.2 ± 0.01) times its average amplitude.

The average transmitted power shall be in accordance with [ITU-T V.2].

The average power outside the band 2 100 ± 200 Hz produced by using an approximation to the 15 Hz sinewave envelope is at least 24 dB below the average power within that band. See Figure 13.

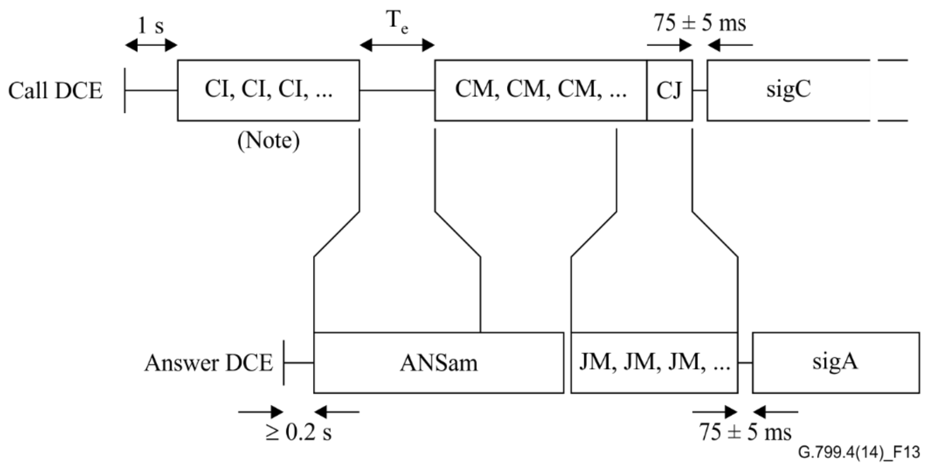


Figure 13 – Use of the CI call signal and exchange of CM/JM menu (Figure 1 of [ITU-T V.8])

The detection channel bandwidth should be chosen wide enough to encompass these tones (and possibly other tones used within national networks). At the same time, the detection channel bandwidth should be such that, in conjunction with guard action and timing, adequate protection is provided against false operation of the detector by speech signals. The detector channel sensitivity (threshold level) should be such that the detector will operate on the lowest expected power of the tone.

9.2.5 Detector characteristics for [ITU-T V.22]

To activate the de-jitter buffer at the calling end or called end for procedures according to [ITU-T V.22], the detector shall detect unscrambled binary 1 for 155 ± 50 ms from the calling terminal and [ITU-T V.25] answer sequence from the called terminal. See Table 4 and Figure 14.

Table 4 – Events for ITU-T V.22/V.25 answer sequence

Signal	Frequency
Answer tone (ANS)	2 100 Hz
Unscrambled binary 1 for 155 ± 50 ms from the calling terminal	

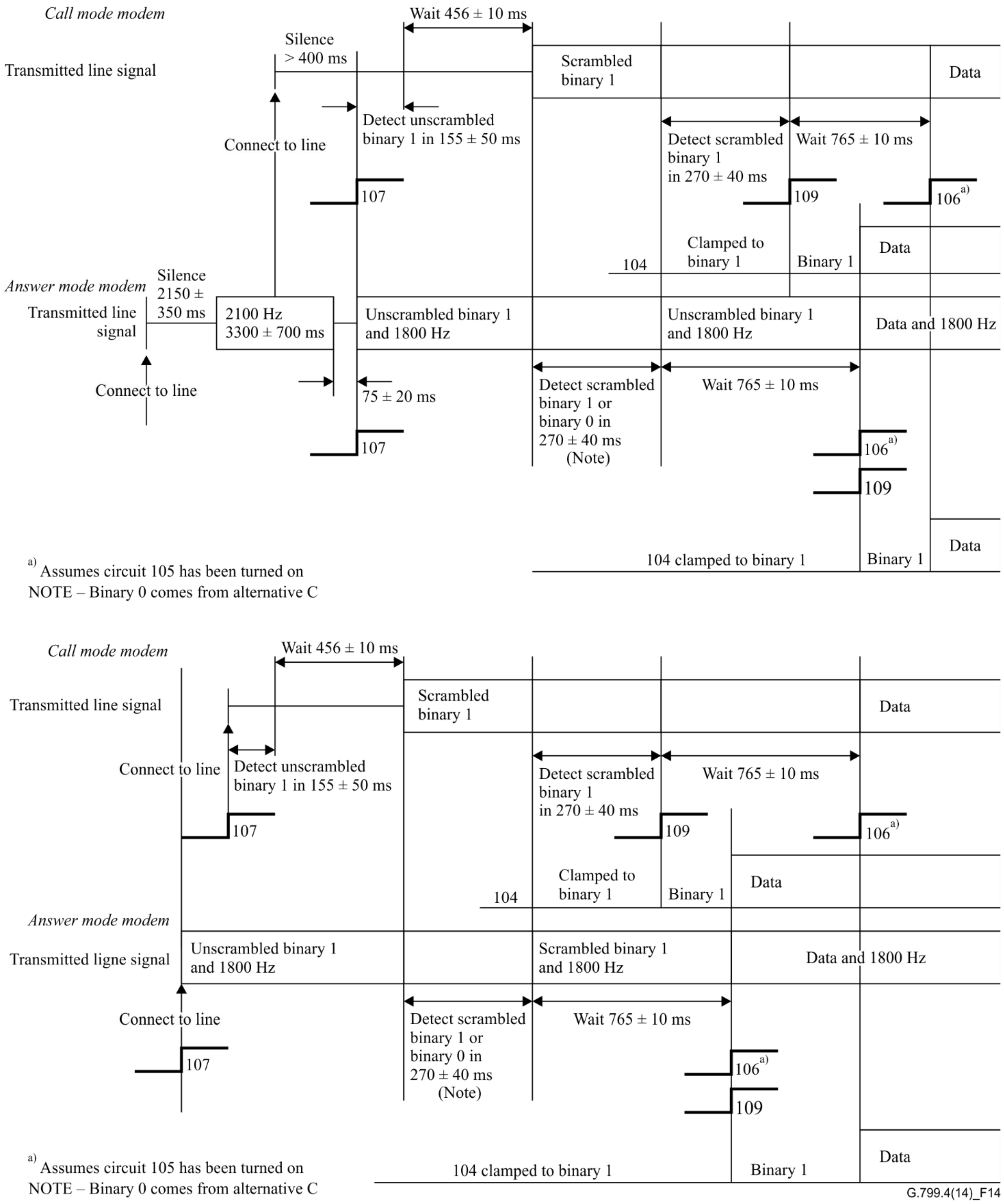


Figure 14 – Handshake sequence for Alternatives A and B (with ITU T V.25 auto-answering)

9.2.6 Detector characteristics for 2 100 Hz with phase reversals [ITU-T V.25]

[ITU-T V.25] specifies the exchange of two tone signals: CT and ANS.

To activate the de-jitter buffer at the calling end or called end for procedures according to [ITU-T V.25], the tone detector shall detect frequencies described in Table 5.

Table 5 – Events for [ITU-T V.25] signals

Signal	Frequency
Answer tone (ANS)	2 100 Hz
/ANS	2 100 Hz with phase reversals at an interval of 450 ± 25 ms
CT	1 300 Hz

The CT is transmitted from the calling end. This may be 1 300 Hz or any tone corresponding to binary 1 of the DCE. The CT and calling station response should not contain power in the band 2 100 ± 250 Hz. The power levels of the signals specified in this Recommendation shall conform to the levels specified in [ITU-T V.2].

The CT consists of a series of interrupted bursts at 1 300 Hz, on for a duration of not less than 0.5 s and not more than 0.7 s and off for a duration of not less than 1.5 s and not more than 2.0 s. See Figures 15, 16 and 17.

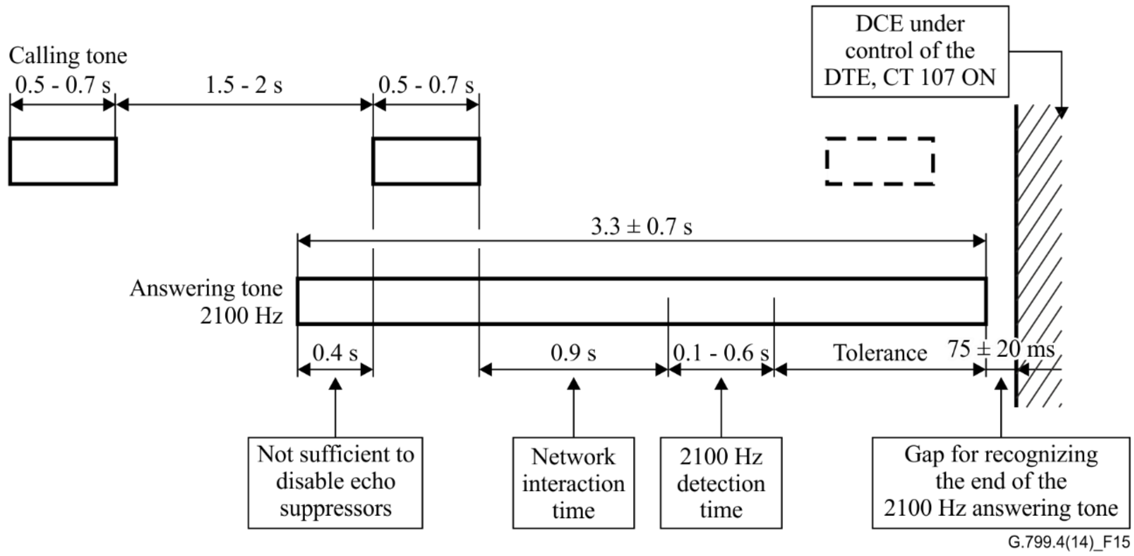


Figure 15 – Timing of line signals

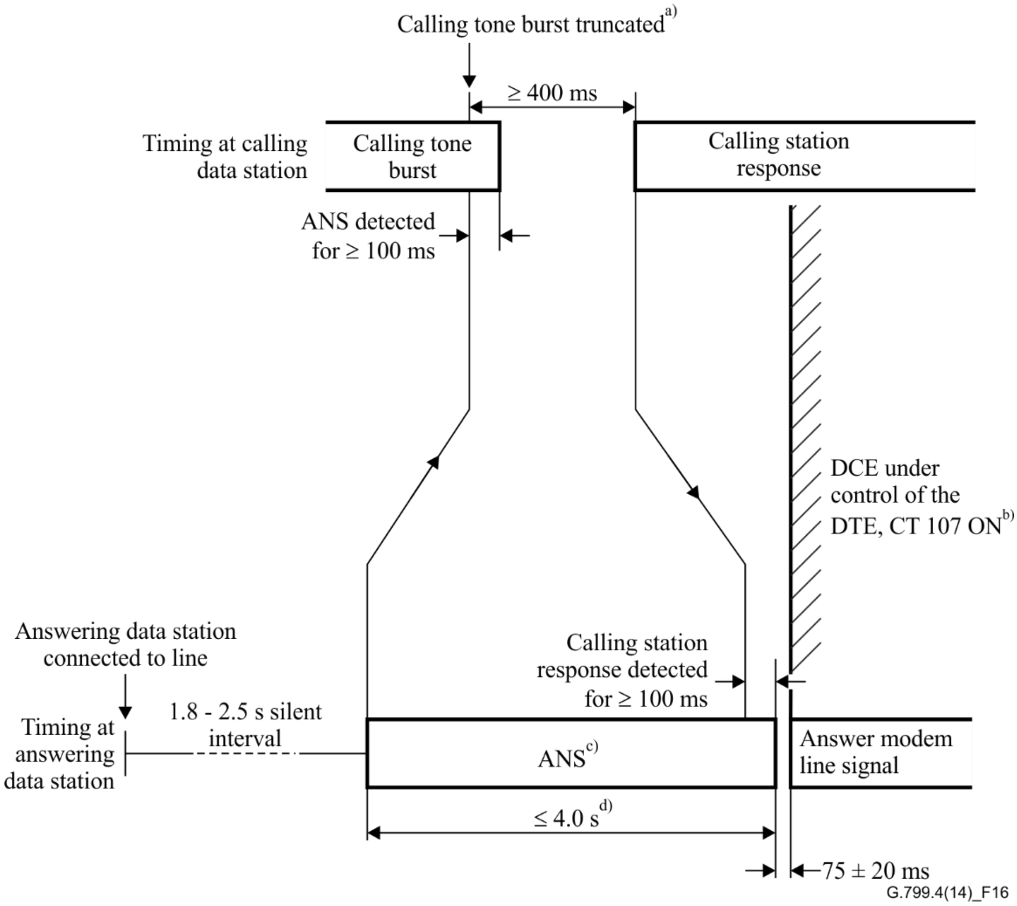
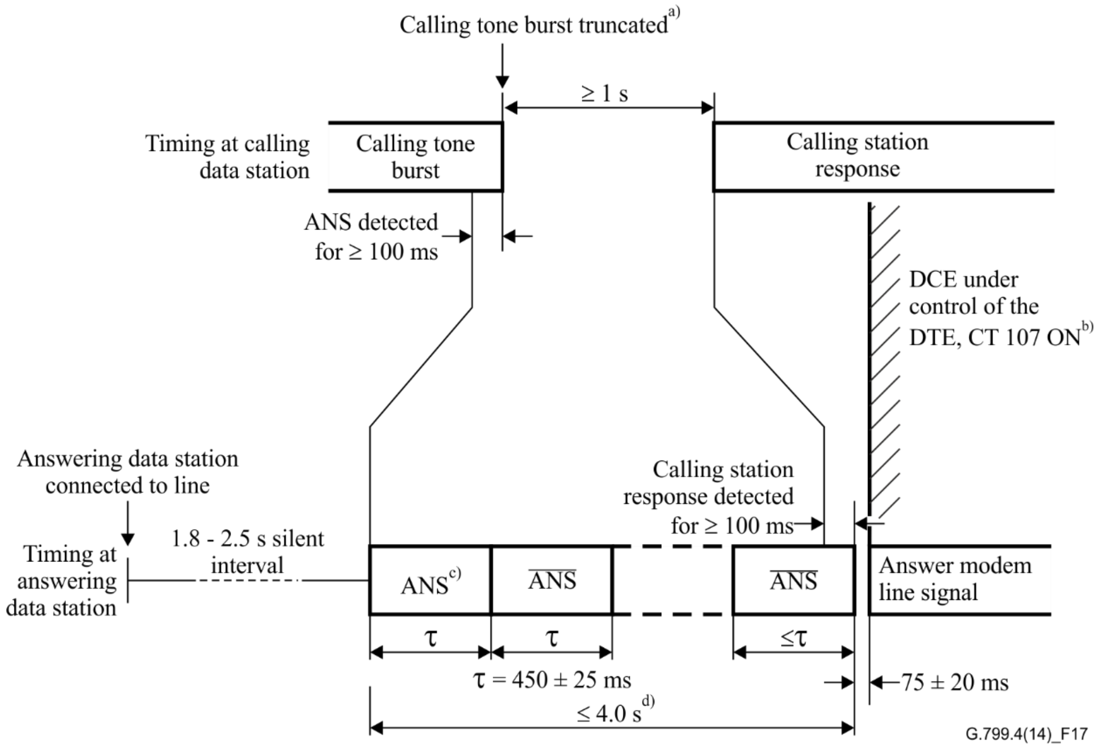


Figure 16 – Timing of line signals – Optional calling station response



a) If ANS is detected during a calling tone burst, the burst may be truncated. If it is not truncated, the calling station response must be delayed until at least 1 s after the end of the burst.

b) ANS denotes the answer tone. $\overline{\text{ANS}}$ denotes the answer tone with its phase reversed.

c) The answer tone duration must be at least 2.6 s if a calling station response is not received.

Figure 17 – Timing of line signals, provision for de-jitter buffers enabling

9.2.7 Detector characteristics for ITU-T V.32/V.32bis

Operating over the public telephone network, the start-up follows the ITU-T V.25 answering procedure (see clause 5.2.4).

[ITU-T V.32] describes a modem type using phase-shift keying with quadrature amplitude modulation. It operates on a carrier at 1 800 Hz, modulated at 2 400 symbols/s. The basic data rates for [ITU-T V.32] are 4 800 bits/s and 9 600 bits/s. [ITU-T V.32bis] extends the data rates up to 14 400 bits/s.

9.2.8 Detector characteristics for [b-ITU-T T.30]

To activate the de-jitter buffer at the calling end or called end for procedures according to [b-ITU-T T.30], the tone detector shall detect frequencies described in Table 6. See Figure 18.

Table 6 – Events for ITU-T T.30 signals

Signal	Frequency
CED (Called tone which is physically identical to ITU-T V.25 ANS)	2 100 Hz
/CED Called tone which is physically identical to ITU-T V.25 /ANS)	2 100 Hz with phase reversals at an interval of 450 ± 25 ms
CEDam Called tone which is physically identical to ITU-T V.25 ANSam)	2 100 Hz amplitude-modulated by a sinewave at 15 Hz (clauses 4.1.2 and 6 of [b-ITU-T T.30]) [b-ITU-T V.34]
/CEDam Called tone which is physically identical to ITU-T V.25 ANSam)	2 100 Hz amplitude-modulated by a sinewave at 15 Hz with phase reversals at an interval of 450 ± 25 ms, See Note
CNG (Calling tone)	1 100 Hz
ITU-T V.21 preamble flag	(ITU-T V.21 bits)

NOTE — Clause 11.1.2.1 of [b-ITU-T V.34] states: "Upon connection to line, the modem shall initially remain silent for a minimum of 200 ms and then transmit signal ANSam according to the procedure in Recommendation V.8. If duplex operation is intended, this signal shall include phase reversals as specified in Recommendation V.8. If half-duplex operation is intended, phase reversals are optional. The modem shall condition its receiver to detect CM and, possibly, calling modem responses from other appropriate Recommendations".

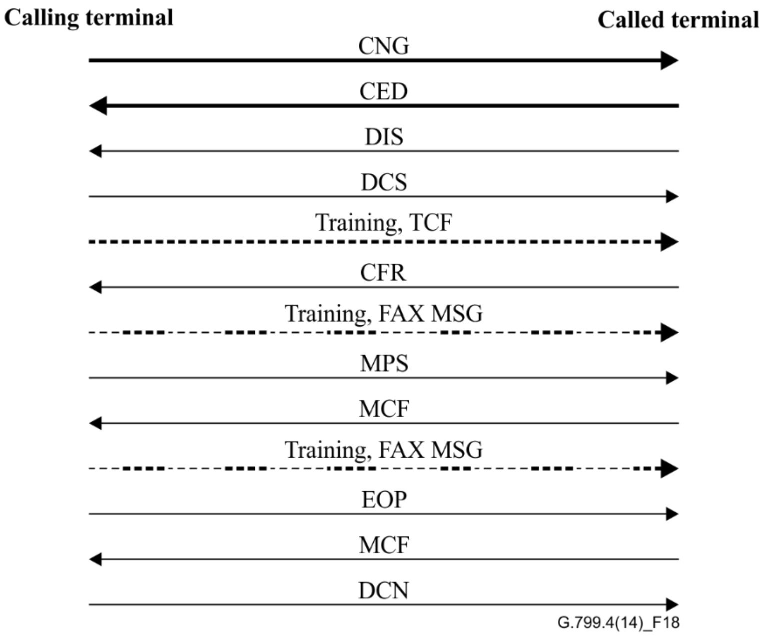


Figure 18 – ITU-T T.30 procedure

The detection channel bandwidth should be chosen wide enough to encompass these tones (and possibly other tones used within national networks). At the same time, the detection channel bandwidth should be such that, in conjunction with guard action and timing, adequate protection is provided against false operation of the detector by speech signals. The detector channel sensitivity (threshold level) should be such that the detector operates on the lowest expected power of the tone.

9.2.9 Noise tolerance

The detector should operate correctly with white noise less than or equal to 11 dB below the level of the signal which should be detected. No definitive guidelines can be given for the range between 5 dB and 11 dB because of the variations in the test equipment used. In particular, performance may vary with the peak-to-average ratio of the noise generator used. It is noted that it is possible to design a detector capable of operating correctly at 5 dB signal-to-noise ratio.

9.2.10 Operate time

The operate time should be sufficiently long to provide immunity from false operation due to voice signals, but not so long as to needlessly extend the time to disable. The de-jitter buffer activator is required to operate within 1 s of the receipt of the activating signal.

9.2.11 False operation due to speech signals

It is desirable that the de-jitter buffer activator should rarely operate falsely on speech signals. To this end, a reasonable objective is that, for a de-jitter buffer installed on a working circuit, usual speech signals should not on the average cause more than 10 false operations during 100 h of speech. In addition to the talk-off protection supplied by the disabling channel bandwidth, by guard band operation and by the operate time, talk-off protection can be supplied by recycling. That is, if speech which simulates the signal is interrupted because of inter-syllabic periods, before the change in de-jitter buffer behaviour has taken place, the operate timing mechanism should reset. However, momentary absence or change of level in a true signal should not reset the timing.

9.2.12 Release time

For further study.

9.2.13 Other considerations

Both the echo of the activating tone and the echo of the calling tone may disturb the detection of the de-jitter buffer enabling tone. As such, it is not recommended that the receive and transmit signal inputs be added together to form an input to a single detector.

10 Requirements for configuration of de-jitter buffers

10.1 Fixed de-jitter buffers

In the case of VBD, it is the goal to keep the audio end-to-end delay constant during the entire call. The de-jitter buffer has to be implemented in such a way that any jitter occurring during the entire call will not change the end-to-end delay.

The de-jitter buffer may adapt if there is an overflow or underrun.

10.2 Adaptive de-jitter buffers

In the case of voice, the strategy of de-jitter buffer implementation is to keep the end-to-end audio delay as low as possible under all jitter conditions. Any de-jitter buffer implementation should mostly not impair the listening speech quality as perceived by the user.

For adaptive de-jitter buffers, the maximum aberration from the real jitter in the network should be one packetization time interval. It is recommended that the jitter measurement period for jitter should be two to three packet intervals, not only a single packet interval. The adaptation interval towards higher values should be done immediately after the jitter measurement period. The adaptation towards lower values should be after at least several seconds or during silence periods.

10.3 Activation procedure into the fixed mode

The detection of the initiating calling tones should last a maximum of 200 ms, after which the de-jitter buffer shall adapt the de-jitter buffer delay from the adaptive to the fixed de-jitter buffer rate. When the DJB adapts to the fixed state, there will be some time (DJB adaption time) without audio information due to the increasing DJB delay, which will often be replaced with supposed audio information by a PLC algorithm.

For some applications, the DJB adaption time can be critical, see Table 7. In this case, the early VBD detection procedure shall be used.

Table 7 – Examples of de-jitter buffer adaption time

De-jitter buffer adaptive	De-jitter buffer fixed	De-jitter buffer adaption time
20 ms	100 ms	40 ms
20 ms	200 ms	90 ms
40 ms	100 ms	30 ms
40 ms	200 ms	80 ms

10.4 Transition from VBD to voice mode ([ITU-T V.152])

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

a) In the direction from the general switched telephone network (GSTN) to IP network of any of the following stimuli:
   1) end of modem or facsimile signals;
   2) voice signals;
   3) silence detection in both directions, PSTN to IP and IP to PSTN, with the following caveats:
      i) for text telephones, the appropriate silence detection shall be considered because text telephone conversations may have long silence periods,
      ii) in the case of facsimile calls, the silence period should be greater than the T2 timer defined in [b-ITU-T T.30];
   4) MGC signalling or other out of band signalling method.

b) 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).

The above described transition criteria are also summarized in Figure 19.

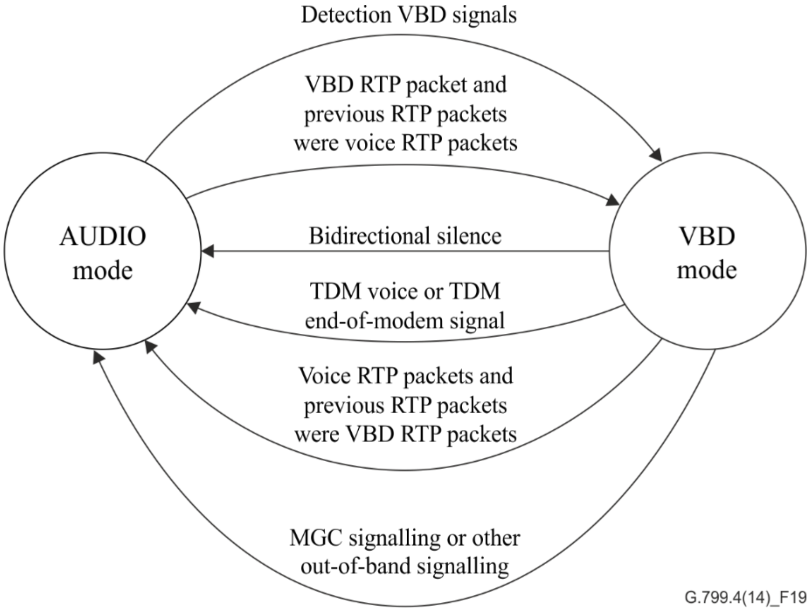


Figure 19 – Voice-VBD Transitioning state diagram (Figure 9 of [ITU-T V.152])

10.5 Handling of de-jitter buffer in case of lost or late packets

As the receiving decoder expects to be fed with voice packets at the same fixed rate, the de-jitter buffer shall insert dummy packets if the packets are lost or if they arrived too late. Depending on the de-jitter buffer algorithm, the lost packets will often be replaced with supposed audio information by a PLC algorithm. See Figure 20.

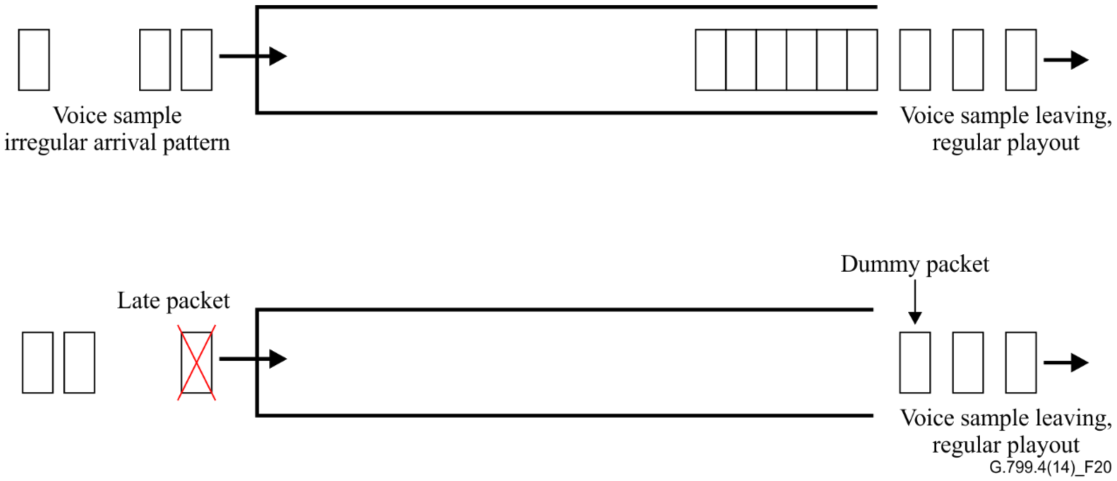


Figure 20 – Handling of de-jitter buffer in case of lost or late packets

Annex A

De-jitter buffer facsimile tests

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

These tests should ensure that the de-jitter buffer located at each end of a connection adjusts rapidly on the initial handshaking sequences of a facsimile and VBD call. The test and requirements were originally developed to overcome problems in the network due to the turnaround of fax and modem handshaking signals.

A.1 Measurement method

The test method is based on the analysis of measurement of delay over time of the device under test. Artificial introduced jitter can be used to help with this analysis. See Figures A.1 and A.2.

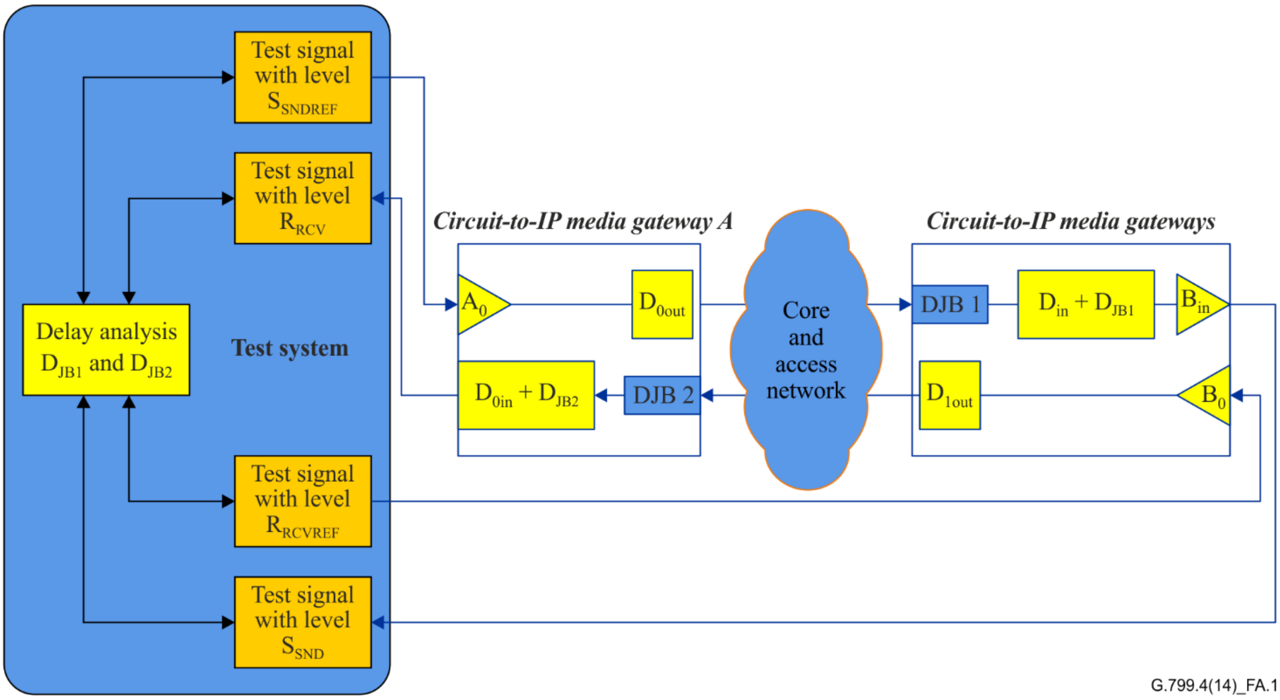


Figure A.1 – Test configuration, A calls B

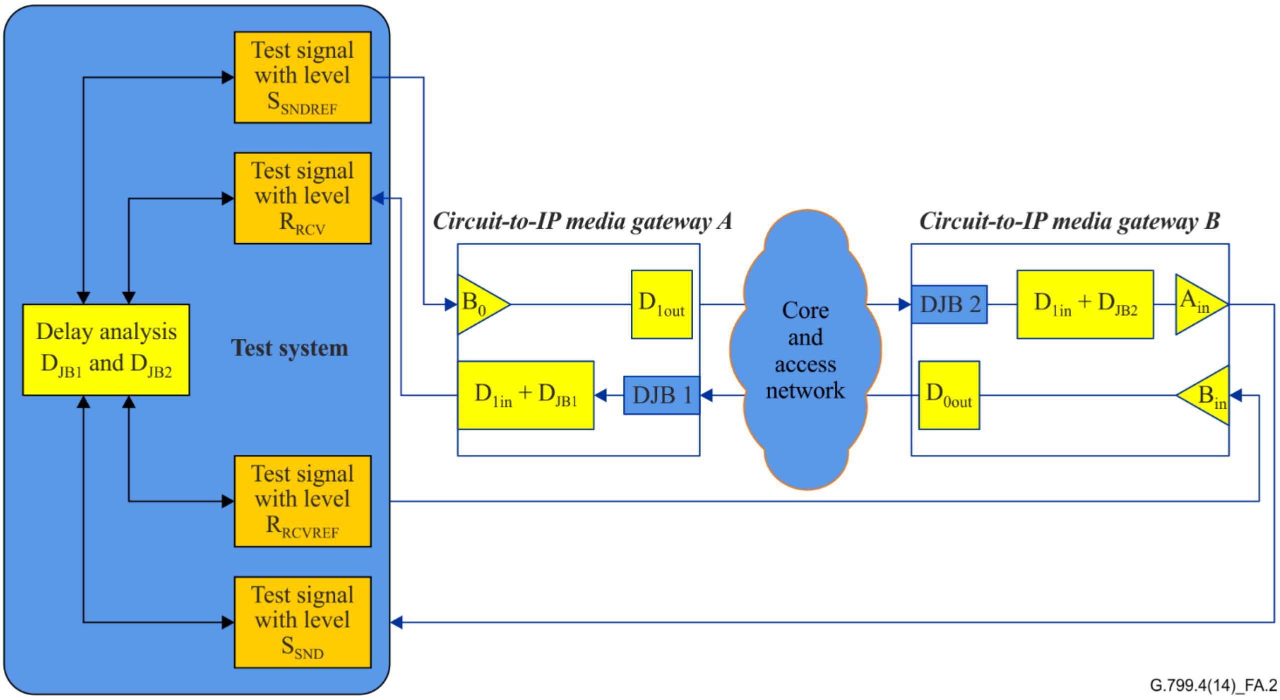


Figure A.2 – Test configuration, B calls A

A_0 – output level interface A

A_{in} – input level interface A

B_{in} – input level interface B

B_0 – output level interface B

D_{0out} – sending delay interface A (coder delay, see Table 22 of [ITU-T V.8])

D_{JB1} – de-jitter buffer delay interface B

D_{1in} – receiving delay interface B (Decompression time per block + Serialization time + PLC)

D_{1out} – sending delay interface B (coder delay, see Table 22 of [ITU-T V.8])

D_{JB2} – de-jitter buffer delay interface A

D_{0in} – receiving delay interface B (Decompression time per block + Serialization time + PLC)

S_{SNDREF}, R_{RCVREF} = Transmitted signals

S_{snd}, R_{rcv} = Network Transmitted and Received attenuations

NOTE – Decoder delay = $D_{in} + D_{JB}$

Used signals

C16	Signal of Test 2A ([ITU-T G.168]; [ITU-T P.501]), average level −16 dBm0 Gaussian white noise signal which is used to identify the echo-path impulse response.
ANS	2 100 Hz sine ([ITU-T V.25]) The duration of the tone is set to T1 = 1.35 s.
ANSam	2 100 Hz sine with a 20% amplitude modulation by a 15 Hz sine ([ITU-T V.25]).
/ANS	2 100 Hz sine with 180° phase shift every 450 ms ([ITU-T V.25])
/ANSam	2 100 Hz sine with a 20% amplitude modulation by a 15 Hz sine and 180° phase shift every 450 ms ([ITU-T V.25]).
CI	CI sequence consists of 10 ONEs followed by 10 synchronization bits and the call function octet. For the transmission: 980 Hz for '1' (mark) and 1 180 Hz for '0' (space) (low channel uses); and the frequencies 1 650 Hz for '1' and 1 850 Hz for '0' (high channel uses) are used.

CT	1 300 Hz sine CT (calling tone) consists of a series of interrupted bursts of a 1 300 Hz tone, on for a duration of not less than 0.5 s and not more than 0.7 s and off for a duration of not less than 1.5 s and not more than 2.0 s.

CNG	1 100 Hz sine Duration On for 0.5 s to 0.7 s, Off 1.8 s to 2.5 ([ITU-T V.25])

FAX	Sequence No. 1 (clause 6.4.2.11 of [ITU-T G.168])

Table A.1 provides a summary of the various modem and fax tests that are detailed in the subsequent unnumbered tables. (M = mandatory, O = optional)

Table A.1 – Test overview

Test number	Transmission type	
Fax tests		**Options**
1.1.1.1 FAX	Facsimile with ANS / −12 dBm0	M
1.1.1.2 FAX	Facsimile with ANS / −31 dBm0	M
1.1.2.1 FAX	Facsimile with early VBD detection with CNG / −12 dBm0	O
1.1.2.2 FAX	Facsimile with early VBD detection with CNG / −31 dBm0	O
1.1.3.1 FAX	Facsimile with early VBD detection with V.17 data transmission with ANS / −12 dB	O
1.1.3.2 FAX	Facsimile with early VBD detection with V.17 data transmission with ANS / −31 dBm0	O
1.1.4.1 FAX	Facsimile with /ANS / −12 dBm0	M
1.1.4.2 FAX	Facsimile with /ANS / −31 dBm0	M
1.1.5.1_FAX	Facsimile with V.34 data transmission with /ANS / −12 dBm0	M
1.1.5.2 FAX	Facsimile with V.34 data transmission with /ANS / −31 dBm0	M
1.1.6.1 FAX	Facsimile with /ANSam / −12 dBm0	M
1.1.6.2 FAX	Facsimile with /ANSam / −31 dBm0	M
1.1.7.1 FAX	Facsimile with V.34 data transmission with /ANSam / −12 dBm0	M
1.1.7.2 FAX	Facsimile with V.34 data transmission with /ANSam / −31 dBm0	M
Modem tests		**Options**
2.1.1 MODEM	Modem with ANS / −12 dBm0	M
2.1.2 MODEM	Modem with ANS / −31 dBm0	M
2.1.3 MODEM	Modem with early VBD detection with CT / −12 dBm0	O
2.1.4 MODEM	Modem with early VBD detection with CT / −31 dBm0	O
2.2.1 MODEM	Modem with ANSam / −12 dBm0	M
2.2.2 MODEM	Modem with ANSam / −31 dBm0	M
2.3.1 MODEM	Modem with early VBD detection with CI Signal (ITU-T V.8) / −12 dBm0	O
2.3.2 MODEM	Modem with early VBD detection with CI Signal (ITU-T V.8) / −31 dBm0	O
2.4.1 MODEM	Modem with /ANS / −12 dBm0	M
2.4.2 MODEM	Modem with /ANS / −31 dBm0	M
2.5.1 MODEM	Modem with /ANSam / −12 dBm0	M
2.5.2 MODEM	Modem with /ANSam / −31 dBm0	M

Test number:	**1.1.1.1 FAX**
Transmission type:	Facsimile with **ANS / ** −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and Jitter • Buffer 2 Apply signal ANS to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending ANS (from B) • Apply signal C16 to Interface B and determine level S_{SND} and R_{CV} and the delay D_{JB2} after sending ANS (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay DJB1 • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANS to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANS (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	
Called station identification (CED) conditions	CED (ANS) 2 100 Hz ± 15 Hz sine (ITU-T V.25) Duration 3 s The amplitude of the tone is −12 dBm0

Test number:	**1.1.1.2 FAX**
Transmission type:	Facsimile with **ANS** / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANS to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending ANS (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending ANS (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANS to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANS (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	
Called station identification (CED) conditions	CED (ANS) 2 100 Hz ± 15 Hz sine (ITU-T V.25]) Duration 3 s The amplitude of the tone is −31 dBm0

Test number:	**1.1.2.1 FAX**
Transmission type:	Facsimile with early VBD detection with CNG / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CNG to Interface A • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending CNG from A • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending CNG from A II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CNG to Interface B • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending CNG from B • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending CNG from B
Requirement	I) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed II) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	CNG 1 100 Hz sine Duration On for 0.5 s to 0.7 s, Off 1.8 s to 2.5 The amplitude of the tone is −12 dBm0
Called station identification (CED) conditions	

Test number:	**1.1.2.2 FAX**
Transmission type:	Facsimile with early VBD detection with CNG / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CNG to Interface A • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending CNG from A • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending CNG from A II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CNG to Interface B • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending CNG from B • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending CNG from B
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second Call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay jitter for Voice b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
FAX test sequences: **Calling tone (CNG) conditions**	CNG 1 100 Hz sine Duration On for 0.5 s to 0.7 s, Off 1.8 s to 2.5 The amplitude of the tone is −31 dBm0
Called station identification (CED) conditions	

Test number:	**1.1.3.1 FAX**
Transmission type:	Facsimile with early VBD detection with V.17 data transmission with ANS / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CNG to Interface A • Apply signal ANS to Interface B • Apply transmission of FAX • Apply signal C16 to Interface A and determine the delay D_{JB1} after transmission of FAX • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending FAX • The transmission of the signal C16 shall be without time interruption after the transmission of the FAX transmission II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CNG to Interface B • Apply signal ANS to Interface A • Apply transmission of FAX • Apply signal C16 to Interface B and determine the delay D_{JB1} after transmission of FAX from B • Apply signal C16 to Interface A and determine the delay D_{JB2} after transmission of FAX • The transmission of the signal C16 shall be without time interruption after the transmission of the FAX transmission
Requirement	I) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; JB1 = D_{JB2} for Delay for De-jitter buffer Fixed II) a) First call; D_{JB1} = D_{JB2} Delay jitter for Voice b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	CNG 1 100 Hz sine Duration On for 0.5 s to 0.7 s, Off 1.8 s to 2.5 The amplitude of the tone is −12 dBm0
Called station identification (CED) conditions	CED (ANS) 2 100 Hz ± 15 Hz sine (ITU-T V.25) Duration 3 s The amplitude of the tone is −12 dBm0

Test number:	**1.1.3.2 FAX**
Transmission type:	Facsimile with early VBD detection with V.17 data transmission with ANS / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CNG to Interface A • Apply signal ANS to Interface B • Apply transmission of FAX • Apply signal C16 to Interface A and determine the delay D_{JB1}after transmission of FAX • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending FAX • The transmission of the signal C16 shall be without time interruption after the transmission of the FAX transmission II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CNG to Interface B • Apply signal ANS to Interface A • Apply transmission of FAX • Apply signal C16 to Interface B and determine the delay D_{JB1} after transmission of FAX from B • Apply signal C16 to Interface A and determine the delay D_{JB2} after transmission of FAX • The transmission of the signal C16 shall be without time interruption after the transmission of the FAX transmission
Requirement	I) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed II) a) First call; D_{JB1} = D_{JB2} Delay jitter for Voice b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed
FAX test sequences: **Calling tone (CNG) conditions**	CNG 1 100 Hz sine Duration On for 0.5 s to 0.7 s, Off 1.8 s to 2.5 The amplitude of the tone is −31 dBm0
Called station identification (CED) conditions	CED (ANS) 2 100 Hz ± 15 Hz sine (ITU-T V.25) Duration 3 s The amplitude of the tone is −31 dBm0

Test number:	**1.1.4.1_FAX**
Transmission type:	Facsimile with /ANS / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANS to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending /ANS (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANS (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANS to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending /ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANS (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
FAX test sequences: **Calling tone (CNG) conditions**	
Called station identification (CED) conditions	/ANS – 2 100 Hz ± 15 Hz sine (ITU-T V.25) with 180° phase shift every 450 ms (ITU-T V.25) Duration 3 s The amplitude of the tone is −12 dBm0

Test number:	**1.1.4.2_FAX**
Transmission type:	Facsimile with /ANS / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANS to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending /ANS (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANS (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANS to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending /ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANS (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	
Called station identification (CED) conditions	/ANS 2 100 Hz ± 15 Hz sine (ITU-T V.25) with 180° phase shift every 450 ms (ITU-T V.25) Duration 3 s The amplitude of the tone is −31 dBm0

Test number:	**1.1.5.1_FAX**
Transmission type:	Facsimile with V.34 data transmission with /ANS / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal /ANS to Interface B • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply transmission of FAX • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending FAX from interface A • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending FAX from interface A II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANS to Interface A • Apply transmission of FAX • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending FAX from B • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending FAX from B
Requirement	I) a) First call;D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed II) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	
Called station identification (CED) conditions	/ANS 2 100 Hz ± 15 Hz sine (ITU-T V.25) with 180° phase shift every 450 ms (ITU-T V.25) Duration 3 s The amplitude of the tone is −12 dBm0

Test number:	**1.1.5.2_FAX**
Transmission type:	Facsimile with V.34 data transmission with /ANS / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal /ANS to Interface B • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply transmission of FAX • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending FAX from interface A • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending FAX from interface A II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANS to Interface A • Apply transmission of FAX • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending FAX from B • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending FAX from B
Requirement	I) a) First call;$D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	
Called station identification (CED) conditions	/ANS 2 100 Hz ± 15 Hz sine (ITU-T V.25) with 180° phase shift every 450 ms (ITU-T V.25) Duration 3 s The amplitude of the tone is −31 dBm0

Test number:	**1.1.6.1_FAX**
Transmission type:	Facsimile with /ANSam / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending /ANSam (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANSam (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending /ANSam (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending /ANSam (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
FAX test sequences: **Calling tone (CNG) conditions**	
Called station identification (CED) conditions	/ANSam – 2 100 Hz ± 15 Hz sine (ITU-T V.25) 2 100 Hz amplitude-modulated by a sine wave at 15 Hz with phase reversals at an interval of 450 ± 25 ms Duration 3 s The amplitude of the tone is −12 dBm0

Test number:	**1.1.6.2_FAX**
Transmission type:	Facsimile with /ANSam / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending /ANSam (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANSam (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending /ANSam (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending /ANSam (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
FAX test sequences: **Calling tone (CNG) conditions**	
Called station identification (CED) conditions	/ANSam 2 100 Hz ± 15 Hz sine (ITU-T V.25) 2 100 Hz amplitude-modulated by a sine wave at 15 Hz with phase reversals at an interval of 450 ± 25 ms Duration 3 s The amplitude of the tone is −31 dBm0

Test number:	**1.1.7.1_FAX**
Transmission type:	Facsimile with V.34 data transmission with /ANSam / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface B • Apply transmission of FAX • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending /ANSam (from B) + fax transmission • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANSam (from B) + fax transmission II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface A • Apply transmission of FAX • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending fax transmission • Apply signal C16 to Interface A and determine the delay D_{JB2} after fax transmission
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	
Called station identification (CED) conditions	/ANSam 2 100 Hz ± 15 Hz sine (ITU-T V.25) 2 100 Hz amplitude-modulated by a sine wave at 15 Hz with phase reversals at an interval of 450 ± 25 ms Duration 3 s The amplitude of the tone is −12 dBm0

Test number:	**1.1.7.2_FAX**
Transmission type:	Facsimile with V.34 data transmission with /ANSam / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface B • Apply transmission of FAX • Apply signal C16 to Interface A and determine the delay D_{JB1} after sending /ANSam (from B) + fax transmission • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANSam (from B) + fax transmission II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface A • Apply transmission of FAX • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending fax transmission • Apply signal C16 to Interface A and determine the delay D_{JB2} after fax transmission
Requirement	I) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed II) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed
FAX test sequences: Calling tone (CNG) conditions	
Called station identification (CED) conditions	/ANSam 2 100 Hz ± 15 Hz sine (ITU-T V.25) 2 100 Hz amplitude-modulated by a sine wave at 15 Hz with phase reversals at an interval of 450 ± 25 ms Duration 3 s The amplitude of the tone is −31 dBm0

De-jitter buffer Modem tests

Test number:	**2.1.1 MODEM**
Transmission type:	Modem with ANS / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANS to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending ANS (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending ANS (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANS to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANS (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call;$D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
Modem test sequences: Calling tone conditions	
Called station identification conditions	ANS 2 100 Hz ± 15 Hz sine (ITU-T V.25) Duration 2.6 s to 4 s The amplitude of the tone is −12 dBm0

Test number:	**2.1.2 MODEM**
Transmission type:	Modem with ANS / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANS to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending ANS (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending ANS (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANS to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANS (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call;$D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
Modem test sequences: Calling tone conditions	
Called station identification conditions	ANS 2 100 Hz ± 15 Hz sine (ITU-T V.25) Duration 2.6 s to 4 s The amplitude of the tone is −31 dBm0

Test number:	**2.1.3 MODEM**
Transmission type:	Modem with early VBD detection with CT / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CT, 1 300 Hz signal (ITU-T V.25) to Interface A • Apply signal C16 to Interface A and determine the delay D_{JB1} after CT, 1 300 Hz signal (ITU-T V.25) to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending CT, 1 300 Hz signal (ITU-T V.25) to Interface A II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CT, 1 300 Hz signal (ITU-T V.25) to Interface B • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANS (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
Modem test sequences: **Calling tone conditions**	Signal 1 300 Hz Duration On for 0.5 s to 0.7 s, Off for 1.5 s to 2 s The amplitude of the tone is −12 dBm0
Called station identification conditions	

Test number:	**2.1.4 MODEM**
Transmission type:	Modem with early VBD detection with CT / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CT, 1 300 Hz signal (ITU-T V.25) to Interface A • Apply signal C16 to Interface A and determine the delay D_{JB1}after CT, 1 300 Hz signal (ITU-T V.25) to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending CT, 1 300 Hz signal (ITU-T V.25) to Interface A II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CT, 1 300 Hz signal (ITU-T V.25) to Interface B • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANS (from A)
Requirement	I) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call;D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed II) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} De-jitter buffer Fixed
Modem test sequences: **Calling tone conditions**	Signal 1 300 Hz Duration On for 0.5 s to 0.7 s, Off for 1.5 s to 2 s The amplitude of the tone is −31 dBm0
Called station identification conditions	

Test number:	**2.2.1 MODEM**
Transmission type:	Modem with ANSam / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANSam to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending ANSam (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending ANSam (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANSam to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending ANSam (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANSam (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
Modem test sequences: **Calling tone conditions**	
Called station identification conditions	ANSam 2 100 Hz sine with a 20% amplitude modulation by a 15 Hz sine (ITU-T V.25) The amplitude of the tone is −12 dBm0

Test number:	**2.2.2 MODEM**
Transmission type:	Modem with ANSam / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANSam to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending ANSam (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending ANSam (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal ANSam to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending ANSam (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending ANSam (from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
Modem test sequences: **Calling tone conditions**	
Called station identification conditions	ANSam 2 100 Hz sine with a 20% amplitude modulation by a 15 Hz sine [ITU-T V.25] The amplitude of the tone is −31 dBm0

Test number:	**2.3.1 MODEM**
Transmission type:	Modem with early VBD detection with CI Signal (ITU-T V.8) / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CI Signal (ITU-T V.8) to Interface A • Apply signal C16 to Interface A and determine the delay D_{JB1} after sending CI Signal (ITU-T V.8) to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending CI Signal (ITU-T V.8) to Interface A II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CI Signal (ITU-T V.8) to Interface B • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending CI Signal (ITU-T V.8) to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending CI Signal (ITU-T V.8) to Interface B
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call;$D_{JB1} = D_{JB2}$ for Delay De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call;$D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
Modem test sequences: **Calling tone conditions**	CI Signal (ITU-T V.8) The amplitude of the tone is −12 dBm0
Called station identification conditions	

Test number:	**2.3.2 MODEM**
Transmission type:	Modem with early VBD detection with CI Signal (ITU-T V.8) / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CI Signal (ITU-T V.8) to Interface A • Apply signal C16 to Interface A and determine the delay D_{JB1} after sending CI Signal (ITU-T V.8) to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending CI Signal (ITU-T V.8) to Interface A II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal CI Signal (ITU-T V.8) to Interface B • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending CI Signal (ITU-T V.8) to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending CI Signal (ITU-T V.8) to Interface B
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
Modem test sequences: **Calling tone conditions**	CI Signal (ITU-T V.8) The amplitude of the tone is −31 dBm0
Called station identification conditions	

<table>
<tr><td>Test number:</td><td>2.4.1 MODEM</td></tr>
<tr><td>Transmission type:</td><td>Modem with /ANS / −12 dBm0</td></tr>
<tr><td>Measurement procedure</td><td>I)
• Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2
• Apply signal C16 to Interface A and determine the delay D_{JB1}
• Apply signal C16 to Interface B and determine the delay D_{JB2}
• Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2
• Apply signal /ANS to Interface B
• Apply signal C16 to Interface A and determine the delay D_{JB1} after sending /ANS (from B)
• Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANS (from B)
II)
• Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2
• Apply signal C16 to Interface B and determine the delay D_{JB2}
• Apply signal C16 to Interface A and determine the delay D_{JB1}
• Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2
• Apply signal /ANS to Interface A
• Apply signal C16 to Interface B and determine the delay D_{JB1} after sending /ANS (from A)
• Apply signal C16 to Interface A and determine the delay D_{JB2} after sending /ANS (from A)</td></tr>
<tr><td>Requirement</td><td>I)
a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive
b) Second call;D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed
II)
a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive
b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed</td></tr>
<tr><td>Modem test sequences:
Calling tone conditions</td><td></td></tr>
<tr><td>Called station identification conditions</td><td>/ANS 2 100 Hz sine with 180° phase shift every 450 ms (ITU-T V.25)
Duration 2.6 s to 4 s
The amplitude of the tone is −12 dBm0</td></tr>
</table>

Test number:	**2.4.2 MODEM**
Transmission type:	Modem with /ANS / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANS to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1}after sending /ANS (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANS (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • • Apply signal /ANS to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending /ANS (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending /ANS (from A)
Requirement	I) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call;D_{JB1} = D_{JB2} De-jitter buffer Fixed II) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay De-jitter buffer Fixed
Modem test sequences: **Calling tone conditions**	
Called station identification conditions	/ANS 2 100 Hz sine with 180° phase shift every 450 ms (ITU-T V.25) Duration 2.6 s to 4 s The amplitude of the tone is −31 dBm0

Test number:	**2.5.1 MODEM**
Transmission type:	Modem with /ANSam / −12 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1} after sending /ANSam (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANSam (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending /ANSam (from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending /ANSam (from A)
Requirement	I) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call;D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed II) a) First call; D_{JB1} = D_{JB2} Delay for De-jitter buffer Adaptive b) Second call; D_{JB1} = D_{JB2} for Delay for De-jitter buffer Fixed
Modem test sequences: **Calling tone conditions**	
Called station identification conditions	/ANSam 2 100 Hz sine with a 20% amplitude modulation by a 15 Hz sine and 180° phase shift every 450 ms (ITU-T V.25) The amplitude of the tone is −12 dBm0

Test number:	**2.5.2 MODEM**
Transmission type:	Modem with /ANSam / −31 dBm0
Measurement procedure	I) • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface A and determine the delay D_{JB1} • Apply signal C16 to Interface B and determine the delay D_{JB2} • Establishing a new call from A to B and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface B • Apply signal C16 to Interface A and determine the delay D_{JB1} after sending /ANSam (from B) • Apply signal C16 to Interface B and determine the delay D_{JB2} after sending /ANSam (from B) II) • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal C16 to Interface B and determine the delay D_{JB2} • Apply signal C16 to Interface A and determine the delay D_{JB1} • Establishing a new call from B to A and reset De-jitter buffer 1 and De-jitter buffer 2 • Apply signal /ANSam to Interface A • Apply signal C16 to Interface B and determine the delay D_{JB1} after sending /ANSam(from A) • Apply signal C16 to Interface A and determine the delay D_{JB2} after sending /ANSam(from A)
Requirement	I) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed II) a) First call; $D_{JB1} = D_{JB2}$ Delay for De-jitter buffer Adaptive b) Second call; $D_{JB1} = D_{JB2}$ for Delay for De-jitter buffer Fixed
Modem test sequences: Calling tone conditions	
Called station identification conditions	/ANSam 2 100 Hz sine with a 20% amplitude modulation by a 15 Hz sine and 180° phase shift every 450 ms) The amplitude of the tone is −31 dBm0

Appendix I

Features of ITU-T V.17 fax and V.34 fax

(This appendix does not form an integral part of this Recommendation.)

This appendix provides a summary of key features of ITU-T V.17- and V.34-based facsimile. The text and Figures I.1, I.2 and I.3 are based on the publication [b-FaxIntro].

Additionally, Figure I.1 provides a comparison of the supported bit rates amongst different fax modulation standards defined by ITU in the V-series Recommendations.

I.1 Features of V.17 fax (V.17 fax modem)

– Half-duplex mode of operation for fax applications.
– QAM is used for the channel with synchronous line transmission at 2 400 baud.
– Data signalling rates: 14 400 bit/s, 12 000 bit/s, 9 600 bit/s, 7 200 bit/s, 4 800 bit/s and 2 400 bit/s synchronous.
– Trellis coding at rates from 7 200 bit/s to 14 000 bit/s.
– Exchange of rate sequences is provided during start-up to establish the data-rate, coding, and any other special facilities.
– The frequency carrier operates at 1 800 Hz.
– Transmitted power levels conform to V.2.
– Modulation rate is 2 400 symbols/s.
– Supports V.24 interchange circuits.

I.2 V.34 high-speed fax

I.2.1 Features

– Fully compliant Group 3 Facsimile Support
– Full and half duplex modes
– Primary data channel supports 14 data rates in the range of 2 400 bit/s to 33 600 bit/s, in increments of 2 400 bit/s
– Control channel rates are 1 200 bit/s and 2 400 bit/s.

I.2.2 ITU-T V.34 fax [V.34 high-speed fax]

[The V.34 fax standard was derived from the V.34 data modem standard established by the International Telecommunications Union (ITU).] The V.34 data modem standard is a full-duplex implementation for sending and receiving data across telephone lines with a maximum data rate of 33.6 kbit/s. Certain elements of the V.34 data modem standard were eliminated for V.34 fax while new features, such as a control channel and mandatory error correction mode (ECM), were added to enable fast and reliable fax transmission.

Data rates supported (kbit/s)	ITU Recommendation		
	V.27 V.29	V.17	V.34
2.4	×		×
4.8	×		×
7.2	×	×	×
9.6	×	×	×
12		×	×
14.4		×	×
16.8			×
19.2			×
21.6			×
24			×
26.4			×
28.8			×
31.2			×
33.6			×

Figure I.1 – Comparison amongst fax modulation speeds

I.3 The ITU-T V.34 fax connection and session

In order to understand the benefits of the ITU-T V.34 fax standard, it is first necessary to understand how a fax transmission works. ITU-T V.34 session management and set up were designed with a similar mechanism to legacy handshaking procedures. The first step of a fax session is to establish a "handshake" between the sending and the receiving devices. During handshaking, the sending and receiving devices negotiate key parameters for how the fax call should be set up such as determining what is the highest transmission speed supported by both devices. The handshaking process itself is performed at 300 bit/s in legacy devices. In ITU-T V.34 fax capable devices, handshaking is performed at a much faster data rate of 1.2 kbit/s. The result is a handshake time that is reduced from approximately 16 s of legacy systems to 9 s for ITU-T V.34. See Figure I.2.

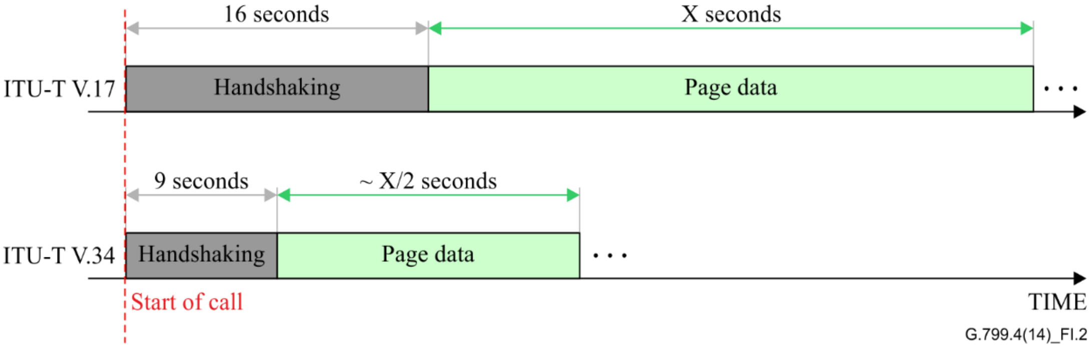


Figure I.2 – Time-wise comparison between ITU-T V.34 and ITU-T V.17 fax

After handshaking is complete, the next stage of a fax session is the transmission of the actual fax page data. The retraining and re-synchronization process takes place after each page is transmitted in legacy schemes, where capabilities such as supported modulation and transfer are renegotiated. In case of error in the transmission, entire pages may need to be retransmitted. This cycle of page data retrain and retransmit repeats until the fax call is completed, and account for significant inefficiency of legacy fax machines. ITU-T V.34 provides the most extensive range of supported data transmission rates, allowing it to optimize both speed and reliability over a wide range of line conditions. With ITU-T V.34, fax page data is transmitted at 33.6 kbit/s, twice the speed of ITU-T V.17. In addition, ITU-T V.34 uses ECM as a mandatory feature that handles page transmission error in a much more efficient way.

I.4 ECM as a mandatory feature

ECM is a mandatory feature for ITU-T V.34 fax as opposed to ITU-T V.17, where it is optional. The ECM protocol was designed to automatically detect and correct errors in the fax transmission process caused by factors such as telephone line noise. The page data to be transferred is divided into small blocks of data called octets. Once all octets are received, they are examined using check-sums. This is illustrated in Figure I.3.

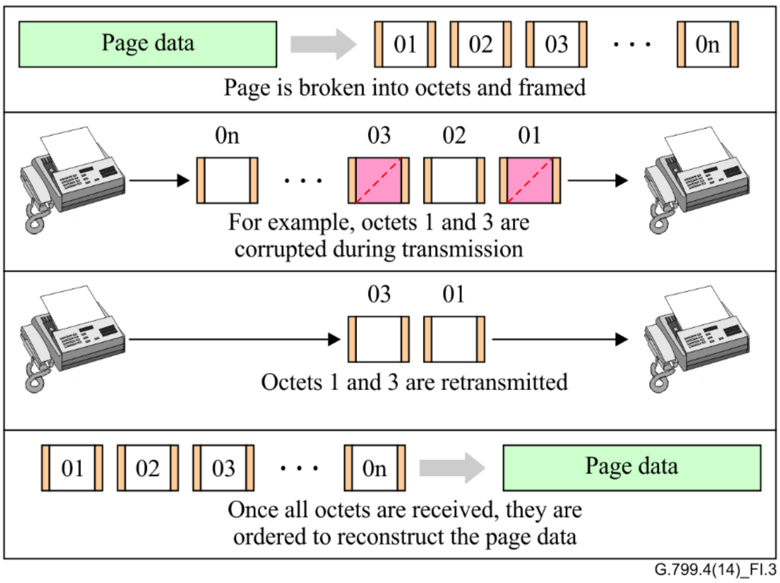


Figure I.3 – ECM-enabled fax transmission

If any errors in the checksums are detected, the receiving fax device signals the transmitting fax device to retransmit the octets that were received incorrectly. The transmitter then retransmits only the needed blocks rather than the whole page. Once all octets are received correctly, they are ordered and the page data is reconstructed by removing the octet frame and signalling flags. Generally, this results in a faster and more successful fax transmission than in a scenario where entire page data is retransmitted once or multiple times.

I.5 Suitable XoIP emulation services for V.34 fax

The two communicating ITU-T V.34-capable G3FEs may be interconnected via an IP network using gateways (see Figure 2 of [ITU-T F.185]). The two gateways (e.g., called emitting and receiving gateway in [ITU-T T.38]) would have to provide an adequate XoIP emulation service.

There are only two options in case of ITU-T V.34-capable G3FE, the use of either:

– ITU-T V.152 compliant VBDoIP emulation service (but not pseudo-VBDoIP) or

– ITU-T T.38v4 compliant FoIP emulation service (i.e., ITU-T T.38 (2010), but not pre-2010 versions of ITU-T T.38).

NOTE 1 – The FoIP gateways have to support ITU-T T.38/V.34 G3 endpoints (see clause 3.1.5). The support of ITU-T T.38/G3 endpoints (see clause 3.1.4) is not sufficient.

The establishment of correspondent media configurations in both gateways implies an explicit indication and negotiation phase in the network signalling plane (e.g., support of correspondent SDP attributes in case of SIP as call control protocol; NOTE 2).

NOTE 2 – Required at a very minimum,

a) in case of ITU-T V.152 VBDoIP: SDP attribute "a=gpmd:… vbd=yes";

b) in case of ITU-T T.38 FoIP: SDP attribute "a= T38ModemType".

Bibliography

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

[b-ITU-T G.799.3] Recommendation ITU-T G.799.3 (2012), *Signal processing functionality and performance of an IP-to-IP voice gateway optimized for the transport of voice and voiceband data*.

[b-ITU-T G.1020] Recommendation ITU-T G.1020 (2006), *Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks*.

[b-ITU-T I.231.1] Recommendation ITU-T I.231.1 (1988). *Circuit-mode bearer service categories: Circuit-mode 64 kbit/s unrestricted, 8 kHz structured bearer service.*

[b-ITU-T T.4] Recommendation ITU-T T.4 (2003), *Standardization of Group 3 facsimile terminals for document transmission*.

[b-ITU-T T.6] Recommendation ITU-T T.6 (1998), *Facsimile coding schemes and coding control functions for Group 4 facsimile apparatus.*

[b-ITU-T T.30] Recommendation ITU-T T.30, *Procedures for document facsimile transmission in the general switched telephone network*.

[b-ITU-T V.2] Recommendation ITU-T V.2 (1988), *Power levels for data transmission over telephone lines*.

[b-ITU-T V.18] Recommendation ITU-T V.18, *Procedures for starting sessions of data transmission over the public switched telephone network*.

[b-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*.

[b-ITU-T V.150.1] Recommendation ITU-T V.150.1, *Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs*.

[b-ETSI TR 183 072] ETSI TR 183 072 V3.1.1 (2010), *Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Emulation Services for PSTN Modem Calls*.

[b-FaxIntro] *Introduction to V.34 High-Speed Fax.* <http://www.gaoresearch.com/V34Fax/V34Fax.php>, latest access 24 February 2015.

[b-IETF RTC 4040] IETF RFC 4040 (2005). *Payload Format for a 64 kbit/s Transparent Call*.

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