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SERIES Q: SWITCHING AND SIGNALLING

Testing specifications – Testing specifications for next generation networks

Reference benchmarking, background traffic profiles and KPIs for VoIP and FoIP in fixed networks

Recommendation ITU-T Q.3933

ITU-T

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Recommendation ITU-T Q.3933

Reference benchmarking, background traffic profiles and KPIs for VoIP and FoIP in fixed networks

Summary

Recommendation ITU-T Q.3933 describes key performance indicators (KPIs) for voice over IP (VoIP) and fax over IP (FoIP) as well as framework requirements for reference benchmarking particularly with regard to background traffic.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T Q.3933	2015-06-13	11	11.1002/1000/12498

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FOREWORD

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The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

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

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

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1 Scope

The offer of new next generation network (NGN) services requires new key performance indicators (KPIs), quality of service (QoS), measurement, and benchmarking methods which are needed to ensure the quality of new services. To ensure the comparability of test results, reference benchmarking methods and background traffic load profiles are needed.

This Recommendation identifies and defines KPIs for voice and fax telephony services, as well as benchmarking methods for the spectrum of potential applications.

The scope of the defined testing procedures is the evaluation of the network access by VoIP and FoIP fixed-network services. The measurements are conducted stationary between a subscriber access-point to a measurement point emulating an idealized termination point in the core network. All access technologies offered by the operator under test are considered. In this context, the measurements and KPIs determinations are performed by analysing signals which are accessible on the network.

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 E.453] Recommendation ITU-T E.453 (1994), *Facsimile image quality as corrupted by transmission-induced scan line errors.*

[ITU-T E.458] Recommendation ITU-T E.458 (1996), *Figure of merit for facsimile transmission performance*.

[ITU-T E.800] Recommendation ITU-T E.800 (2008), *Definitions of terms related to quality of service*.

[ITU-T G.131] Recommendation ITU-T G.131 (2003), *Talker echo and its control*.

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

[ITU-T H.248.1] Recommendation ITU-T H.248.1 V3 (2013), *Gateway control protocol: Version 3*.

[ITU-T O.41] Recommendation ITU-T O.41/P.53 (1994), *Psophometer for use on telephone-type circuits*.

[ITU-T P.56] Recommendation ITU-T P.56 (2011), *Objective measurement of active speech level*.

[ITU-T P.340] Recommendation ITU-T P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free terminals*.

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

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[ITU-T P.863.1] Recommendation ITU-T P.863.1 (2014), *Application guide for Recommendation ITU-T P.863*.

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[ITU-T T.30] Recommendation ITU-T T.30 (2005), *Procedures for document facsimile transmission in the general switched telephone network*.

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[ITU-T V.17] Recommendation ITU-T V.17 (1991), *A 2-wire modem for facsimile applications with rates up to 14 400 bit/s.*

[ITU-T V.34] Recommendation ITU-T V.34 (1998), *A modem operating at data signalling rates of up to 33 600 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits*.

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

[ETSI EG 202 057-2] ETSI EG 202 057-2 V1.3.2 (2011), *Speech and multimedia Transmission Quality (STQ);User related QoS parameter definitions and measurements;Part 2: Voice telephony, Group 3 fax, modem data services and SMS.*

[ETSI EG 202 425] ETSI EG 202 425 V1.1.1 (2007), *Speech Processing, Transmission and Quality Aspects (STQ); Definition and implementation of VoIP reference point*.

[ETSI EN 300 175-8] ETSI EN 300 175-8 V2.5.1 (2013), *Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission*.

[ETSI ES 202 737] ETSI ES 202 737 V1.4.1 (2015), *Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user*.

[ETSI ES 202 739] ETSI ES 202 739 V1.4.1 (2015), *Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user*.

[ETSI ES 202 765-2] ETSI ES 202 765-2 V1.2.1 (2014), *Speech and multimedia Transmission Quality (STQ); QoS and network performance metrics and measurement methods; Part 2 : Transmission Quality Indicator combining Voice Quality Metrics*.

[ETSI ES 203 021-3] ETSI ES 203 021-3 V2.1.2 (2006), *Access and Terminals (AT); Harmonized basic attachment requirements for Terminals for connection to analogue interfaces of the Telephone Networks; Update of the technical contents of TBR 021, EN 301 437, TBR 15, TBR 17; Part 3: Basic Interworking with the Public Telephone Networks.*

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[ETSI TBR 003 ed.1] ETSI TBR 003 ed.1 (1995), *Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN basic access*.

[ETSI TBR 004 ed.1] ETSI TBR 004 ed.1 (1995), *Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN primary rate access*.

[ETSI TBR 21] ETSI TBR 21 (1996-06), *Terminal Equipment (TE); Attachment requirements for pan-European approval for connection to the analogue Public Switched Telephone Networks (PSTNs) of TE (excluding TE supporting the voice telephony service) in which network addressing, if provided, is by means of Dual Tone Multi Frequency (DTMF) signalling..*

[ETSI TR 103 138] ETSI TR 103 138, *Speech and multimedia Transmission Quality (STQ); Speech samples and their use for QoS testing.*

[ETSI TS 101 563] ETSI TS 101 563 V1.3.1 (2014), *Speech and multimedia Transmission Quality (STQ); IMS/PES/VoLTE exchange performance requirements*.

[ETSI TS 102 250-2] ETSI TS 102 250-2 V2.3.1 (2014), *Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in GSM and 3G networks; Part 2: Definition of Quality of Service parameters and their computation*.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following term defined elsewhere:

3.1.1 benchmark [ITU-T E.800]: Evaluation of performance value/s of a parameter or set of parameters for the purpose of establishing value/s as the norm against which future performance achievements may be compared or assessed.

3.2 Terms defined in this Recommendation

None.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

AGCF Access Gateway Control Function

AS Application Server

CFR Call Failure Rate

CSS Composite Source Signal

DL Downlink

DSS1	Digital Subscriber Signalling System No. 1
FoIP	Fax over IP
FOM	Figure of Merit
IAD	Integrated Access Device
IP	Internet Protocol
IPTV	Internet Protocol Television
IMS	Internet Multimedia Subsystem
ISDN	Integrated Services Digital Network
KPI	Key Performance Indicator
LTE	Long Term Evolution
MOS	Mean Opinion Score
MOS-LQO	MOS Listening Quality Objective
MMTel	Multimedia Telephony service
PES	PSTN Emulation Subsystem
PSTN	Public Switched Telephone Network
QoS	Quality of Service
P-CSCF	Proxy Call Session Control Function
S-CSCF	Service Call Session Control Function
SIP	Session Initiation Protocol
SIPUA	SIP User Agent
SNR	Speech Signal level/Noise level
SSL	Secure Sockets Layer
SWB	Super Wideband
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UAS	User Agent Server
UE	User Equipment
UL	Uplink
VGW	Voice Gateway
VoIP	Voice over IP

5 Conventions

None.

6 Management summary

The spectrum of potential applications of a benchmarking platform requires measurements including, but not limited to, the following: Analogue (a/b), integrated services digital network (ISDN), VoIP (including session initiation protocol (SIP) trunking), and high-speed internet.

The performance data which are collected will be relevant for the real-world environment encompassing a mix of technologies. The scope of the defined testing procedures is the evaluation of the network access by VoIP and FoIP fix-network services. The measurements are conducted stationary between a subscriber access-point and a measurement point emulating an idealized termination point in the core network.

6.1 Scope of functionality

A benchmarking platform can be distributed across a larger region or an entire country. In this case, several server systems should also be part of the set-up, including: A business intelligence platform, data warehouse, management system, and a system for evaluating media (e.g., video, audio, and voice) quality. See Figure 1.

The measurement systems at the user premises are connected electrically to ISDN ports via a voice gateway (VGW), integrated access device (IAD), or directly to a CPE or Ethernet port (e.g., multimedia telephony service (MMTel) fixed access).

The test system (quality of service (QoS) control and data server) is connected through ISDN connections (via internet multimedia subsystem (IMS), PSTN emulation subsystem (PES) with access gateway control function (AGCF) (or public switched telephone network (PSTN), or ISDN access) or IMS PES with VGW) or MMTel (IMS) fixed access lines for voice quality measurements.

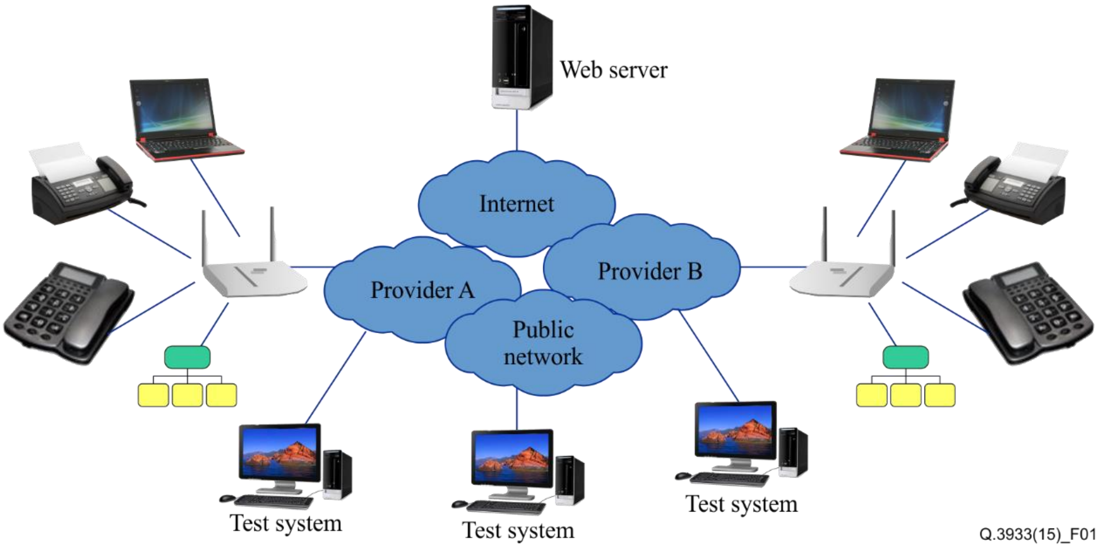

Figure 1 – Set-up of the benchmarking platform

7 Technical concept

7.1 Voice over internet protocol (IP)

The conduction of voice quality measurements is following the descriptions that can be found in [ETSI EG 202 057-2], [ITU-T Q.543], [ETSI TS 101 563] and clauses 6.6.3.1 and 6.6.3.2 of [ETSI TS 102 250-2].

The access points of the test equipment which are used for inserting or retrieving the signals needed for determining the speech quality parameters shall conform to the reference characteristics as laid down in the following relevant standards:

– [ETSI EG 202 425] for VoIP access;

– [ETSI TBR 21], [ETSI ES 203 021-3] for analogue access;

– TBR 3 [ETSI TBR 003 ed.1] for ISDN BRI access;
– TBR 4 [ETSI TBR 004 ed.1] for ISDN PRI access;
– [ETSI EN 300 175-8], Digital enhanced cordless telecommunications (DECT); common interface (CI); Part 8: Speech and audio coding and transmission.

The properties of the test equipment shall be known and the values measured for each parameter, and shall be corrected accordingly by the impairments introduced by the test equipment. Especially, any delay introduced by the test equipment shall be known, and the measurement results shall be corrected by the delay introduced by the test equipment.

The simultaneous transmission of voice and data through uploads, downloads, or internet protocol television (IPTV) is an additional user-related scenario. For this reason, voice quality measurements have been included where in parallel to the voice connection active upload and download of data is simulated. This provides information about any potential prioritization of voice data when the entire bandwidth is being utilized.

The KPIs listed in Table 1 are recorded as part of the voice quality measurements.

Table 1 – Overview of KPIs for voice quality measurements

1.	Call set-up delay, see [ITU-T Q.543] and session initiation call set-up delay, see [ETSI TS 101 563]
2.	Call set-up time (post dialling delay), see [ETSI ES 202 765-2]
3.	Premature release probability (call failure rate), see clause 7.4
4.	Telephony cut-off call ratio [%] (call drop rate), see clause 7.5
5.	Unsuccessful call ratio, see clause 7.6
6.	Media establishment delay, see clause 7.7
7.	Level of active speech signal, see clause 7.8
8.	Noise level, see clause 7.9
9.	Signal to noise ratio, see clause 7.10
10.	Speech signal attenuation, see clause 7.11
11.	Talker echo delay, see clause 7.12
12.	Double talk, see clause 7.1.3
13	Interrupted voice transmission see clause 7.14
14.	Listening speech quality, see clause 7.15
15.	Listening speech quality stability, see clause 7.16
16.	End-to-end audio delay, see clause 7.17
17.	End-to-end audio delay variation, see clause 7.18
18.	Frequency response, see clause 7.19
19.	Fax transmission ITU-T T.30 (fax, bit rate ≤ 14.4 kbit/s and Fax, bit rate ≥ 14.4 kbit/s), see clause 7.20
20.	Early media see clause 7.21
21.	Jitter buffer and IP periodization response time, see clause 7.22

7.2 Call set-up delay and session initiation call set-up delay

The testing methodology for the call set-up delay is described in [TS 101 563].

Call set-up delay is defined as the interval from the instant when the signalling information required for outgoing circuit selection is received from the incoming signalling system until the instant when the corresponding signalling information is passed to the outgoing signalling system.

For SIP (e.g., SIP trunking, IMS) session initiation call set-up delay is defined as the interval from the instant when the INVITE signalling information is received from the calling user on the originating Gm interface until the instant when the corresponding INVITE signalling information is passed on from the terminating Gm interface to the called user.

Figure 2 depicts some of the call set-up measurement options between AGCF/VGW and the Gm interface.

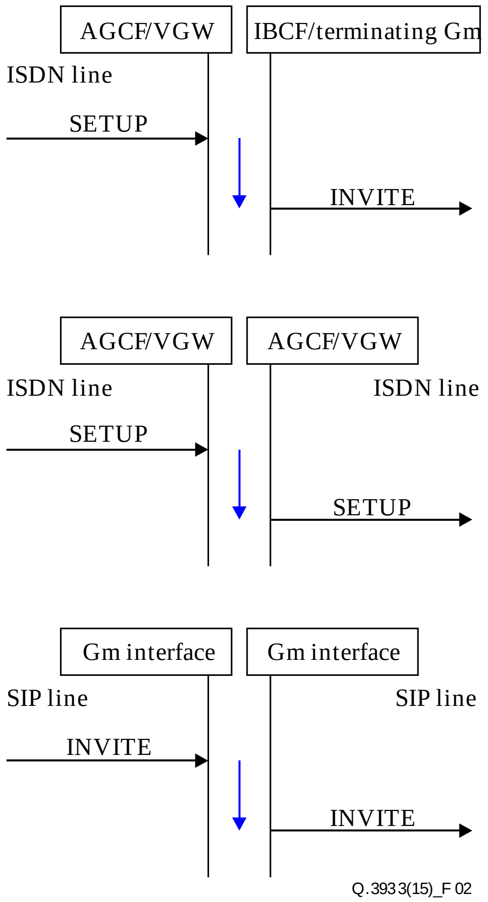

Figure 2 – Call set-up delay and session initiation call set-up delay: En-bloc sending is used

Table 2 gives an overview of the call set-up delay configuration options.

Table 2 – Call set-up delay configurations

	From	To
Call set-up delay and session initiation call set-up delay	MMTel (IMS) fixed access	MMTel (IMS) fixed access
	MMTel (IMS) fixed access	IMS PES with AGW (PSTN or ISDN access)
	MMTel (IMS) fixed access	IMS PES with VGW
	IMS PES with AGW (PSTN or ISDN access)	MMTel (IMS) fixed access
	IMS PES with AGW (PSTN or ISDN access)	IMS PES with AGW(PSTN or ISDN access)
	IMS PES with AGW (PSTN or ISDN access)	IMS PES with VGW
	IMS PES with VGW	IMS PES with VGW
	IMS PES with VGW	IMS PES with AGW (PSTN or ISDN access)
	IMS PES with VGW	IMS PES with VGW
NOTE – The call set-up delay values are specified in [ETSI TS 101 563]		

Figure 3 illustrates the session processing model used by the AGCF and VGW functional entities.

An AGCF is modelled as comprising ITU-T H.248 media gateway controller (MGC), feature manager (FM), and SIP user agent (SIP UA) functionality.

An AGCF interfaces to a media gateway (MG) and also to the service call session control function (S-CSCF) (via P1 and Mw reference points respectively).

A functional modelling of the VGW contains an entity similar to ITU-T H.248 media gateway controller, a feature manager, a SIP UA, and MGW functionality. The VGW interfaces to the proxy call session control function (P-CSCF) using the Gm reference point.

The SIP UA functionality provides the interface to the other components of the IMS-based architecture. It is involved in registration and session processing as well as in event subscription/notification procedures with application servers (ASs).

The MGC functionality enables the session processing functionality to interface with existing line signalling such as analogue signalling or digital subscriber signalling system No. 1 (DSS1).

Session and registration processing in the AGCF or VGW involves two halves: ITU-T H.248 based MGC processing and SIP.

User agent (UA) processing (see Figure 3). MGC processing focuses on the interactions with the media gateway functions, while SIP UA processing focuses on the interactions with the IMS components. The feature manager (FM) coordinates the two processing activities.

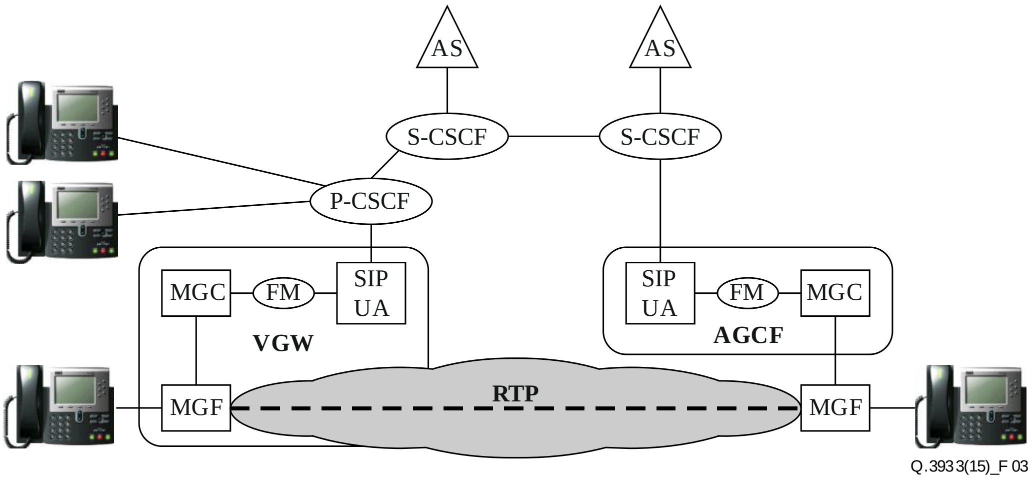

Figure 3 – AGCF/VGW session processing models

7.3 Call set-up time (post dialling delay)

Call set-up time: The period starting when the address information required for setting up a call is received by the network (e.g., recognized on the calling user's access line) and finishing when the called party busy tone, ringing tone, or answer signal is received by the calling party (e.g., recognized on the calling user's access line) (see [ETSI ETR 138]).

To determine the call set-up time in an ISDN implementation, the time in seconds from sending the DSS1 SETUP signal through the "A" side (calling number + "Sending complete" information), until the receipt of the DSS1 CONNECT signal on the "A" side is measured, or the time in seconds from the sending of the DSS1 SETUP signal through the "A" side (calling number + "Sending complete" information), until the receipt of the DSS1 ALERTING signal is measured on the "A" side is measured. In Figures 4 and 5 below, this time is indicated by the green arrow.

For analogue subscriber lines the post dialling delay shall be used. It is the time interval between the end of dialling by the caller and the caller’s reception back of the appropriate ringing tone or recorded announcement.

To determine the call set-up time in a VoIP implementation, the time in seconds from the sending of the INVITE signal through the "A" side until the receipt of the 200 OK signal on the "A" side is measured, or the time in seconds from the sending of the INVITE signal through the "A" side until the receipt of the 180 Ringing signal on the "A" side is recorded. In Figures 6 and 7, this time is indicated by the grey arrow.

Table 3 gives an overview of the call set-up time configurations options.

Table 3 – Call set-up time configurations

	From	To
Call set-up time	MMTel (IMS) fixed access	MMTel (IMS) fixed access
	MMTel (IMS) fixed access	IMS PES with AGW (PSTN or ISDN access)
	MMTel (IMS) fixed access	IMS PES with VGW
	IMS PES with AGW (PSTN or ISDN access)	MMTel (IMS) fixed access
	IMS PES with AGW (PSTN or ISDN access)	IMS PES with AGW(PSTN or ISDN access)
	IMS PES with AGW (PSTN or ISDN access)	IMS PES with VGW
	IMS PES with VGW	IMS PES with VGW
	IMS PES with VGW	IMS PES with AGW (PSTN or ISDN access)
	IMS PES with VGW	IMS PES with VGW

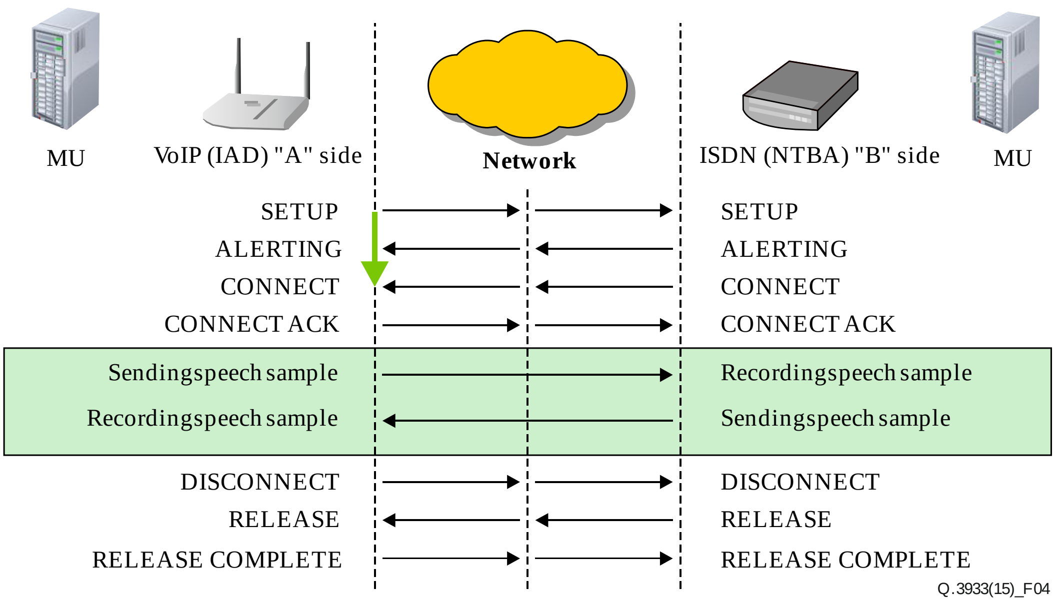

Figure 4 – Measurement of the call set-up duration, option A with CONNECT

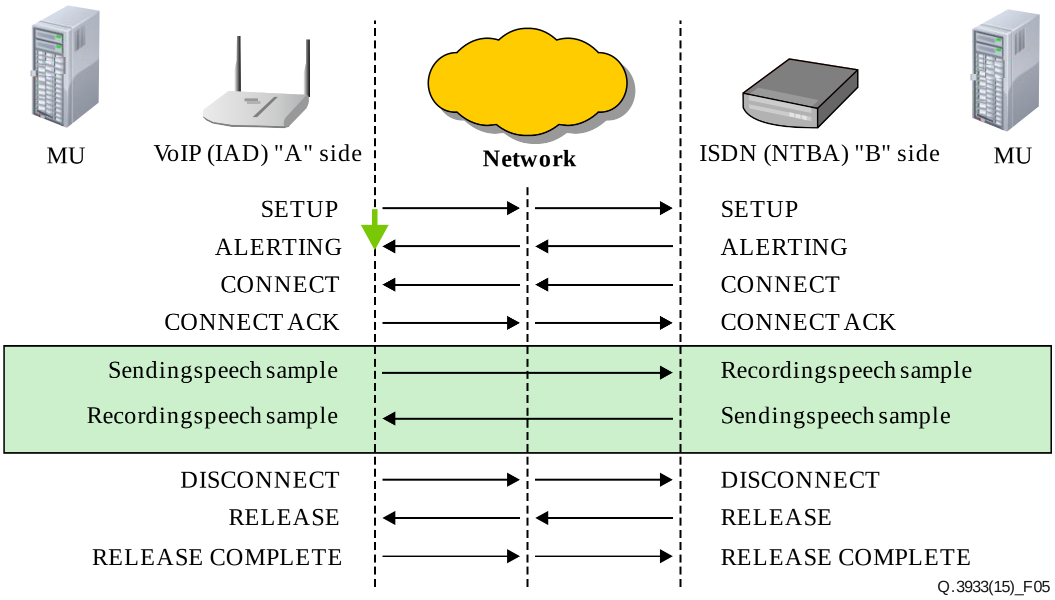

Figure 5 – Measurement of the call set-up duration, option B with ALERTING

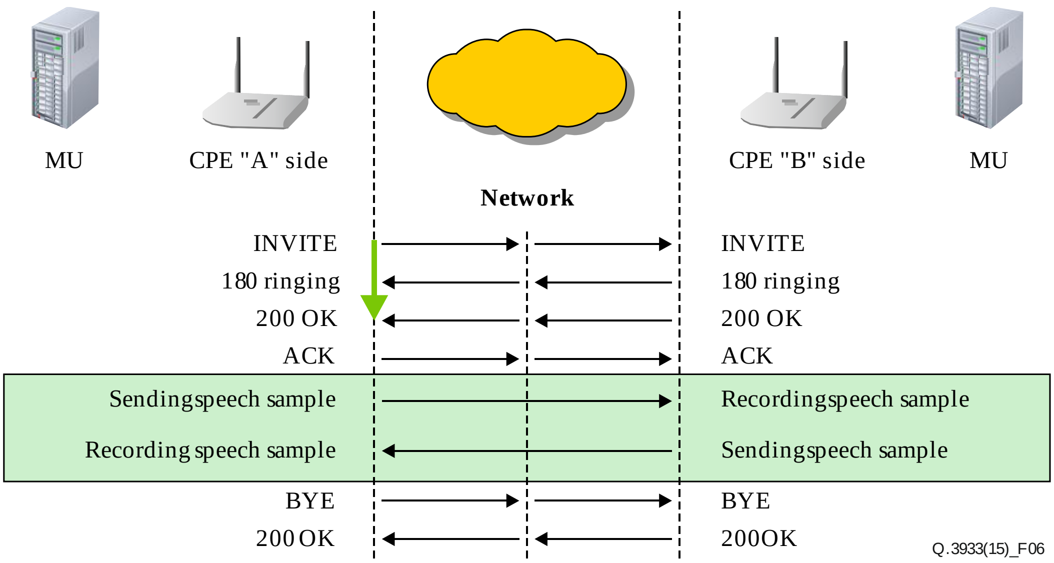

Figure 6 – VoIP measurement of the call set-up duration, option A with 200 OK

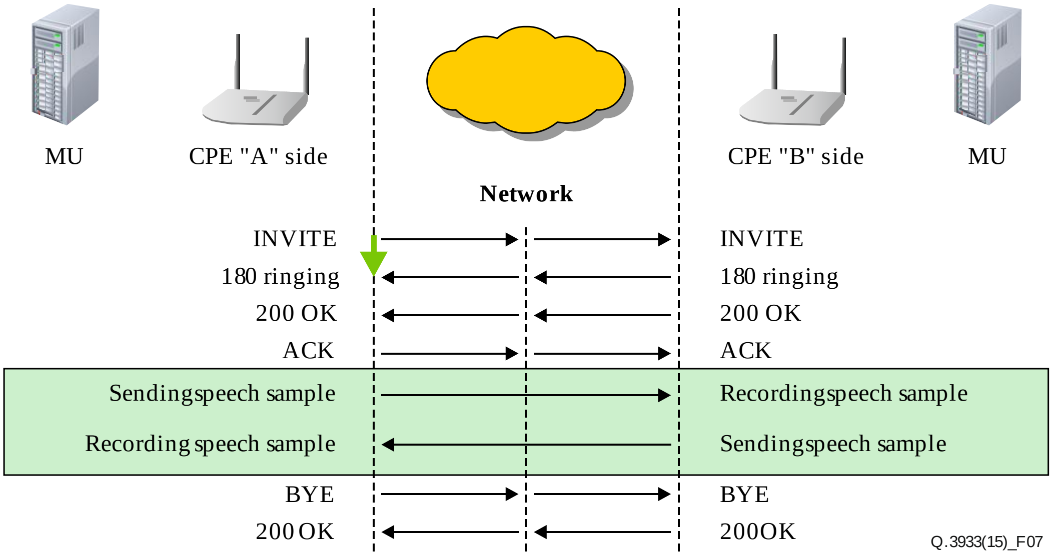

Figure 7 – VoIP measurement of the call set-up duration, option B with 180 Ringing

7.4 Premature release probability (call failure rate (CFR))

Premature release probability (CFR) is based on the measurement of the number of released communications in comparison with the number of established communications. Released communications are defined as communications released before voluntary action from one of the ends of the transmission.

7.5 Telephony cut-off call ratio [%] (call drop rate)

The cut-off call ratio (call drop rate) is the percentage of the number of calls that are dropped after connection to the system or network has been established. See [ETSI TS 102 250-2] for the formula.

In an ISDN implementation, a call is completely established with the connect message [ETSI TS 102 250-2] and is considered dropped if the call is not ended intentionally.

In a SIP implementation, a call is completely established with the arrival of the INVITE 200 OK on the caller side and is considered dropped if the call is not terminated intentionally.

Figures 8 and 9 below show determinations of the call drop ratio.

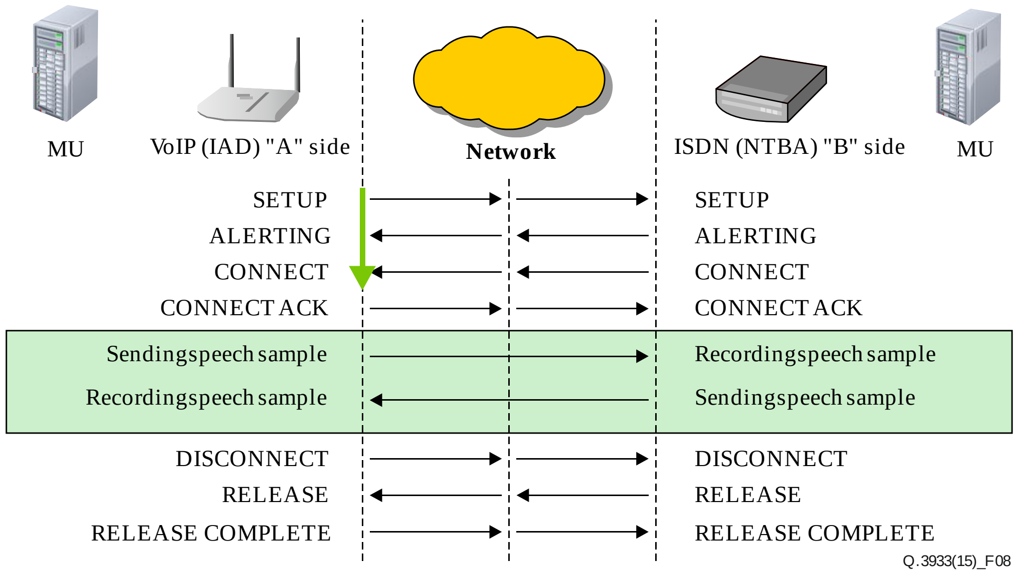

Figure 8 – Determination of the call drop ratio

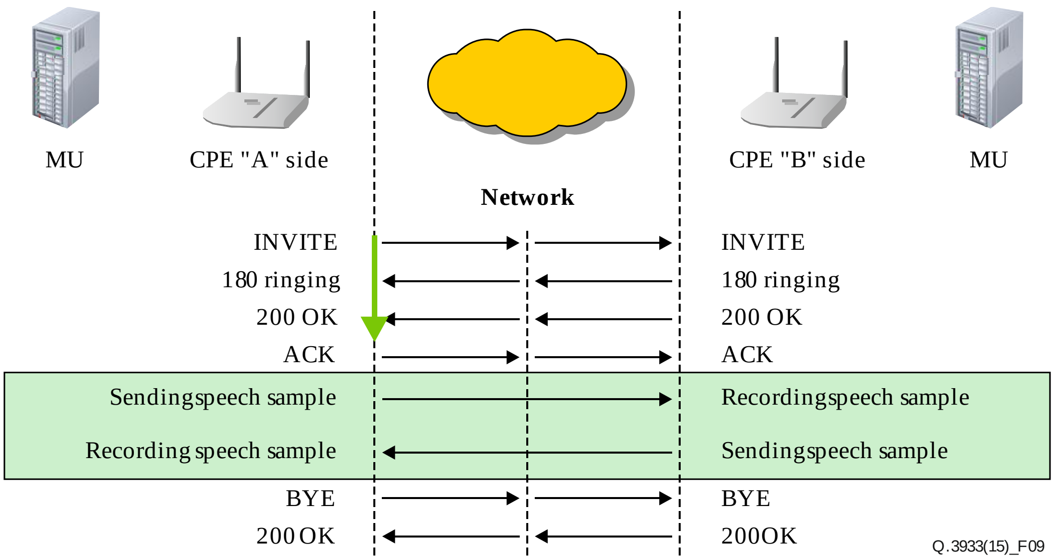

Figure 9 – VoIP – determination of the call drop ratio

7.6 Unsuccessful call ratio [ETSI ETR 138]

Unsuccessful call ratio is defined as the ratio of unsuccessful calls to the total number of call attempts in a specified time period.

An unsuccessful call is a call attempt to a valid number, properly dialled, where neither the called party busy tone, nor ringing tone, nor answer signal, is recognized on the access line of the calling user within 30 seconds from the instant when the address information required for setting up a call is received by the network.

7.7 Media establishment delay

The media establishment delay determines one of the two accesses of the communication between the off hook of the called and the beginning of voice signal receive. The detailed testing method is described in [ETSI ES 202 765-2].

7.8 Level of active speech signal in receive direction

A typical method for the measurement of this parameter, based on a sample by sample approach and a moving threshold between noise and speech, is given in [ITU-T P.56].

7.9 Noise level in receive direction

Level of noise determined in the receive direction in the non-speech segments of a speech sample. For the actual measurement, the noise in between speech signals (idle noise) is analysed. The analysis window length needs to be adapted accordingly.

The noise level is measured in the frequency range from 100 Hz to 4 kHz in narrowband and from 100 Hz to 8 kHz in wideband. The analysis window is applied directly to the end of a speech signal until the start of the following speech signal. The average time is determined by the length of this segment.

In narrowband, the noise level is measured in dBm0p (*psophometric* weighting, see [ITU-T O.41]). In wideband, the noise level is determined in dBm0 (A).

7.10 Signal-to-noise ratio in receive direction

The signal-to-noise ratio in the receive direction is defined as the difference between the active speech level and the level of noise in the receive direction (speech signal level/noise level (SNR)).

The signal level is the average level of the complete speech signal. The signal level is measured using a speech level voltmeter according to [ITU-T P.56]. This level is the speech signal level.

The noise level in receive direction is determined as described in clause 7.9.

The weighted noise signal level is referenced to the speech signal level.

7.11 Speech signal attenuation (or gain)

The speech signal attenuation is the difference between the active speech level at the receiving and at the sending point.

7.12 Talker echo delay

In telecommunications, the term talker echo describes delayed and undesired feedback from the send signal into the receive path. The so-called echo source is the reflection point between send and receive directions. Talker echo delay is the round-trip delay of the echo path. The impact of user perception of talker echo in conjunction with delay is explained in [ITU-T G.131]. The detailed test description can be found in [ETSI ES 202 765-2].

In general, the test of talker echo delay can be based on cross correlation between the speech signal inserted, and the echo signal received. The measurement is corrected by delays which are caused by the test equipment. The maximum of the cross-correlation function is used for the determination. However, it shall be noted that such measurements can only be made in case the echo signal is sufficiently high to allow a reliable calculation of the cross correlation.

NOTE – In case the talker echo is received at a very low level, the echo loss might be artificially decreased in order to allow for the calculation of talker echo delay.

7.13 Double talk performance

This parameter looks into the situation when the talk spurts of both partners of a conversation overlap for a period of time. Degradations due to bad double talk performance can be perceived as very annoying because this impairment has a potential to frequently interrupt the flow of the conversation.

During double talk, the speech is mainly determined by two parameters: Impairment caused by echo during double talk and level variation between single and double talk (attenuation range).

In order to allow for sufficient quality under double talk conditions the talker echo loudness rating (TELR) should be high and the attenuation inserted should be as low as possible. Connections which do not allow double talk in any case should provide a good echo attenuation which is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see [ITU-T P.340] and [ITU-T P.502]):

– attenuation range in send direction during double talk $A_{H,S,dt}$.;

– attenuation range in receive direction during double talk $A_{H,R,dt}$.;

– echo attenuation during double talk.

The categorization of a connection is based on the three categories defined in the following clauses and this categorization is given by the "worst" of the three parameters, e.g., if $A_{H,S,dt}$ provides 2a, $A_{H,R,dt}$ 2b and echo loss 1, the categorization of the terminal is 2b.

Test Signal

The test signal to determine the attenuation range during double talk is the double talk speech sequence as defined in clause 7.3.5 of [ITU-T P.501] as shown in Figure 10. The competing speaker is always inserted as the double talk sequence $s_{dt}(t)$ either in send or receive and is used for analysis.

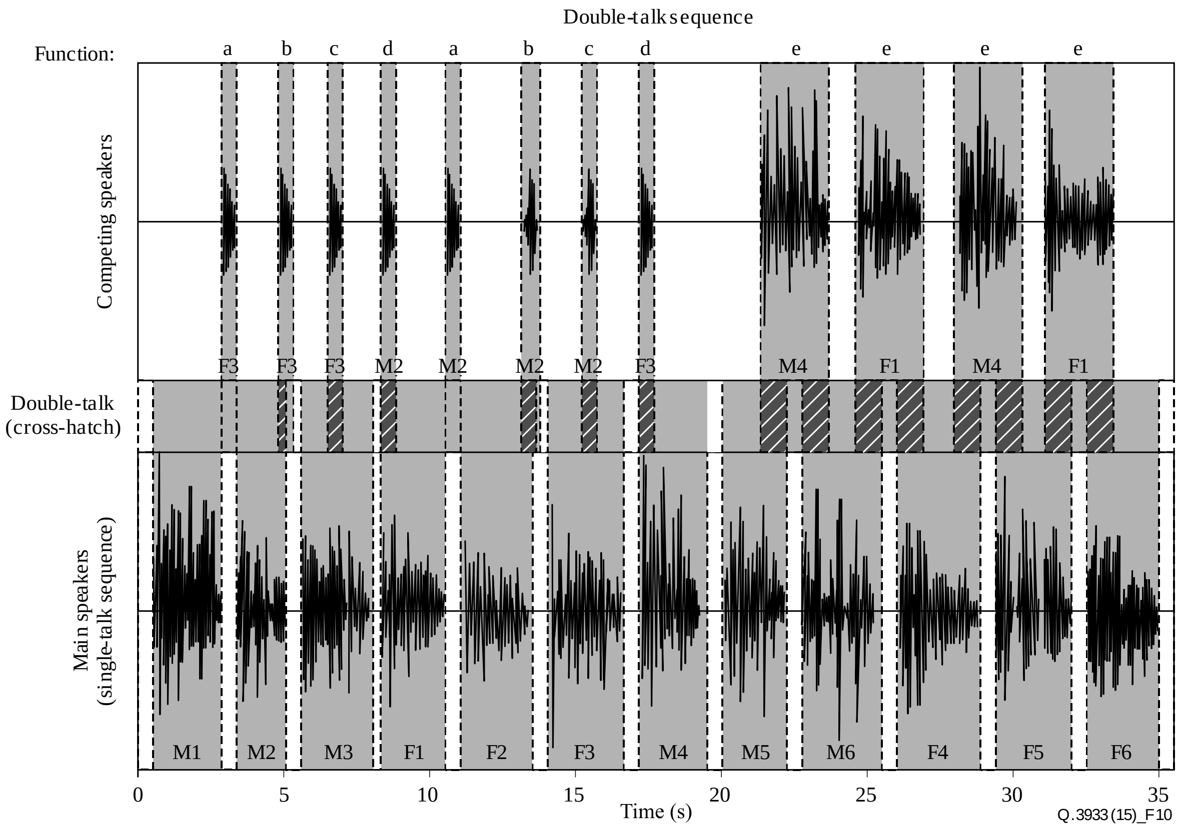

Figure 10 – Double talk test sequence with overlapping speech sequences in send and receive direction

Measurement method

The attenuation range during double talk is determined as described in Appendix III of [ITU-T P.502]. The double talk performance is analysed for all sequences of the competing speaker. The requirement has to be met for each word and sentence produced by the competing speaker.

7.14 Interrupted voice transmission

A call is defined as interrupted if the duration of the interruption of the voice transmission is > 1 s and the call connection is maintained.

7.15 Listening speech quality

7.15.1 General aspects of listening speech quality

Listening speech quality represents the intrinsic quality of speech signal as perceived by the user at the receiving end. This indicator takes into account the impairments introduced by the transmission system. The mean opinion score-listening quality objective (MOS-LQO) is obtained by comparing speech samples:

- the original undistorted reference speech signal;
- the degraded signal received at the local end, where the measurement is applied.

[ITU-T P.863] recommends two samples from each of two male and two female speakers, i.e., eight sentence pairs. Some applications may only permit shorter test durations. Typically, test sentence material in subjective tests has a 0.5 second silence lead in, two sentences, and then 0.5 s of silence at the end of the signal. Further information can be found in [ETSI TR 103 138] and [ITU-T P.863.1].

To ensure comparable voice quality results, it shall be ensured that the test equipment uses the codec described in the first line of the m line in the SDP part which is the preconfigured codec by the network operator.

7.15.2 General aspects of voice channel test calls

For the all voice channel tests, an aligned structure of the voice call shall be used. In this call, sentence pairs (male/female) fulfilling the requirements of [ITU-T P.863.1] shall be transmitted from A to B and from B to A. Speech files specially tested for use with [ITU-T P.863] are published in Annex C of [ITU-T P.501], where samples in different languages are covered.

In principle, all voice channel tests consist of three parts:

– Channel convergence quality test;

– Listening speech quality test;

– DTMF test.

Which parts are actually used and how they are structured is defined for the individual test cases in the sections below.

The channel convergence quality test starts with a listening speech quality test from B to A after the connection is established. This initial test provides information about the listening quality during convergence of the channel.

For the analysis of the initial listening speech quality during convergence of the channel, the method according to [ITU-T P.863] in super wideband (SWB) mode based on only two sentences (one female voice and one male voice) is used. For this purpose, a male voice (e.g., "four hours of steady work faced us") and a female voice (e.g., "the hogs were fed chopped corn and garbage") can be selected from the test sentences provided in Annex C of [ITU-T P.501].

After convergence of the channel, the regular listening speech quality test starts using [ITU-T P.863] in SWB mode based on eight sentences (two male and two female voices, two sentences each).

Usually, the listening speech quality tests should start 10 s after the connection is established. This 10 s pause is recommended for converging the speech processing components and building up the IP-buffer at the receiving side and can be used for the channel convergence quality test as described above. It is assumed that the convergence has finished after 10 s. In the event of a proven shorter convergence, the pause can be shorter.

In case the channel can be assumed as converged from the beginning, and/or the separation of the channel convergence quality is not of interest, the listening quality test can start at any time after the connection is established.

Within the listening speech quality test, for example the following English samples can be selected from the test sentences provided in Annex C of [ITU-T P.501]:

– *Female 1:*
  - These days a chicken leg is a rare dish.
  - The hogs were fed with chopped corn and garbage.
– *Female 2:*
  - Rice is often served in round bowls.
  - A large size in stockings is hard to sell.
– *Male 1:*
  - The juice of lemons makes fine punch.
  - Four hours of steady work faced us.
– *Male 2:*
  - The birch canoe slid on smooth planks.
  - Glue the sheet to the dark blue background.

If a global application is of interest, optionally the male and female test sentences of other languages provided in Annex C of [ITU-T P.501] can be used.

After conducting all evaluations, the derived MOSs for each sample in the listening test are averaged over all received and scored samples separately for each direction A-B and B-A.

DTMF test: DTMF tones are often used for remote controlling equipment and must also be tested in an established voice channel for correct transmission. It is recommended to test DTMF before or after the listening speech quality test but in each case after the channel has converged. The DTMF test should consist of DTMF tones (100 ms signal, 100 ms pause) and shall contain the tones 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, *, #.

Technical comments

– If the interrupted voice transmission time is > 1 s and the call connection is maintained, the call is rated as interrupted (see clause 7.14).
– If all 8 sentences (4 samples) are sent within one file, the score calculation according to [ITU-T P.863] shall be performed separately for each sample (2 sentences per sample).
– In case the sampling frequency at the input measuring interface is 8 kHz, as is usual for ISDN and narrowband applications, the input speech signal used shall be band limited at 3 800 kHz (see [ETSI TR 103 138]).
– When the sampling frequency at the input measuring interface is 16 kHz as required for wideband telephony, the input speech signal used shall be band limited between 100 Hz and

7 600 kHz with a band pass filter providing a minimum of 24 dB/oct. filter roll off, when feeding into the receive direction (see [ETSI TR 103 138]).

– The input test signal levels are referred to the average level of the (band limited in the receive direction) test signal, averaged over the complete test sequence unless specified otherwise. It is recommended to adjust the active speech level to −26dB OVL as specified in [ETSI TR 103 138].

7.15.3 Connections without parallel data transfer

7.15.3.1 Connections with one voice channel

For the single voice channel test, a test call consisting of the three following parts should be used:

– Channel convergence quality test;

– Listening speech quality test;

– DTMF test.

Figure 13 depicts the detailed description of the single voice channel test. The general technical aspects are described in clause 7.15.2.

Table 4 gives an overview of the connection options without parallel data transfer.

Table 4 – Connection options without parallel data transfer

	From	**To**
Connections without parallel data transfer	MMTel (IMS) fixed access	MMTel (IMS) fixed access
	MMTel (IMS) fixed access	IMS PES with AGW (PSTN or ISDN access)
	MMTel (IMS) fixed access	IMS PES with VGW
	IMS PES with AGW (PSTN or ISDN Access)	MMTel (IMS) fixed access
	IMS PES with AGW (PSTN or ISDN Access)	IMS PES with AGW(PSTN or ISDN access)
	IMS PES with AGW (PSTN or ISDN Access)	IMS PES with VGW
	IMS PES with VGW	IMS PES with VGW
	IMS PES with VGW	IMS PES with AGW (PSTN or ISDN access)
	IMS PES with VGW	IMS PES with VGW

The derived MOSs in the listening test are averaged over all received and scored samples separately for each direction A-B and B-A.

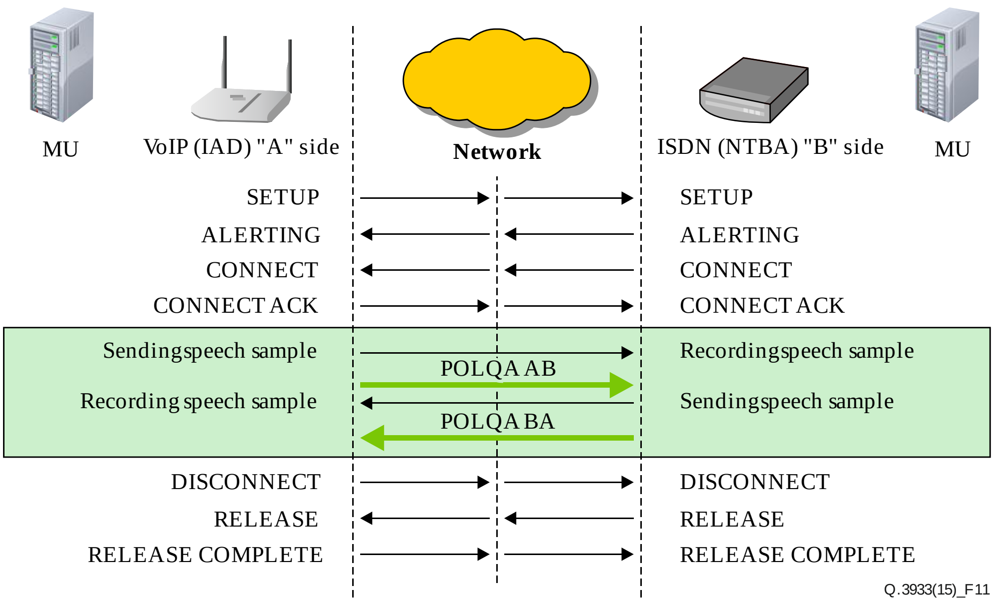

Figure 11 – Measurement of voice quality

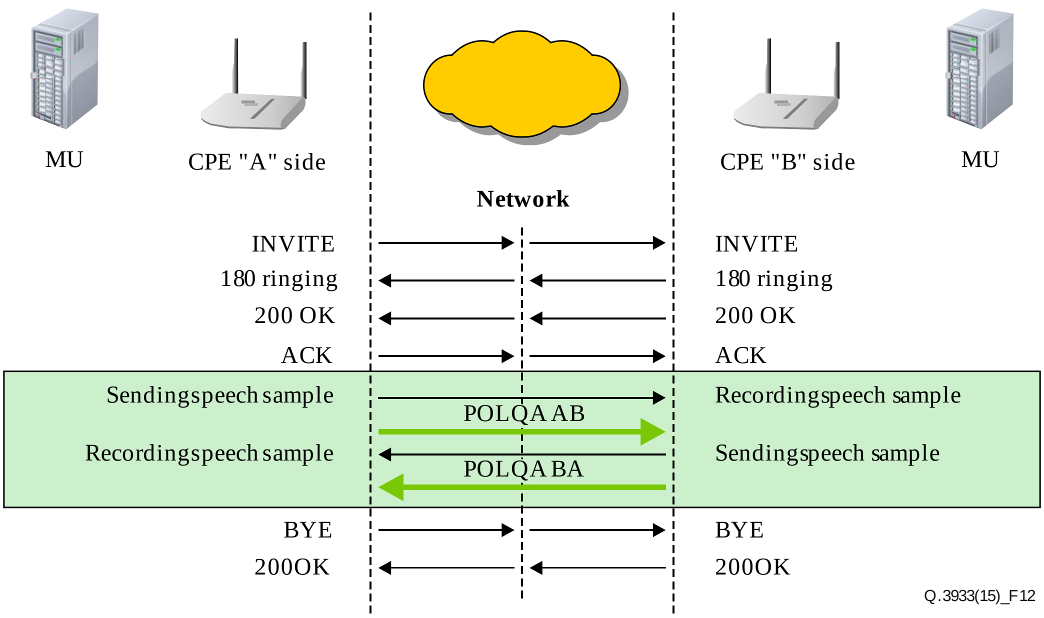

Figure 12 – VoIP measurement of voice quality, MMTel fixed to MMTel fixed

Relative Time	Test equipment A		NETWORK		Test equipment B
	CALL A to B				
T0 – 2	SETUP / INVITE	→		→	SETUP / INVITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK/ ACK	→		→	CONNECT ACK/ ACK
	Start convergence quality test				
T0	Start Audio Receive BA_1 (female & male)	←		←	Start Audio Send BA_1 (female & male)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
	Stop convergence quality test				
	Listening speech quality test				
T0 + 10 s	Start Audio Send AB_1 (female 1)	→		→	Start Audio Receive AB_1 (female 1)
	End Audio Send AB_1 (female 1)	→		→	End Audio Receive AB_1 (female 1)
	Start Audio Send AB_2 (female 2)	→		→	Start Audio Receive AB_2 (female 2)
	End Audio Send AB_2 (female 2)	→		→	End Audio Receive AB_2 (female 2)
	Start Audio Send AB_3 (male 1)	→		→	Start Audio Receive AB_3 (male 1)
	End Audio Send AB_3 (male 1)	→		→	End Audio Receive AB_3 (male 1)
	Start Audio Send AB_4 (male 2)	→		→	Start Audio Receive AB_4 (male 2)
	End Audio Send AB_4 (male 2)	→		→	End Audio Receive AB_4 (male 2)
	Pause				
	Start Audio Receive BA_1 (female 1)	←		←	Start Audio Send BA_1 (female 1)
	End Audio Receive BA_1 (female 1)	←		←	End Audio Send BA_1 (female 1)
	Start Audio Receive BA_2 (female 2)	←		←	Start Audio Send BA_2 (female 2)
	End Audio Receive BA_2 (female 2)	←		←	End Audio Send BA_2 (female 2)
	Start Audio Receive BA_3 (male 1)	←		←	Start Audio Send BA_3 (male 1)
	End Audio Receive BA_3 (male 1)	←		←	End Audio Send BA_3 (male 1)
	Start Audio Receive BA_4 (male 2)	←		←	Start Audio Send BA_4 (mal 2)
	End Audio Receive BA_4 (male 2)	←		←	End Audio Send BA_4 (male 2)
	Pause				
	Start DTMF Send AB_1	→		→	Start DTMF Receive AB_1
	End DTMF Send AB_1	→		→	End DTMF Receive AB_1
	Start DTMF Receive BA_1	←		←	Start DTMF Send BA_1
	End DTMF Receive BA_1	←		←	End DTMF Send BA_1
	DISCONNECT / BYE	→		→	DISCONNECT /BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE

Figure 13 – Single voice channel test

7.15.3.2 Multiple voice channel access

In the case of multiple voice channel access or SIP trunking, during the complete testing phase the first call (single voice channel test) from user A to user B is active (see Figure 14).

The single voice channel test consists of the following parts:

– Convergence quality test;

– Listening speech quality test;

– DTMF test.

The result of each part shall be listed in the test report.

Figure 15 depicts the detailed description of the single voice channel test for the multiple voice channel access.

The four speech files should be repeated while the call is active. For n channels n −1 cycles (duration of one cycle is approximately 80 seconds) are recommended (e.g., for 4 channels the duration is approximately 360 seconds).

Parallel to the single voice channel test additional calls should be established (multiple voice channel test). The multiple voice channel test consists of the following parts:

– Convergence quality test;

– Listening speech quality test.

Figure 16 depicts the detailed description of the multiple voice channel access. The general aspects for the multiple voice channel tests are given in clause 7.15.2.

Figure 14 depicts an example of the multiple voice channel access test for five channels.

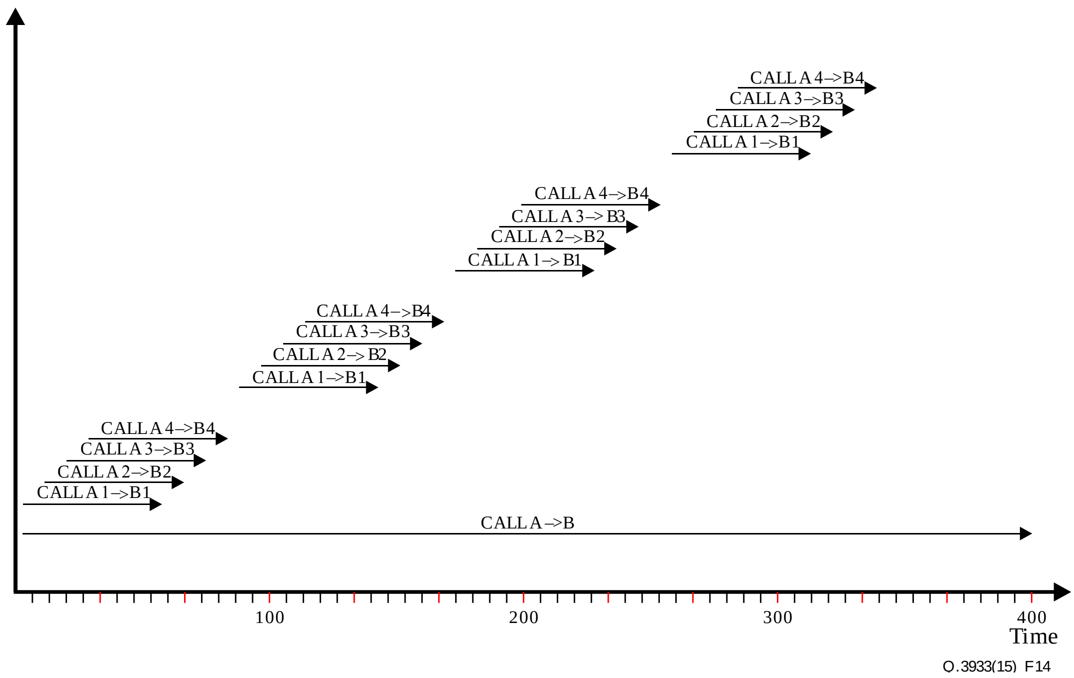

Figure 14 – Example of multiple voice channel access test

Relative Time	Test equipment A		NETWORK		Test equipment B
T0 – 2	SETUP / INVITE	→		→	SETUP / INVITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK/ ACK	→		→	CONNECT ACK/ ACK
	Start convergence quality test				
T0	Start Audio Receive BA_1 (female & male)	←		←	Start Audio Send BA_1 (male & female)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
	Stop convergence quality test				
	Listening speech quality test				
T0 + 10 s	Start Audio Send AB_1 (female 1)	→		→	Start Audio Receive AB_1 (female 1)
	End Audio Send AB_1(female 1)	→		→	End Audio Receive AB_1 (female 1)
	Start Audio Send AB_2 (female 2)	→		→	Start Audio Receive AB_2 (female 2)
	End Audio Send AB_2 (female 2)	→		→	End Audio Receive AB_2 (female 2)
	Start Audio Send AB_3 (male 1)	→		→	Start Audio Receive AB_3 (male 1)
	End Audio Send AB_3 (male 1)	→		→	End Audio Receive AB_3 (male 1)
	Start Audio Send AB_4 (male 2)	→		→	Start Audio Receive AB_4 (male 2)
	End Audio Send AB_4 (male 2)	→		→	End Audio Receive AB_4 (male 2)
	Pause				
	Start Audio Receive BA_1 (female 1)	←		←	Start Audio Send BA_1 (female 1)
	End Audio Receive BA_1 (female 1)	←		←	End Audio Send BA_1 (female 1)
	Start Audio Receive BA_2 (female 2)	←		←	Start Audio Send BA_2 (female 2)
	End Audio Receive BA_2 (female 2)	←		←	End Audio Send BA_2 (female 2)
	Start Audio Receive BA_3 (male 1)	←		←	Start Audio Send BA_3 (male 1)
	End Audio Receive BA_3 (male 1)	←		←	End Audio Send BA_3 (male 1)
	Start Audio Receive BA_4 (male 2)	←		←	Start Audio Send BA_4 (mal 2)
	End Audio Receive BA_4 (male 2)	←		←	End Audio Send BA_4 (male 2)
	Pause				
	Start DTMF Send AB_1	→		→	Start DTMF Receive AB_1
	End DTMF Send AB_1	→		→	End DTMF Receive AB_1
	Start DTMF Receive BA_1	←		←	Start DTMF Send BA_1
	End DTMF Receive BA_1	←		←	End DTMF Send BA_1
The four speech files are repeated during the call is active					
	Start DTMF Send AB_1	→		→	Start DTMF Receive AB_1
	End DTMF Send AB_1	→		→	End DTMF Receive AB_1
	Start DTMF Receive BA_1	←		←	Start DTMF Send BA_1
	End DTMF Receive BA_1	←		←	End DTMF Send BA_1
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE

Figure 15 – Single voice channel test for multiple voice channel access

Relative Time	Test equipment A		NETWORK		Test equipment B
T0 – 2	SETUP / INVITE	→		→	SETUP / INVITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK/ ACK	→		→	CONNECT ACK/ ACK
	Start convergence quality test				
T0	Start Audio Receive BA_1 (female & male)	←		←	Start Audio Send BA_1 (male & female)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
	Stop convergence quality test				
	Listening speech quality test				
T0 + 10 s	Start Audio Send AB_1 (female 1)	→		→	Start Audio Receive AB_1 (female 1)
	End Audio Send AB_1 (female 1)	→		→	End Audio Receive AB_1 (female 1)
	Start Audio Send AB_2 (female 2)	→		→	Start Audio Receive AB_2 female 2)
	End Audio Send AB_2 (female 2)	→		→	End Audio Receive AB_2 (female 2)
	Start Audio Send AB_3 (male 1)	→		→	Start Audio Receive AB_3 (male 1)
	End Audio Send AB_3 (male 1)	→		→	End Audio Receive AB_3 (male 1)
	Start Audio Send AB_4 (male 2)	→		→	Start Audio Receive AB_4 male 2)
	End Audio Send AB_4 (male 2)	→		→	End Audio Receive AB_4 (male 2)
	Pause				
	Start Audio Receive BA_1 (female 1)	←		←	Start Audio Send BA_1 (female 1)
	End Audio Receive BA_1 (female 1)	←		←	End Audio Send BA_1 (female 1)
	Start Audio Receive BA_2 (female 2)	←		←	Start Audio Send BA_2 (female 2)
	End Audio Receive BA_2 (female 2)	←		←	End Audio Send BA_2 (female 2)
	Start Audio Receive BA_3 (male 1)	←		←	Start Audio Send BA_3 (male 1)
	End Audio Receive BA_3 (male 1)	←		←	End Audio Send BA_3 (male 1)
	Start Audio Receive BA_4 (male 2)	←		←	Start Audio Send BA_4 (mal 2)
	End Audio Receive BA_4 (male 2)	←		←	End Audio Send BA_4 (male 2)
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE

Figure 16 – Multiple voice channel test

7.15.4 Connections with parallel data transfer

7.15.4.1 Quality measurement of one voice channel and parallel data transfer

In the case when the access link is used for voice and data application, the voice quality measurement sequence with parallel upload/download shall be used. Table 5 gives an overview of the connection options with parallel data transfer, Figures 17 and 18 depict the measurement of voice quality with parallel data load.

Table 5 – Connection options with parallel data transfer

	From		To	
	Voice	**Data**	**Voice**	**Data**
Connections with parallel data transfer	MMTel (IMS) fixed access	User data server or user data application	MMTel (IMS) fixed access	Webserver
	MMTel (IMS) fixed access	User data server or user data application	IMS PES with AGW (PSTN or ISDN Access)	Webserver
	MMTel (IMS) fixed access	User data server or user data application	IMS PES with VGW	Webserver
	IMS PES with VGW	User data server or user data application	IMS PES with VGW	Webserver
	IMS PES with VGW	User data server or user data application	IMS PES with AGW (PSTN or ISDN Access)	Webserver
	IMS PES with VGW	User data server or user data application	MMTel (IMS) fixed access	Webserver

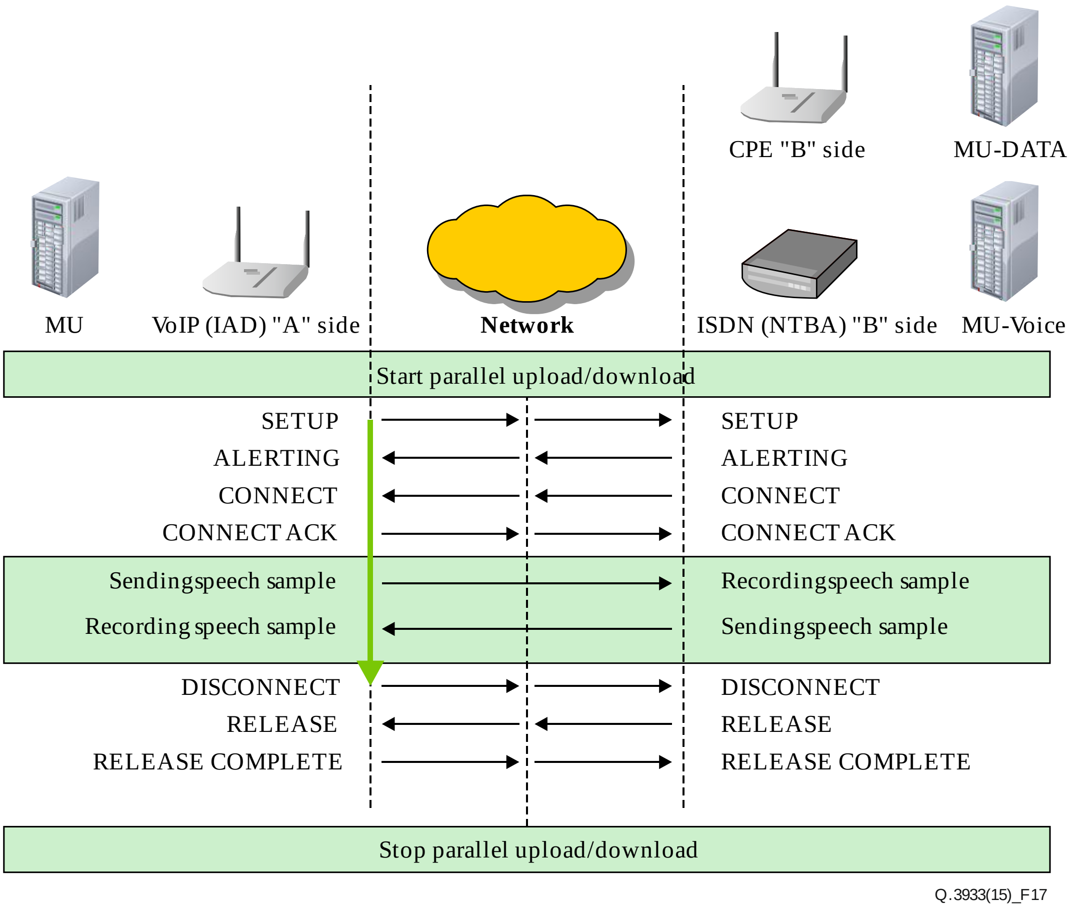

Figure 17 – Measurement of voice quality with parallel data load

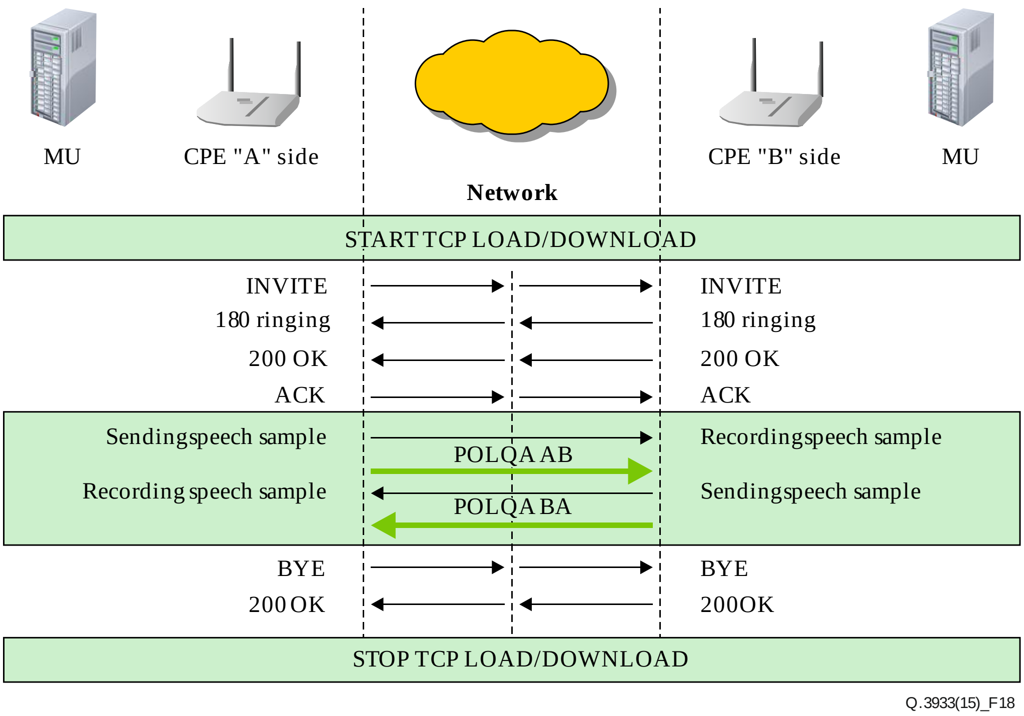

Figure 18 – VoIP measurement of voice quality with parallel data load

During parallel download and upload the data size should be between 80% and 100% of the nominal data capacity of the link or respectively the maximal data capacity which can be provided during the voice transmission.

For data transfer the following constants are used:

– n number of parallel transmission control protocol (TCP) connections for one direction ($1 \leq n \leq 10$); default: $n = 3$

– In case of use of fixed-size chunks, the initial size of data block sent during the test is s = 4.096 Bytes.

– Duration of tests, approximately t = 80 s.

Several parallel data streams are initiated with a number of n parallel TCP connections with an upload and download of data files from the data-reference system. The upload shall start before the call set-up starts, and the download before the voice quality measurement starts.

After time t, each TCP connection shall be reset.

As an option, the server can continuously send data streams consisting of fixed-size chunks of size s (randomly generated data with high entropy). The data should be transferred between client and server over the TCP port 443 using transport layer security (TLS) or secure sockets layer (SSL) in order to avoid interference with firewalls as much as possible. The ports for communication and data transfers between the different servers themselves shall be configurable.

Parallel to the data transmission, the single voice channel test should be established. For this, the upload and download is started at least six seconds before the speech quality measurement and will also continue for at least six seconds after the end of the speech quality measurements in order to ensure full utilization of the bandwidth during the measurement phase. The structure is shown in Figures 19 and 20.

The single voice channel test consists of the following parts as described and explained in clause 7.15.2:

– Convergence quality test;

– Listening speech quality test;

– DTMF test.

Figure 13 depicts the detailed description of the single voice channel test.

Relative Time	Test equipment A		NETWORK		Test equipment B data reference system
Start download and upload procedure					
T0 – 10 s	Start TCP upload connection 1	→		→	
	Start TCP upload connection 2	→		→	
	Start TCP upload connection 3	→		→	
	Single voice channel test				
T0 – 5 s	SETUP / INVITE	→		→	SETUP / INVITE
	Start TCP download connection 1	←		←	
	Start TCP download connection 2	←		←	
	Start TCP download connection 3	←		←	
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK / ACK	→		→	CONNECT ACK / ACK
	Start convergence quality test				
T0	Start Audio Receive BA_1 (female & male)	←		←	Start Audio Send BA_1 (male & female)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
	Stop convergence quality test				
	Listening speech quality test				
T0 + 10 s	Start Audio Send AB_1 (female 1)	→		→	Start Audio Receive AB_1 (female 1)
	………………continued ……………………				
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE
	Stop download and upload procedure				

Figure 19 – Detailed download and upload procedure for automatically controlled test sequence

Relative Time	Test equipment A		NETWORK		Test equipment B data reference system
Start download and upload procedure					
T0 – 10 s	Start TCP upload connection 1	→		→	
	Start TCP upload connection 2	→		→	
	Start TCP upload connection 3	→		→	
	Start TCP download connection 1	←		←	
	Start TCP download connection 2	←		←	
	Start TCP download connection 3	←		←	
	Single voice channel test				
T0 – 5 s	SETUP / INVITE	→		→	SETUP / INVITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK / ACK	→		→	CONNECT ACK / ACK
	Start convergence quality test				
T0	Start Audio Receive BA_1 (female & male)	←		←	Start Audio Send BA_1 (male & female)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
	Stop convergence quality test				
	Listening speech quality test				
T0 + 10 s	Start Audio Send AB_1 (female 1)	→		→	Start Audio Receive AB_1 (female 1)
	……………….continued ……………………				
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE
	Stop download and upload procedure				

Figure 20 – Detailed download and upload procedure for a manually controlled test sequence

The download and upload (see Figures 19 and 20) is based on several parallel data streams initiated with TCP with data files from the data-reference system. This ensures that the maximum data transfer rate during the entire measurement period can be achieved. In the determination of the time window, the effects of TCP congestion control were (overload control) taken into account. Initiating several parallel data streams at the same time reduces the effect of the TCP/IP configuration of the measurement unit to the measurement.

The download and upload procedure shall be repeated while the voice call measurements are active.

7.15.4.2 Parallel quality measurement of one voice channel and data transmission speed

During the parallel download and upload, the data size should be between 80% and 100% of the nominal data capacity of the link or, respectively, the maximal data capacity which can be provided during the voice transmission.

For the data transfer, the following constants are used:

– n number of parallel TCP connections for one direction ($1 \leq n \leq 10$); default: $n = 3$.

– In case of use of fixed-size chunks, the initial size of data block sent during the test is s = 4.096 Bytes.

– t duration of tests, approximately t = 80 s.

Several parallel data streams are initiated with a number of n parallel TCP connections with an upload and download of data files from the data-reference system. The upload shall start before the call set-up starts, the download before the voice quality measurement starts. After time t, each TCP connection shall be reset.

As an option, the server can continuously send data streams consisting of fixed-size chunks of size s (randomly generated data with high entropy). The data should be transferred between client and server over the TCP port 443 using TLS or SSL in order to avoid interference with firewalls as much as possible. The ports for communication and data transfers between the different servers themselves shall be configurable.

For each TCP connection k, $1 \leq k \leq n$, the client records the relative time "t" and the amount of data received from time from 0 to t.

After completion of all tests, the client sends the results and data collected to the data server. Both datasets are then compared by the data server to check the quality and integrity of the result. All tests, successful or unsuccessful, are stored by the data server.

The values presented which are measured every 500 ms shall include the minimum, the average, and the maximum values.

Table 5 gives an overview of the connection options with parallel data transfer.

Parallel to the data transmission, the single voice channel test should be established.

The single voice channel test consists of the following parts as described in clause 7.15.2:

– Convergence quality test;

– Listening speech quality test;

– DTMF test.

The result of each part shall be listed in the test report.

Figure 13 depicts the detailed description of the single voice channel test.

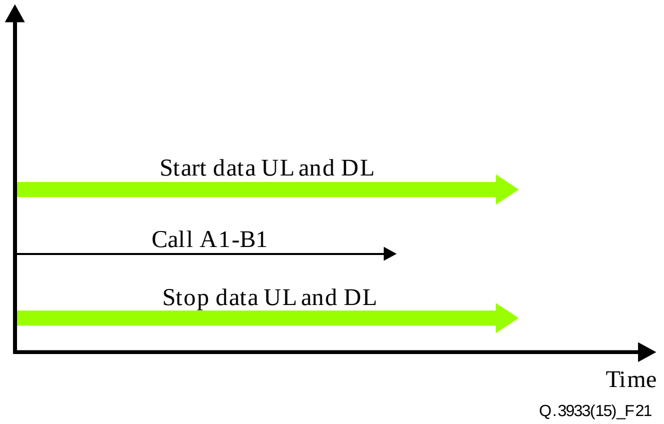

Figure 21 – Parallel measurement of the quality of one voice channel and data transmission speed

Relative time	Test equipment A		NETWORK		Test equipment B data reference system
T0 – 12 s	Ping	→		→	
Start download and upload procedure					
T0 – 10 s	Start TCP upload connection 1	→		→	
	Start TCP upload connection 2	→		→	
	Start TCP upload connection 3	→		→	
	Start TCP download connection 1	←		←	
	Start TCP download connection 2	←		←	
	Start TCP download connection 3	←		←	
The calculation of the throughput values for up/down stream starts					
T0 – 5	SETUP / INITE	→		→	SETUP / INITE
	ALERTING / 180 Ringing	←		←	ALERTING /180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK / ACK	→		→	CONNECT ACK / ACK
Start convergence quality test					
T0	Start Audio Receive BA_1 (male & female)	←		←	Start Audio Send BA_1 (male & female)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
Stop convergence quality test					
Listening speech quality test					
T0 + 10 s	Start Audio Send AB_1 (female 1)	→		→	Start Audio Receive AB_1 (female 1)
	End Audio Send AB_1 (female 1)	→		→	End Audio Receive AB_1 (female 1)
	Start Audio Send AB_2 (female 2)	→		→	Start Audio Receive AB_2 (female 2)
	End Audio Send AB_2 (female 2)	→		→	End Audio Receive AB_2 (female 2)
	Start Audio Send AB_3 (male 1)	→		→	Start Audio Receive AB_3 (male 1)
	End Audio Send AB_3 (male 1)	→		→	End Audio Receive AB_3 (male 1)
	Start Audio Send AB_4 (male 2)	→		→	Start Audio Receive AB_4 (male 2)
	End Audio Send AB_4 (male 2)	→		→	End Audio Receive AB_4 (male 2)
1 s	Pause				
	Start Audio Receive BA_1 (female 1)	←		←	Start Audio Send BA_1 (female 1)

Relative time	Test equipment A		NETWORK		Test equipment B data reference system
	End Audio Receive BA_1 (female 1)	←		←	End Audio Send BA_1 (female 1)
	Start Audio Receive BA_2 (female 2)	←		←	Start Audio Send BA_2 (female 2)
	End Audio Receive BA_2 (female 2)	←		←	End Audio Send BA_2 (female 2)
	Start Audio Receive BA_3 (male 1)	←		←	Start Audio Send BA_3 (male 1)
	End Audio Receive BA_3 (male 1)	←		←	End Audio Send BA_3 (male 1)
	Start Audio Receive BA_4 (male 2)	←		←	Start Audio Send BA_4 (mal 2)
	End Audio Receive BA_4 (male 2)	←		←	End Audio Send BA_4 (male 2)
	Pause				
	Start DTMF Send AB_1	←		←	Start DTMF Receive AB_1
	End DTMF Send AB_1	←		←	End DTMF Receive AB_1
	Start DTMF Receive BA_1	←		←	Start DTMF Send BA_1
	End DTMF Receive BA_1	←		←	End DTMF Send BA_1
	DISCONNECT / BYE	←		←	DISCONNECT / BYE
	RELEASE / 200 OK	→		→	RELEASE /200 OK
	RELEASE COMPLETE	←		←	RELEASE COMPLETE
The data transmission of data streams is stopped					
The calculation of the throughput values for up/down stream					

Figure 22 – Detailed listening speech quality, DTMF procedure and uplink (UL)/downlink (DL) procedure

7.15.4.3 Quality measurement of multiple voice channels and data transfer

In the case of multiple voice channel access or SIP trunking, during the complete testing phase the first call from user A to user B is active. During the parallel download and upload the data size should be between 80% and 100% of the nominal data capacity of the link or, respectively, the maximal data capacity which can be provided during the voice transmission.

For the data transfer the following constants are used:

– *n* number of parallel TCP connections for one direction ($1 \le n \le 10$); default: $n = 3$.

– In case of use of fixed-size chunks, the initial size of data block sent during the test is s = 4.096 bytes.

– t duration of tests: The download and upload procedure shall be repeated while the voice call measurement procedure is active.

Several parallel data streams are initiated with a number of *n* parallel TCP connections with an upload and download of data files from the data-reference system. The upload shall start before the call set-up starts, and the download before the voice quality measurement starts. After time t, each TCP connection shall be reset.

As an option, the server can continuously send data streams consisting of fixed-size chunks of size s (randomly generated data with high entropy). The data should be transferred between client and server over the TCP port 443 using TLS or SSL in order to avoid interference with firewalls as much as possible. The ports for communication and data transfers between the different servers themselves shall be configurable.

Parallel to the data transmission the single voice channel and multiple voice channel test should be established.

The single voice channel test consists of the following parts:

– Convergence quality test;
– Listening speech quality test;
– DTMF test.

Figure 15 depicts the detailed description of the single voice channel test for the multiple voice channel access, where a part of transmitted speech samples are repeated multiple times.

The test sequence shall be performed in both directions (UNI_A to UNI_B and UNI_B to UNI_A).

The four speech files should be repeated while the call is active. For n channels n −1 cycles (approximately 80 seconds) are recommended (e.g., for 4 channels the duration is approximately 360 seconds), see also Figure 15.

Parallel to the single voice channel test, additional calls should be established (multiple voice channel test, as described in clause 7.15.2). The call establishment time between the multiple voice channels should be 1 second (the load is 0.5 calls per second).

The multiple voice channel test consists of the following parts:

– Convergence quality test;
– Listening speech quality test.

Figure 16 depicts the detailed description of the multiple voice channel access.

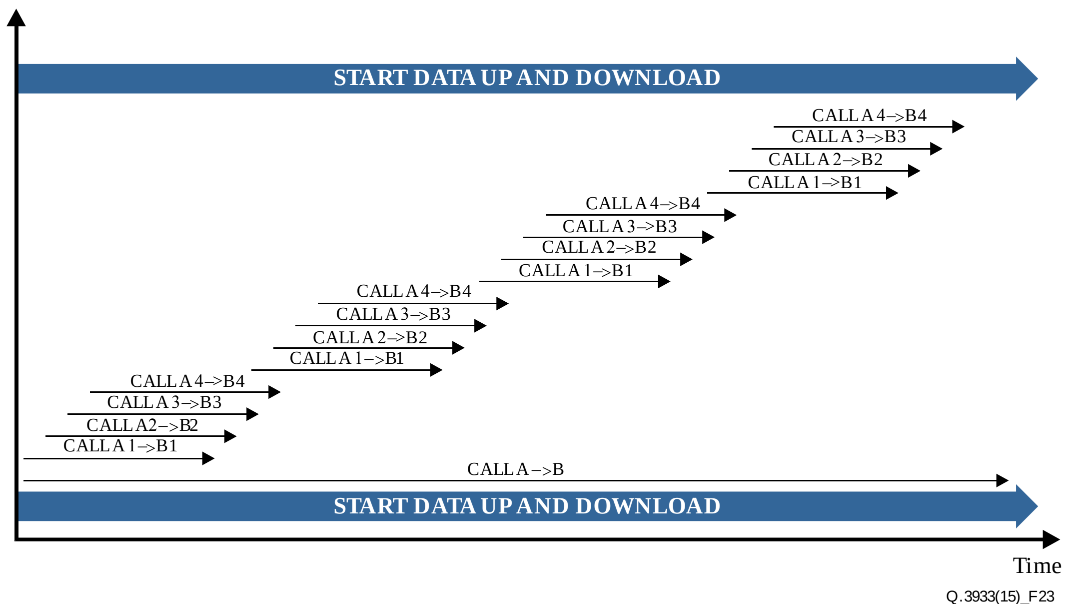

Figure 23 – Multiple voice channel access test

7.15.4.4 Parallel quality measurement of multiple voice channels and data transmission speed

In the case of multiple voice channel access or SIP trunking, during the complete testing phase the first call from user A to user B is active. During the parallel download and upload the data size should be between 80% and 100% of the nominal data capacity of the link or, respectively, the maximal data capacity which can be provided during the voice transmission. For the data transfer the following constants are used:

– n number of parallel TCP connections for one direction ($1 \leq n \leq 10$); default: $n = 3$.
– In case of use of fixed-size chunks, the initial size of data block sent during the test is s = 4.096 Bytes.
– t duration of tests: The download and upload procedure shall be repeated while the voice call measurement procedure is active.

Several parallel data streams are initiated with a number of n parallel TCP connections with an upload and download of data files from the data-reference system. The upload shall start before the call set-up starts, and the download before the voice quality measurement starts. After time t, each TCP connection shall be reset.

As an option, the server can continuously send data streams consisting of fixed-size chunks of size s (randomly generated data with high entropy). The data should be transferred between client and server over the TCP port 443 using TLS or SSL in order to avoid interference with firewalls as much as possible. The ports for communication and data transfers between the different servers themselves shall be configurable.

For each TCP connection k, $1 \leq k \leq n$, the client records the relative time t and the amount of data received from time 0 to t.

After completion of all tests, the client sends the results and data collected to the data server. Both datasets are then compared by the data server to check the quality and integrity of the result. All tests, successful or unsuccessful, are stored by the data server.

The values presented which are measured each 500 ms shall include the minimum, the average, and the maximum values.

Parallel to the data transmission, the single voice channel and multiple voice channel test should be established. The single voice channel test consists of the following parts:

– Convergence quality test;
– Listening speech quality test;
– DTMF test.

The four speech files should be repeated while the call is active. For n channels n −1 cycles (approximately 80 seconds) are recommended (e.g., for 4 channels the duration is 360 seconds), see also Figure 15.

Parallel to the single voice channel test, additional calls should be established (multiple voice channel test).

The multiple voice channel test consists of the following parts:

– Convergence quality test;
– Listening speech quality test.

Figure 16 depicts the detailed description of the multiple voice channel access.

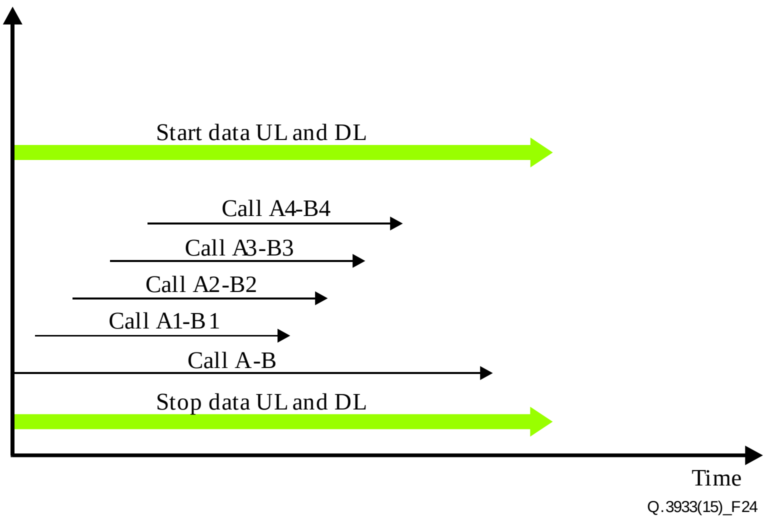

Figure 24 – Parallel quality measurement of multiple voice channels and data transmission speed

Relative time	Test equipment A		NETWORK		Test equipment B data reference system
T0 – 12 s	Ping	→		→	
Start download and upload procedure					
T0 – 10 s	Start TCP upload connection 1	→		→	
	Start TCP upload connection 2	→		→	
	Start TCP upload connection 3	→		→	
	Start TCP download connection 1	←		←	
	Start TCP download connection 2	←		←	
	Start TCP download connection 3	←		←	
The calculation of the throughput values for up/down stream starts					
T0 – 5	SETUP / INITE	→		→	SETUP / INITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK / ACK	→		→	CONNECT ACK / ACK
Start convergence quality test					
T0	Start Audio Receive BA_1 (male & female)	←		←	Start Audio Send BA_1 (male & female)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
Stop convergence quality test					
Listening speech quality test					
T0 + 10 s	Start Audio Send AB_1 (female 1)	→		→	Start Audio Receive AB_1 (female 1)
	End Audio Send AB_1 (female 1)	→		→	End Audio Receive AB_1 (female 1)
	Start Audio Send AB_2 (female 2)	→		→	Start Audio Receive AB_2 (female 2)
	End Audio Send AB_2 (female 2)	→		→	End Audio Receive AB_2 (female 2)
	Start Audio Send AB_3 (male 1)	→		→	Start Audio Receive AB_3 (male 1)
	End Audio Send AB_3 (male 1)	→		→	End Audio Receive AB_3 (male 1)

Relative time	Test equipment A		NETWORK		Test equipment B data reference system
	Start Audio Send AB_4 (male 2)	➔		➔	Start Audio Receive AB_4 (male 2)
	End Audio Send AB_4 (male 2)	➔		➔	End Audio Receive AB_4 (male 2)
	Pause				
	Start Audio Receive BA_1 (female 1)	🡐		🡐	Start Audio Send BA_1 (female 1)
	End Audio Receive BA_1 (female 1)	🡐		🡐	End Audio Send BA_1 (female 1)
	Start Audio Receive BA_2 (female 2)	🡐		🡐	Start Audio Send BA_2 (female 2)
	End Audio Receive BA_2 (female 2)	🡐		🡐	End Audio Send BA_2 (female 2)
	Start Audio Receive BA_3 (male 1)	🡐		🡐	Start Audio Send BA_3 (male 1)
	End Audio Receive BA_3 (male 1)	🡐		🡐	End Audio Send BA_3 (male 1)
	Start Audio Receive BA_4 (male 2)	🡐		🡐	Start Audio Send BA_4 (mal 2)
	End Audio Receive BA_4 (male 2)	🡐		🡐	End Audio Send BA_4 (male 2)
	Pause				
	Start DTMF Send AB_1	➔		➔	Start DTMF Receive AB_1
	End DTMF Send AB_1	➔		➔	End DTMF Receive AB_1
	Start DTMF Receive BA_1	🡐		🡐	Start DTMF Send BA_1
	End DTMF Receive BA_1	🡐		🡐	End DTMF Send BA_1
	DISCONNECT / BYE	🡐		🡐	DISCONNECT / BYE
	RELEASE / 200 OK	➔		➔	RELEASE /200 OK
	RELEASE COMPLETE	🡐		🡐	RELEASE COMPLETE
	The data transmission of data streams is stopped				
	The calculation of the throughput values for up/down stream				

Figure 25 – Detailed listening speech quality, DTMF procedure, and UL/DL procedure

7.16 Listening speech quality stability

The listening speech quality stability should be analysed all along the duration of the call.

This indicator takes into account the degradations generated on the signal by the transmission links.

Several measurements of MOS-LQO performed according to the method described in [ITU-T P.863] are done in succession within the same call.

The detailed testing method is described in [ITU-T G.131].

7.17 End-to-end audio delay

This parameter represents the global delay from one user to the other. This indicator takes into account the transmission delay of networks and also the processing delay in sending and receiving terminals. The end-to-end delay can be measured acoustically from mouth to ear, from one access point to the other. The delay can be calculated based on cross correlation between the signal at the MRP (at one access) and the signal at the ERP (at the other access) using the test methods as described e.g., in [ETSI ES 202 737] and [ETSI ES 202 739].

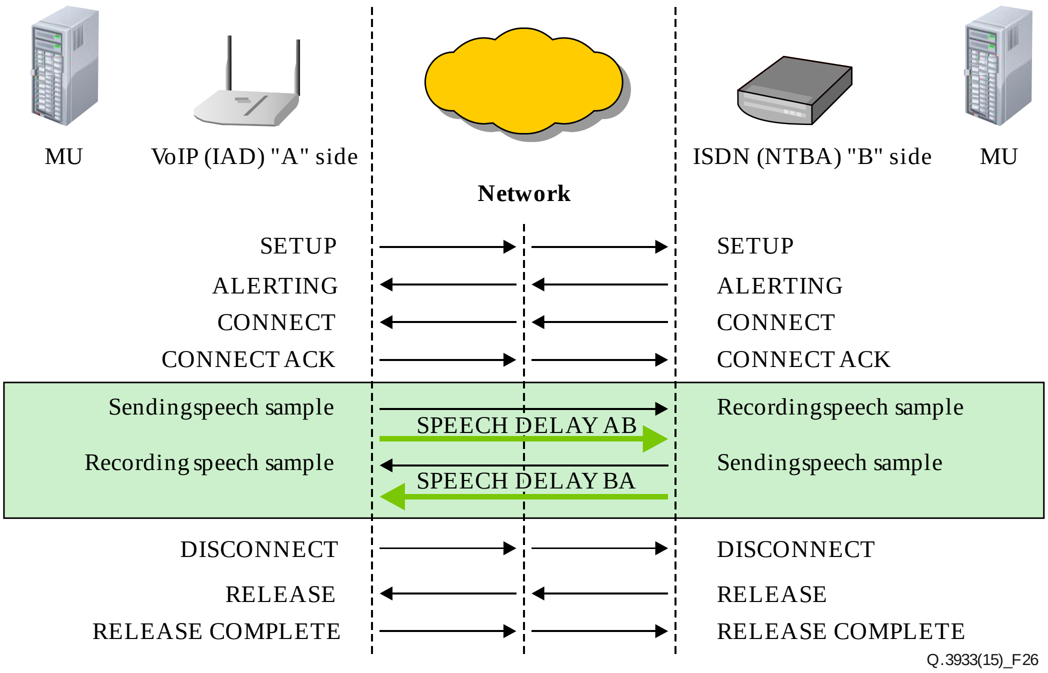

Figure 26 – Measurement of the speech delay

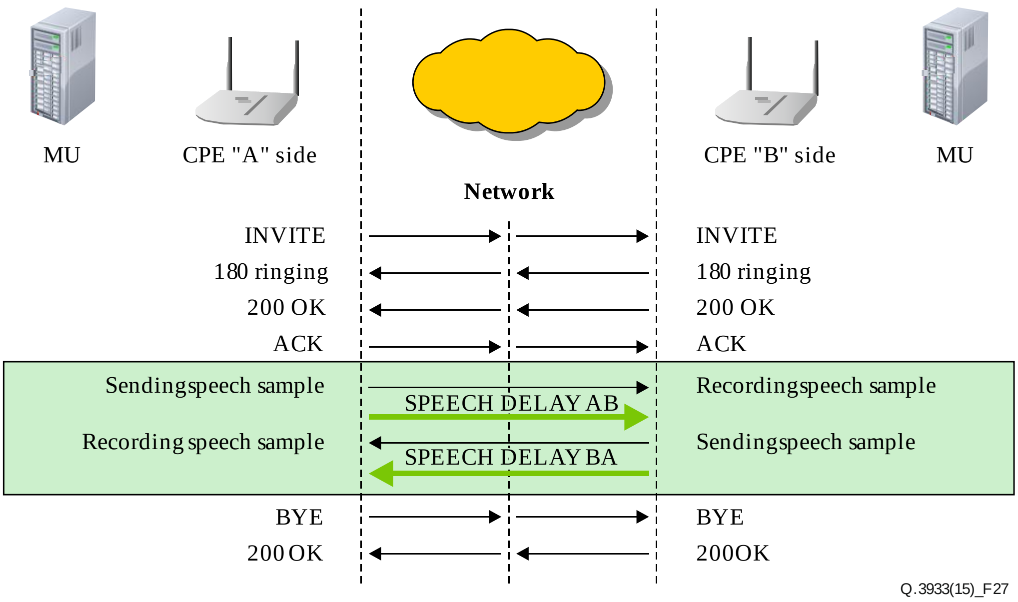

Figure 27 – VoIP measurement of the speech delay

Electrically the end-to-end delay can be measured based on cross correlation between the signal at the electrical measurement point at one access and the signal at the electrical measurement point at the other access.

The test signal consists of a series of composite source signals (CSSs) using a nominal network level of −16 dBm0 as described in [ITU-T P.501]. The test signal consists of the voiced part as described in [ITU-T P.501] followed by a pseudo random noise sequence with a periodicity of minimum 500 ms (described also in [ETSI ES 202 737] and [ETSI ES 202 739]).

NOTE – If the expected delay is higher than 500 ms a pseudo random sequence with a higher periodicity should be used.

7.18 End-to-end audio delay variation

The test signal consists of a series of CSSs using a nominal network level of −16 dBm0 with a total duration of 120 s. The pause of the CSS-sequence should be 150 ms. The delay of every CSS should be measured.

The delay variation for each CSS D(i) compared to the first CSS (as described in [ITU-T P.501]) of the analysis period is calculated:

$$D(i) = T1 - Ti$$

With:

T1 delay of the first CSS

Ti delay CSS number *i*

7.19 Frequency response in receive direction

Narrowband telephony should transmit signals between 300 Hz and 3 400 Hz. Wideband telephony should transmit signals between 50 Hz and 7 000 Hz. The objective of this measurement is to see which bandwidth is used, and also to see whether a partial and unwanted bandwidth limitation is present. The frequency response is the gain (or attenuation) of the speech spectrum after transmission. The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is −16 dBm0. The level is averaged over the complete test signal. For determining the frequency response in wideband the sensitivity frequency response is determined in 1/12th octave bands, as given by [IEC 61260] for frequencies of 100 Hz and 8 kHz, inclusive. In narrowband it is determined for frequencies from 200 Hz to 4 kHz. In each 1/12th octave band, the level of the measured signal is referred to the level of the reference signal averaged over the complete test sequence length. The sensitivity is determined in dBV/V.

7.20 Fax transmission in accordance with [ITU-T T.30] and [ITU-T T.38]

This test applies to fax bit rates ≤ 14.4 kbit/s and fax bit rates ≥ 14.4 kbit/s in accordance with [ITU-T T.30] and [ITU-T T.38]. Figure 28 gives an overview of the fax stack for FoIP.

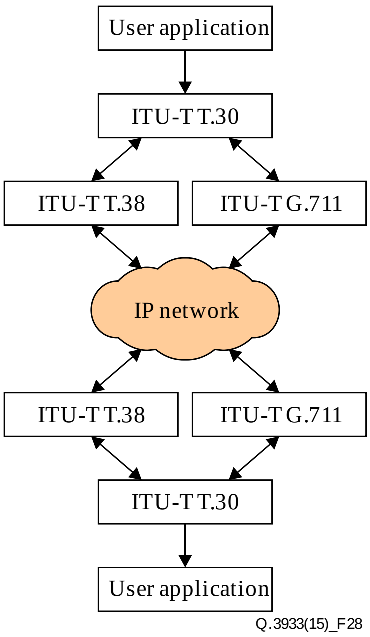

Figure 28 – FAX Stack for FoIP

Following is the electronic version of the test pages related to the test case descriptions.

The files consist of five test pages which are available as electronic attachments to [ITU-T T.24] or in the ITU test signal database: http://www.itu.int/net/itu-t/sigdb/menu.aspx. (The figures from the text of the Recommendation are not suitable for testing purposes). For all text pages, the version with 400 dpi resolution shall be selected for the electronic processing of the fax simulation.

– F03_400
– F04_400
– F07_400
– F09_400
– F20_400

A fax benchmarking test with parallel data transfer shall contain at least the following six test pages which are available as electronic attachments to [ITU-T T.24] or in the ITU test signal database: http://www.itu.int/net/itu-t/sigdb/menu.aspx. For all text pages, the version with 400 dpi resolution shall be selected for the electronic processing of the fax simulation.

– F03_400
– F04_400
– F07_400
– F09_400
– F18_400
– F20_400

For long term evaluation at least the following test pages shall be used which are available as electronic attachments to [ITU-T T.24] or in the ITU test signal database: http://www.itu.int/net/itu-t/sigdb/menu.aspx.

– F01_400
– F03_400
– F04_400
– F05_400
– F06_400
– F07_400
– F09_400
– F20_400
– F18_400
– EDUC
– AERIAL2
– CMPND3

The test pages defined shall be recorded and classified according to the following definitions:

The complete/incomplete transmission of page, received pages shall be stored with the test # as name.

1) Nominal bit rate of transmission;
2) Figure of merit (FOM) as defined in [ITU-T E.458]. There will be only one FOM value reported per fax transmission, independent of the number of pages;
3) Duration of transmission of test page in seconds;
4) Visual inspection of received page for visible errors and missing information.

Table 6 – From ITU-T E.458 – Definition of FOM

Transaction type	Complete	Maximum speed	Image quality
I	Yes	Yes	Error-free
II	Yes	Yes	Errored
III	Yes	Yes	Severely errored
IV	Yes	No	Error-free
V	Yes	No	Errored
VI	Yes	No	Severely errored
VII	No	Not applicable	Not applicable
NOTE 1 – Error-free, errored, and severely errored transactions are as defined in [ITU-T E.453]. NOTE 2 – If the transaction is incomplete, it is categorized as type VII irrespective of the speed and image quality of the completed pages.			

Table 7 – From ITU-T E.453 – Image quality categories

error-free page	No degradation by network impairments
errored page	Information conveyed
severely-errored page	Part of information missing

7.20.1 Fax set-up duration

To determine the fax set-up duration, the time in seconds is measured from the sending of the dialling information by the "A" side to the start of transmission of the fax page on the "A" side (see green arrow in Figure 29).

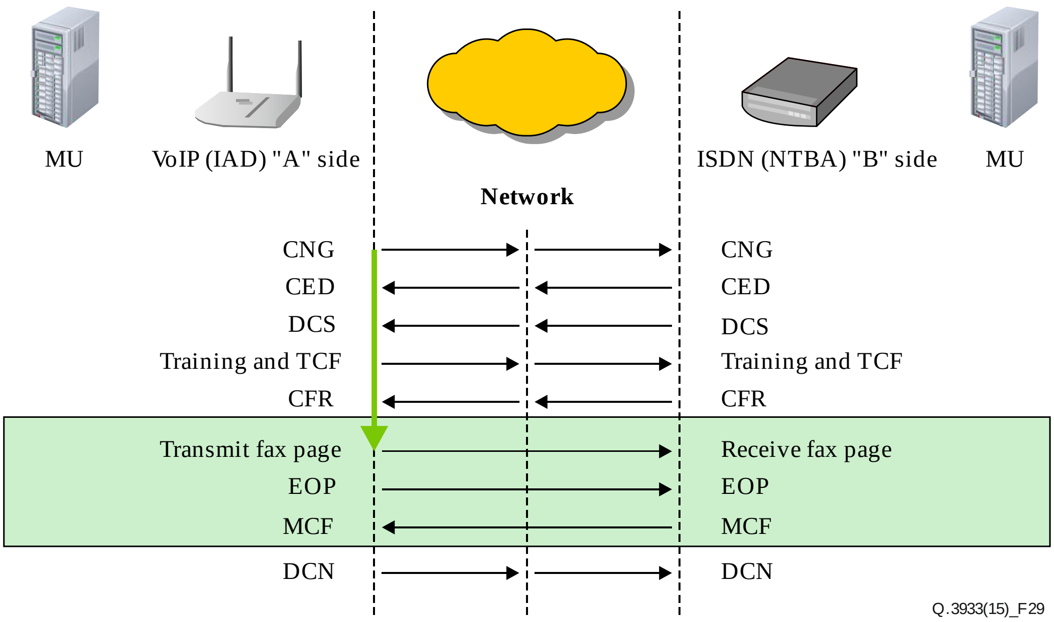

Figure 29 – Fax set-up duration

7.20.2 Fax transmission duration

This value measured shows the transmission time of a fax page in seconds.

The fax transmission duration is defined in the context of this document as the time that elapses from the start of transmission of the fax page by the "A" side until the complete transfer of the fax page to the "B" side (see green arrow in Figure 30).

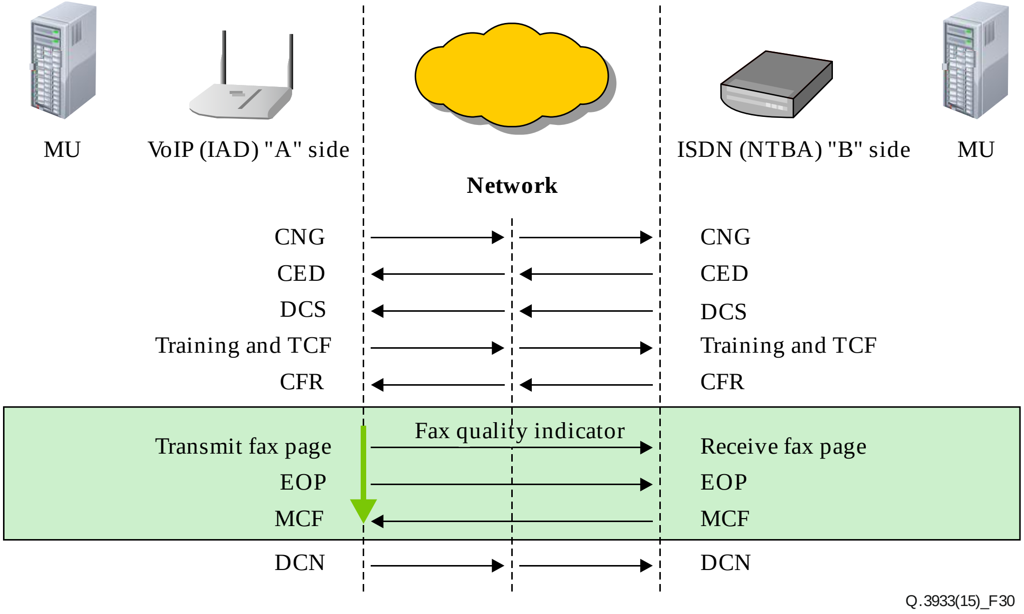

Figure 30 – Fax transmission duration

7.20.3 Fax failure ratio

The fax failure ratio is defined as the ratio of failed fax transmissions and all fax transmissions initiated.

A fax transmission is considered to have failed if the fax connection set-up or the fax transmission is unsuccessful, or if the fax connection set-up and fax were not completed within 180 seconds.

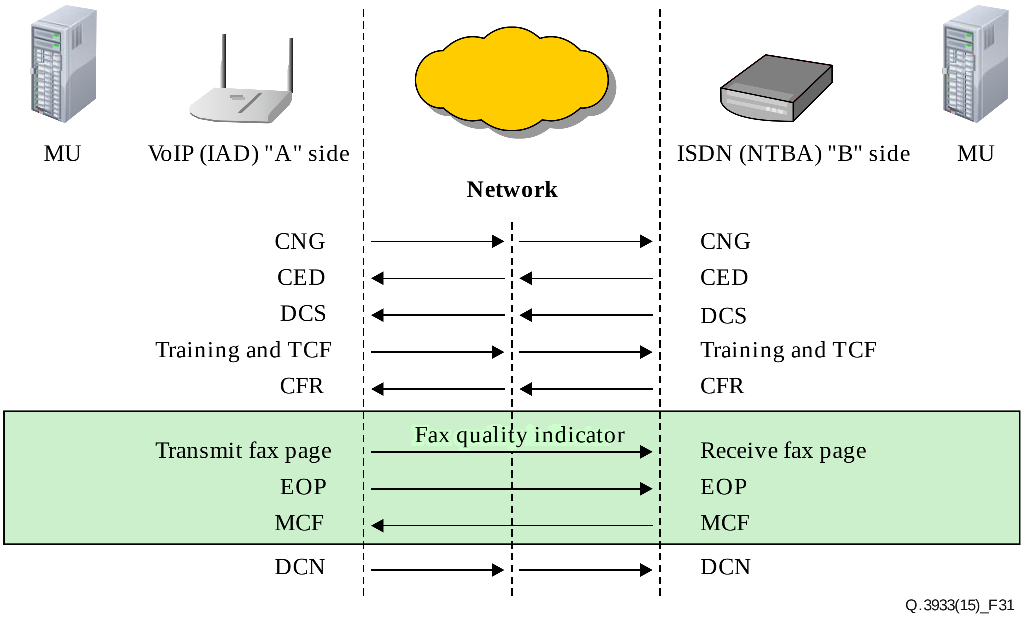

Figure 31 – Fax failure ratio

7.20.4 Test case descriptions

7.20.4.1 Quality measurement of one fax channel

Table 8 gives an overview of the connection options without parallel data transfer; Figure 32 gives a detailed overview of a single fax channel test.

Table 8 – Transmission options without additional data traffic

No.	From		Fax protocol relay	To	
	ITU-T G.711 modem type / ITU-T T.38/ASN.1/t30-indicator	IP gateway type/ interworking		IP gateway type/ interworking	ITU-T G.711 modem type / ITU-T T.38/ASN.1/t30-indicator e
1	ITU-T V.34	CPE ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.34
2	ITU-T V.17	CPE ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.17
3	ITU-T V.34	CPE ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.17
4	ITU-T V.34	CPE ITU-T T.38	ITU-T T.38	AGW/MSAN ITU-T G.711/ ITU-T T.38	ITU-T V.34
5	ITU-T V.17	CPE ITU-T T.38	ITU-T T.38	AGW/MSAN ITU-T G.711/ ITU-T T.38	ITU-T V.17
6	ITU-T V.34	CPE ITU-T T.38	ITU-T T.38	AGW/MSAN ITU-T G.711/ ITU-T T.38	ITU-T V.17
7	ITU-T V.34	CPE ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
8	ITU-T V.17	CPE ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
9	ITU-T V.34	CPE ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
10	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.34
11	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.17
12	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.17
13	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.34
14	ITU-T V.17	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.17
15	ITU-T V.34	AGW / MSAN ITU-T G.711/ ITU-T T.38	ITU-T T.38	CPE ITU-T T.38	ITU-T V.17

Table 8 – Transmission options without additional data traffic

No.	From		Fax protocol relay	To	
16	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
17	ITU-T V.17	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
18	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
19	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
20	ITU-T V.17	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
21	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
22	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
23	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
24	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
25	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T T.38	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
26	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38	ITU-T T.38	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
27	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T T.38	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
28	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.152	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
29	ITU-T V.17	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.152	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
30	ITU-T V.34	AGW ITU-T G.711/ITU-T T.38	ITU-T T.152	AGW ITU-T G.711/ITU-T T.38	ITU-T V.17
31	ITU-T V.34	AGW ITU-T G.711/ITU-T T.38	ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
32	ITU-T V.17	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
33	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T T.152ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17

Table 8 – Transmission options without additional data traffic

No.	From		Fax protocol relay	To	
34	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
35	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38	ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
36	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
37	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T T.152	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
38	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38	ITU-T T.152	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
39	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T T.152	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
40	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T G.711	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
41	ITU-T V.17	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T G.711	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
42	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T G.711	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
43	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
44	ITU-T V.17	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
45	ITU-T V.34	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
46	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
47	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
48	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
49	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T G.711	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
50	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38	ITU-T G.711	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
51	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38	ITU-T G.711	AGW/MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17

Relative Time	Test equipment A		Network		Test equipment B
T0 – 2	SETUP / INVITE	→		→	SETUP / INVITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK/ ACK	→		→	CONNECT ACK/ ACK
	Start Fax Transmission				
T0	Start Fax Send AB_1 (F09_400)	→		→	Start Fax Receive AB_1 (F09_400)
	End Fax Send AB_1 (F09_400)	→		→	End Fax Send AB_1 (F09_400)
	Start Fax Send AB_2 (F03_400)	→		→	Start Fax Receive AB_2 (F03_400)
	End Fax Send AB_2 (F03_400)	→		→	End Fax Receive AB_2 (F03_400)
	Start Fax Send AB_3 (F04_400)	→		→	Start Fax Receive AB_3 (F04_400)
	End Fax Send AB_3 (F04_400)	→		→	End Fax Send AB_3 (F04_400)
	Start Fax Send AB_4 (F07_400)	→		→	Start Fax Receive AB_4 (F07_400)
	End Fax Send AB_4 (F07_400)	→		→	End Fax Receive AB_4 (F07_400)
	Start Fax Send AB_5 (F20_400)	→		→	Start Fax Receive AB_5 (F20_400)
	End Fax Send AB_5 (F20_400)	→		→	End Fax Receive AB_5 (F20_400)
	End Fax Transmission				
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE

Figure 32 – Single fax channel test

7.20.4.2 Quality measurement of one fax channel and parallel data transfer

In the case when the access link is used for voice, fax, and data application the fax quality measurement sequence with parallel upload/download shall be used. Table 9 gives an overview of the connections options with parallel data transfer.

For data transfer the following constants are used:

– n number of parallel TCP connections for one direction ($1 \leq n \leq 10$); default: $n = 3$.

– In case of use of fixed-size chunks, the initial size of data block sent during the test is s = 4.096 Bytes.

Several parallel data streams are initiated with a number of n parallel TCP connections with an upload and download of data files from the data-reference system. The upload shall start before the call set-up starts, the download before the voice quality measurement starts. After time t, each TCP connection shall be reset.

As an option, the server can continuously send data streams consisting of fixed-size chunks of size s (randomly generated data with high entropy). The data should be transferred between client and server over the TCP port 443 using TLS or SSL in order to avoid interference with firewalls as much as possible. The ports for communication and data transfers between the different servers themselves shall be configurable. Figure 33 gives an overview of the quality of measurement of one fax channel and parallel data load, Figure 34 gives an overview of the detailed download and upload procedure for an automatically controlled test sequence and Figure 35 gives an overview of the detailed download and upload procedure for a manually controlled test sequence.

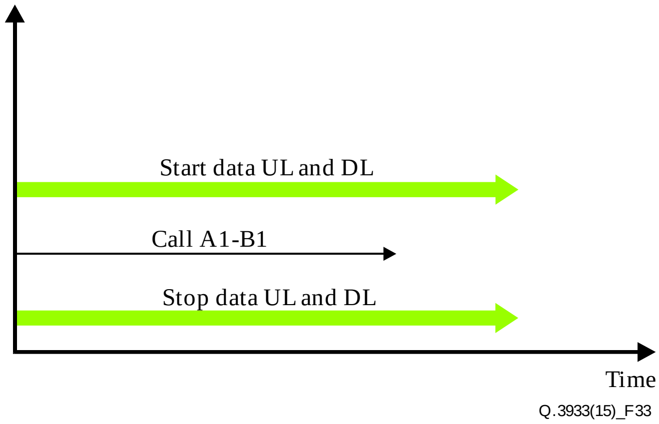

Figure 33 – Quality measurement of one fax channel and parallel data load

Relative Time	Test equipment A		Network		Test equipment B Data reference system
Start download and upload procedure					
T0 - 10 s	Start TCP upload connection 1	→		→	
	Start TCP upload connection 2	→		→	
	Start TCP upload connection 3	→		→	
	Fax test				
T0 - 5 s	SETUP / INVITE	→		→	SETUP / INVITE
	Start TCP download connection 1	←		←	
	Start TCP download connection 2	←		←	
	Start TCP download connection 3	←		←	
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK / ACK	→		→	CONNECT ACK / ACK
Start fax transmission					
	Start Fax Send AB_1 (F09_400)	→		→	Start Fax Receive BA_1 (F09_400)
continued					
	End Fax Send AB_6 (F20_400)	→		→	End Fax Receive AB_6 (F20_400)
End fax transmission					
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE
	Stop download and upload procedure				

Figure 34 – Detailed download and upload procedure for automatically controlled test sequence

Relative Time	Test equipment A		Network		Test equipment B Data reference system
Start download and upload procedure					
T0 - 10 s	Start TCP upload connection 1	→		→	
	Start TCP upload connection 2	→		→	
	Start TCP upload connection 3	→		→	
	Start TCP download connection 1	←		←	
	Start TCP download connection 2	←		←	
	Start TCP download connection 3	←		←	
	Fax Test				
T0 - 5 s	SETUP / INVITE	→		→	SETUP / INVITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK / ACK	→		→	CONNECT ACK / ACK
Start fax transmission					
	Start Fax Send AB_1 (F09_400)	→		→	Start Fax Receive BA_1 (F09_400)
continued					
	End Fax Send AB_6 (F20_400)	→		→	End Fax Receive AB_6 (F20_400)
End fax transmission					
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE
	Stop download and upload procedure				

Figure 35 – Detailed download and upload procedure for manually controlled tests sequence

Table 9 – Transmission options with additional data traffic

No.	From		Fax protocol relay	To	
	ITU-T G.711 modem type / ITU-T T.38/ASN.1/t30-indicator	IP gateway type/ interworking		IP gateway type/ interworking	ITU-T G.711 modem type / ITU-T T.38/ASN.1/ITU-T T.30-indicator e
1	ITU-T V.34	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	CPE ITU-T T.38	ITU-T V.34
2	ITU-T V.17	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	CPE ITU-T T.38	ITU-T V.17
3	ITU-T V.34	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	CPE ITU-T T.38	ITU-T V.17

Table 9 – Transmission options with additional data traffic

No.	From		Fax protocol relay	To	
4	ITU-T V.34	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
5	ITU-T V.17	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
6	ITU-T V.34	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
7	ITU-T V.34	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
8	ITU-T V.17	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
9	ITU-T V.34	CPE ITU-T T.38 with additional data and voice traffic	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
10	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
11	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
12	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.38	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
13	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.38	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
14	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.38	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
15	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.38	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17

Table 9 – Transmission options with additional data traffic

No.	From		Fax protocol relay	To	
16	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
17	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
18	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.152	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
19	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.152	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
20	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.152	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
21	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T T.152	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
22	ITU-T V.34	ITU-T T.30 VGW with additional data and voice traffic	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.34
23	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
24	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T G.711	VGW ITU-T G.711/ITU-T T.38	ITU-T V.17
25	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T G.711	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.34
26	ITU-T V.17	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T G.711	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17
27	ITU-T V.34	VGW ITU-T G.711/ITU-T T.38 with additional data and voice traffic	ITU-T G.711	AGW / MSAN ITU-T G.711/ITU-T T.38	ITU-T V.17

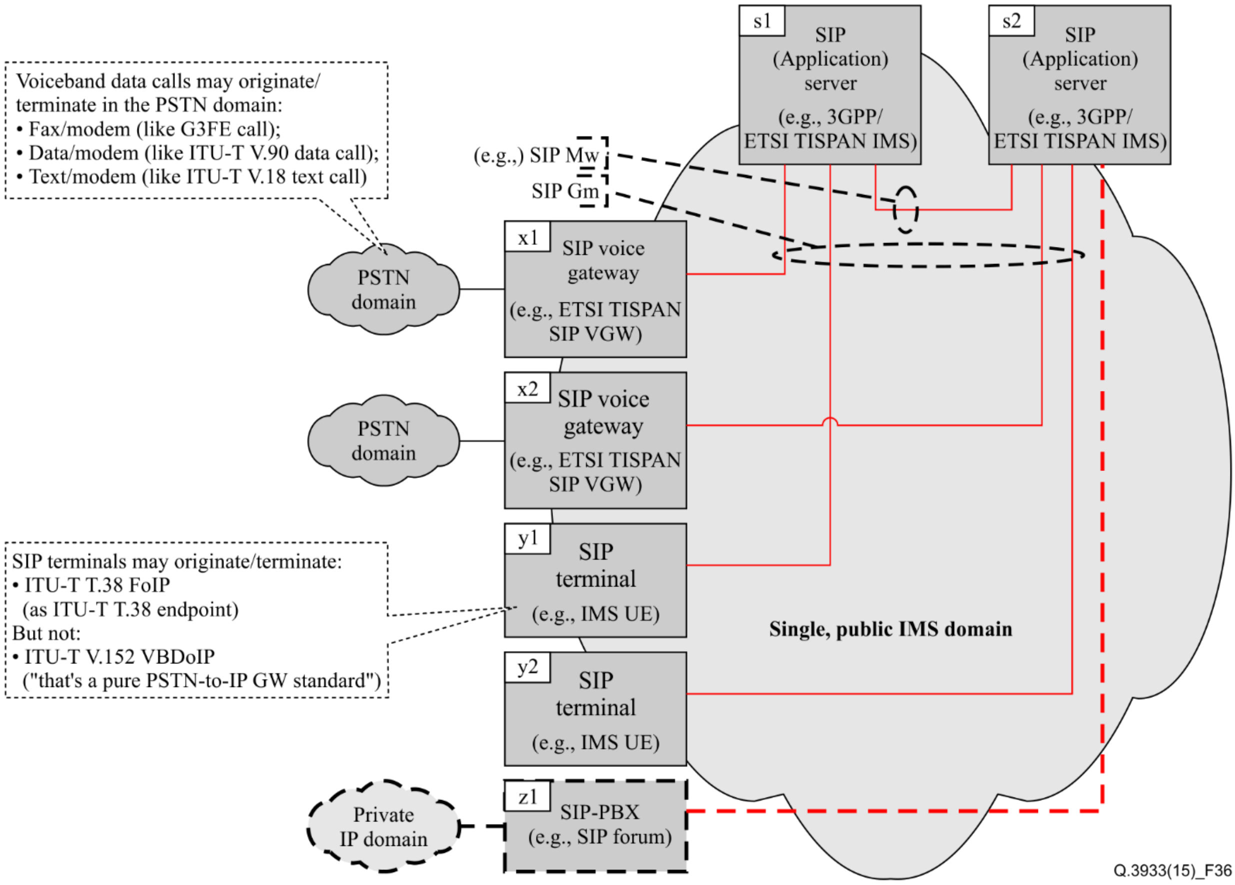

Figure 36 – Mix of SIP VGWs (IMS-based PES) & SIP user equipment (UE) (IMS)

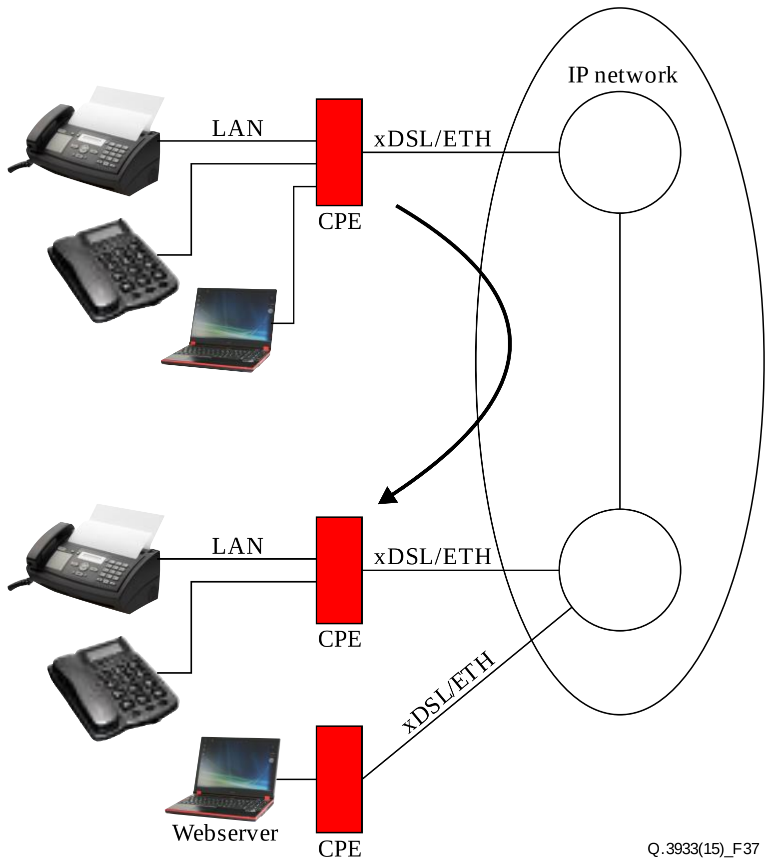

Figure 37 – Call between two MMTel (IMS) fax UE with additional data traffic

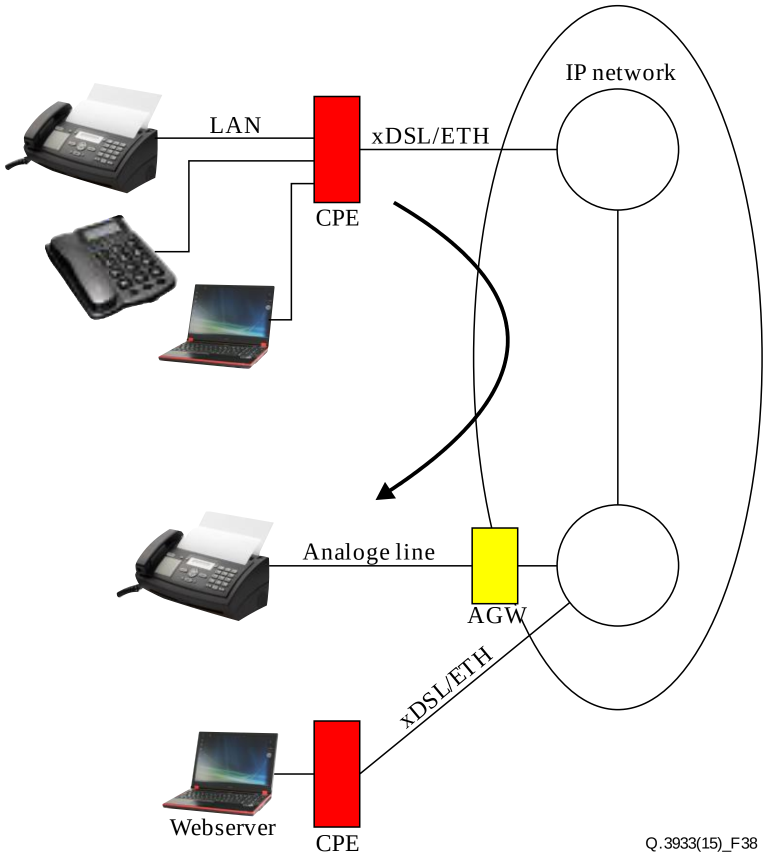

Figure 38 – Call between MMTel (IMS) fax UE with additional data traffic and AGW fax UE

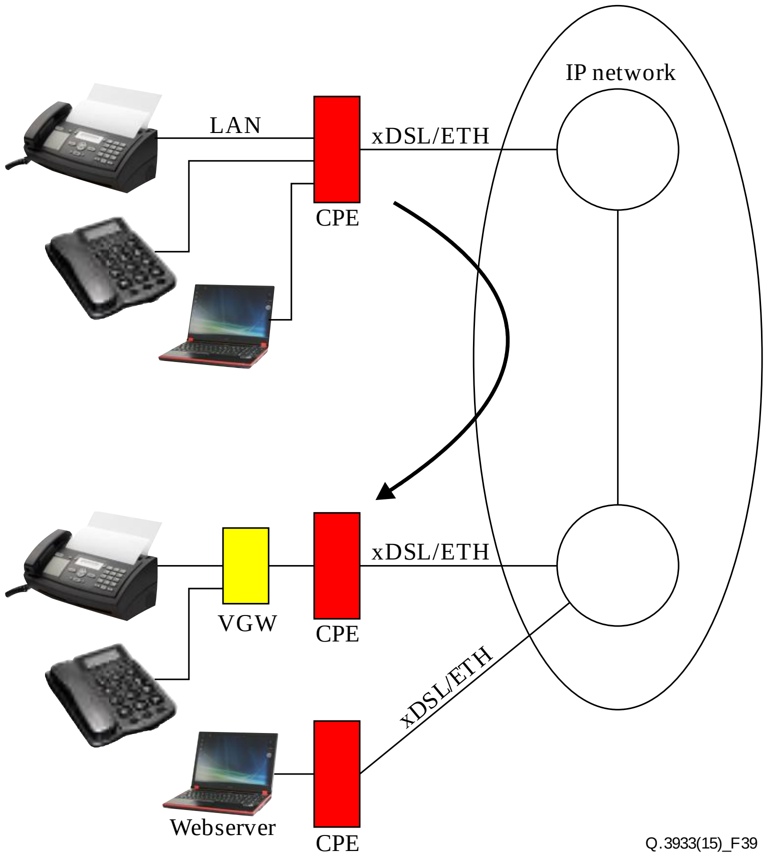

Figure 39 – Call between MMTel (IMS) fax UE with additional data traffic and VGW fax UE

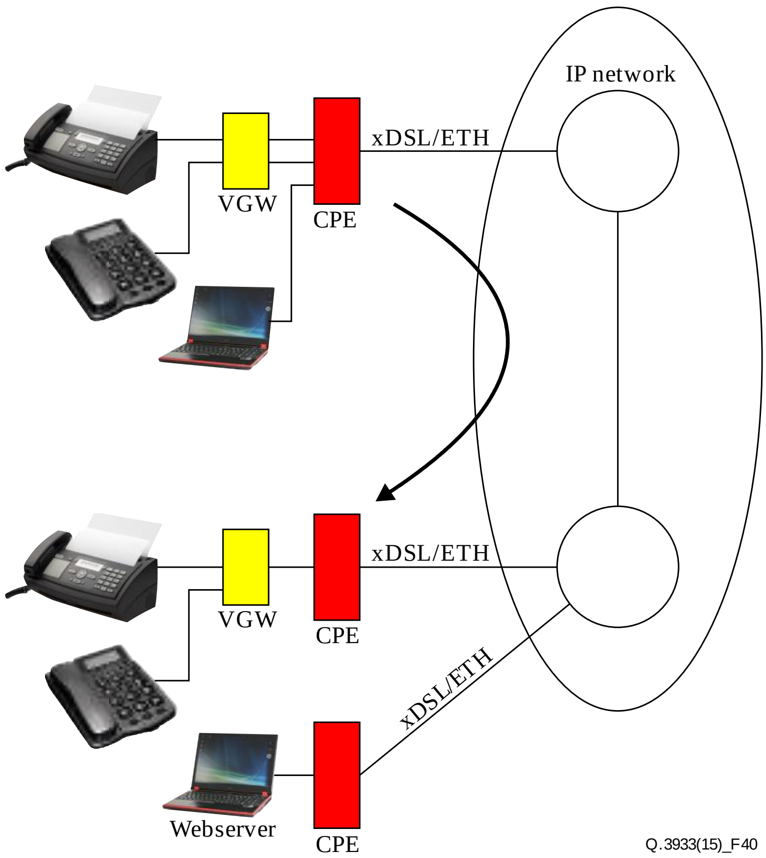

Figure 40 – Call between two VGW fax UE with additional data traffic

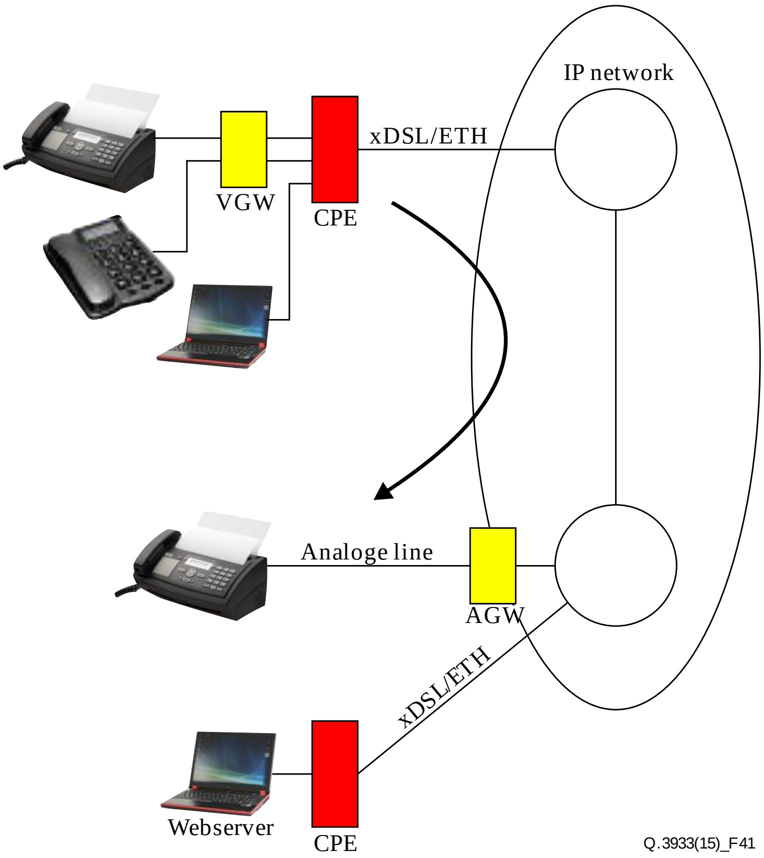

Figure 41 – Call between VGW fax UE with additional data traffic and AGW fax UE

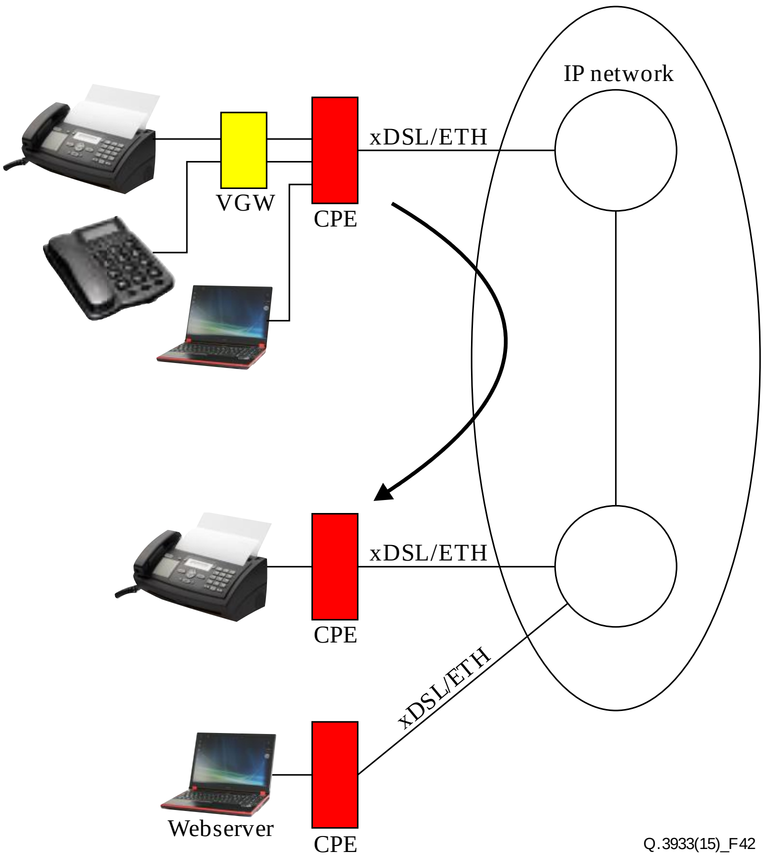

Figure 42 – Call between VGW fax UE with additional data traffic and MMTel (IMS)

7.21 Early media listening speech quality

Early media refers to media (e.g., audio and video) which are exchanged before a particular session is accepted by the called user (in terms of the signalling). Within a dialogue, early media occurs from the moment the initial INVITE is sent until the user agent server (UAS) generates a final response. It may be unidirectional or bidirectional, and can be generated by the caller, the called party, or both. Typical examples of early media generated by the called party are ringing tone and announcements (e.g., queuing status). Early media generated by the caller typically consists of voice commands or dual tone multi-frequency (DTMF) tones to drive interactive voice response (IVR) systems.

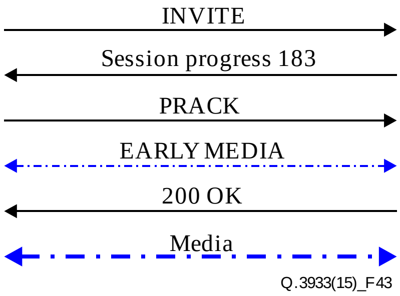

Figure 43 – Early media SIP overview

7.21.1 Early media generated by the called party

To test the early media, the listening speech quality test from B to A after the '183 Session Progress' message is sent. In case of IMS implementations the '183 Session Progress' should contain the P-Early – media header.

In case the called user is an ISDN user an ALERTING with progress indicator #8 or a progress message with the progress indicator should be sent.

For the synchronization of the voice samples a 700 Hz tone (100 ms signal) as trigger event can be used.

The principle of testing 'early media' is the same as defined for the convergence quality test according to clause 7.15.2. However, only one speech sample (male/female voice) has to be transmitted from the called party to the calling party emulating the 'early media' transfer. The general technical aspects for these speech samples are the same as defined in clause 7.15.2.

The call flow and the application of the early media test are shown in Figures 44, 45 and 46.

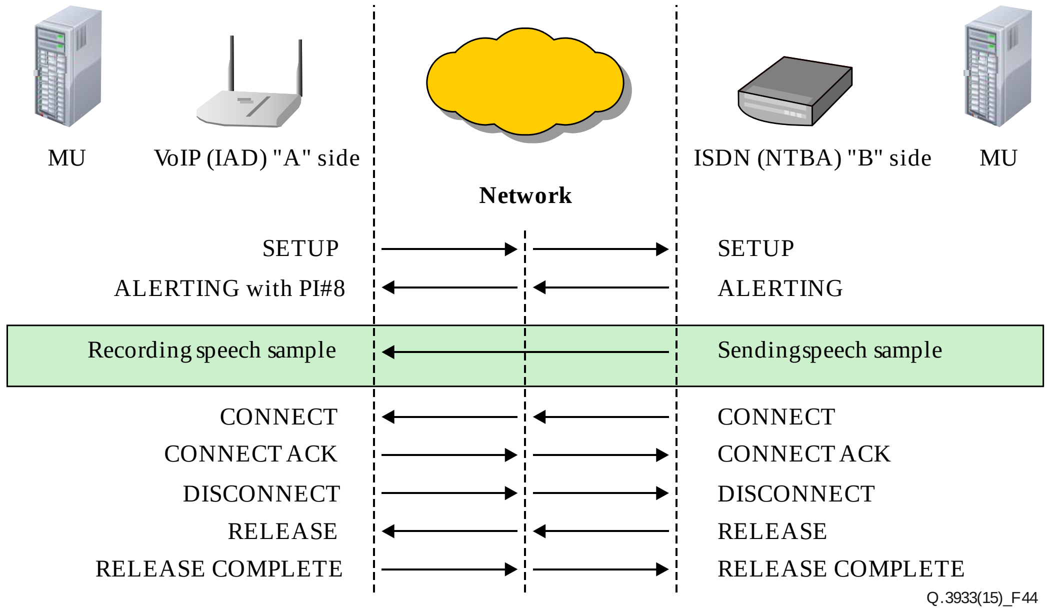

Figure 44 – Early media ISDN case A

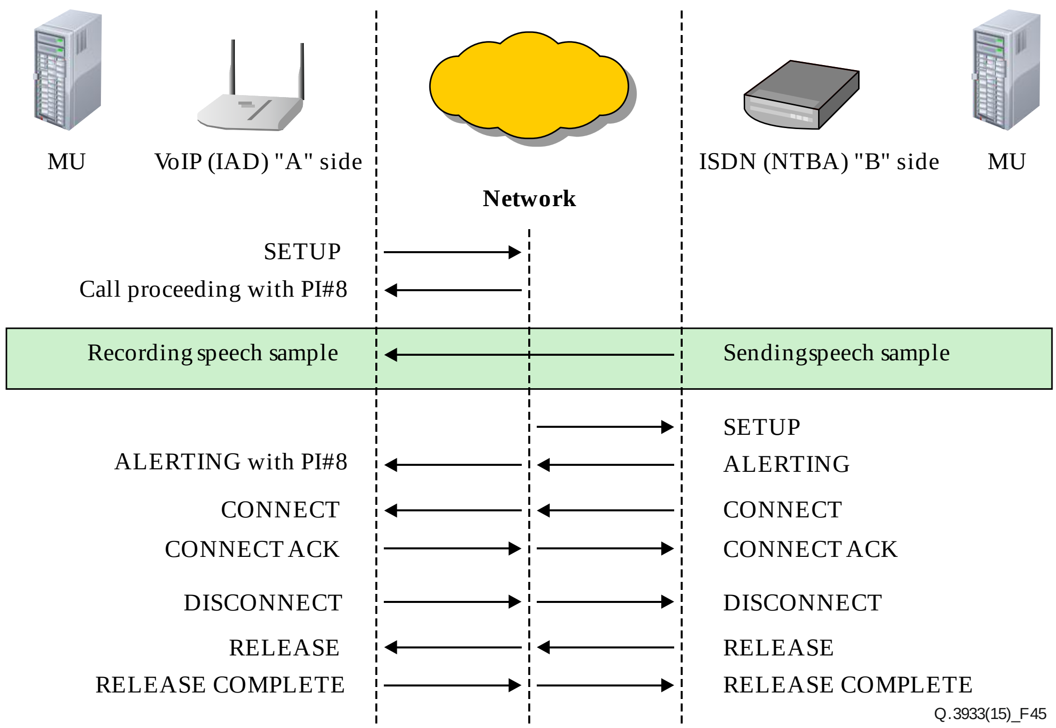

Figure 45 – Early media ISDN case B

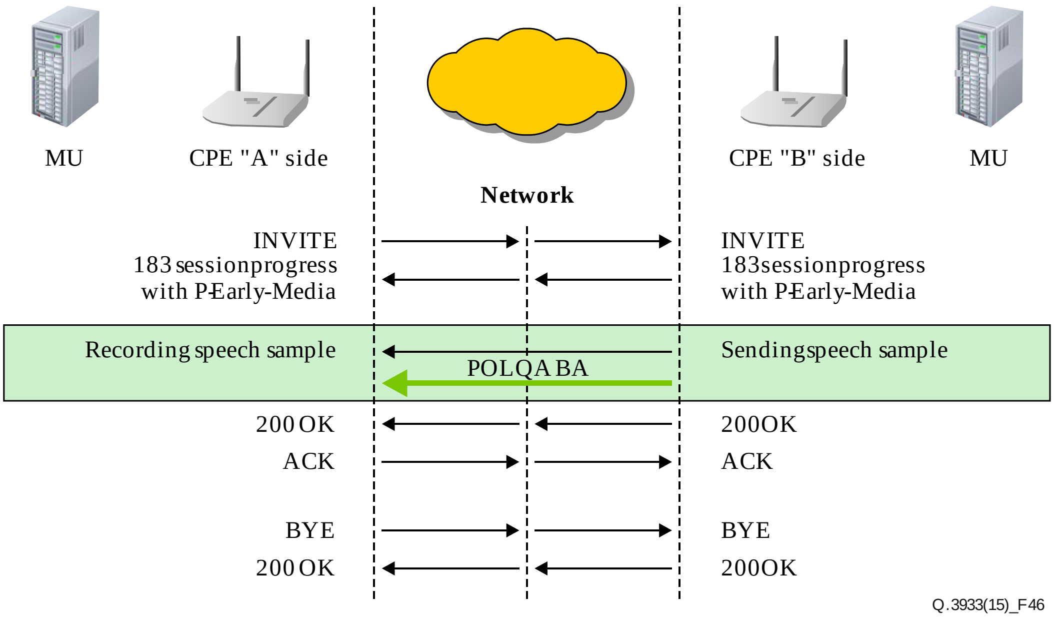

Figure 46 – Early media SIP

7.22 Jitter buffer and IP prioritization response time

7.22.1 Jitter buffer and IP prioritization response time without data transfer

To test the jitter buffer and IP prioritization response time the test starts directly with a listening speech quality test from the called party to the calling party directly after the connection is established. After the call is released the test is repeated in the opposite direction.

Only one speech sample (male/female voice) has to be transmitted from the called party to the calling party emulating the 'early media' transfer. The general technical aspects for the speech samples are the same as defined in clause 7.15.2.

The call flow and the application of the early media test are shown in Figure 47.

Relative Time	Test equipment A		Network		Test equipment B
	CALL A to B				
T0 - 2	SETUP / INVITE	→		→	SETUP / INVITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK/ ACK	→		→	CONNECT ACK/ ACK
T0	Start Audio Receive BA_1 (male & female)	←		←	Start Audio Send BA_1 (male & female)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE
			Call B to A		
T0 - 2	SETUP / INVITE	←		←	SETUP / INVITE
	ALERTING / 180 Ringing	→		→	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	→		→	CONNECT / 200 OK
	CONNECT ACK/ ACK	←		←	CONNECT ACK/ ACK
T0	Start Audio Send AB_1 (male & female) Start Audio Receive AB_1	→		→	Start Audio Receive AB_1 (female & female) Start Audio Send AB_1
	End Audio Receive BA_1	→		→	End Audio Send AB_1
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE

Figure 47 – Jitter buffer and IP prioritization response time tests

7.22.2 Jitter buffer and IP prioritization response time with data transfer

For the data transfer the following constants are used:

– n number of parallel TCP connections for one direction ($1 \leq n \leq 10$); default: $n = 3$.

– In case of use of fixed-size chunks, the initial size of data block sent during the test is s = 4.096 Bytes.

– t duration of tests, approximately t = 22 s.

Several parallel data streams are initiated with a number of *n* parallel TCP connections with an upload and download of data files from the data-reference system.

All TCP transmissions start at the same time, which is denoted as relative time 0. After time t, each TCP connection is reset.

As an option, the server can continuously send data streams consisting of fixed-size chunks of size s (randomly generated data with high entropy). The data should be transferred between client and server over the TCP port 443 using TLS or SSL in order to avoid interference with firewalls as much as possible. The ports for communication and data transfers between the different servers themselves shall be configurable.

Parallel to the data transmission, the jitter buffer and IP prioritization response time test should be established.

To test the jitter buffer and IP prioritization response time the test starts with a listening speech quality test from the called party to the calling party. After the call is released the test is repeated in the opposite direction. The flow of the voice call is the same as in clause 7.22.1, there is just a surrounding and background data transfer active as shown in Figure 48.

Relative Time	Test equipment A		Network		Test equipment B
Start download and upload procedure					
T0 – 10 s	Start TCP upload connection 1	→		→	
	Start TCP upload connection 2	→		→	
	Start TCP upload connection 3	→		→	
	Start TCP download connection 1	←		←	
	Start TCP download connection 2	←		←	
	Start TCP download connection 3	←		←	
Call A to B					
T0 – 5	SETUP / INVITE	→		→	SETUP / INVITE
	ALERTING / 180 Ringing	←		←	ALERTING / 180 Ringing
T0	CONNECT / 200 OK	←		←	CONNECT / 200 OK
	CONNECT ACK/ ACK	→		→	CONNECT ACK/ ACK
T0	Start Audio Receive BA_1(male & female)	←		←	Start Audio Send BA_1 (male & female)
	End Audio Receive BA_1	←		←	End Audio Send BA_1
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE
Call B to A					
T12	SETUP / INVITE	←		←	SETUP / INVITE
	ALERTING / 180 Ringing	→		→	ALERTING / 180 Ringing
T14	CONNECT / 200 OK	→		→	CONNECT / 200 OK
	CONNECT ACK/ ACK	←		←	CONNECT ACK/ ACK
T14	Start Audio Send AB_1 (male & female)	→		→	Start Audio Receive AB_1 (male & female)
	End Audio Receive AB_1	→		→	End Audio Send AB_1
	DISCONNECT / BYE	→		→	DISCONNECT / BYE
	RELEASE / 200 OK	←		←	RELEASE / 200 OK
	RELEASE COMPLETE	→		→	RELEASE COMPLETE
Stop download and upload procedure					

Figure 48 – Jitter buffer and IP prioritization response time test with data transfer

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