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SERIES Q: SWITCHING AND SIGNALLING

Testing specifications – Testing specifications for SIP-IMS

Testing specification of call establishment procedures based on SIP/SDP and ITU-T H.248 for a real-time fax over IP service

Recommendation ITU-T Q.4016

ITU-T

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Recommendation ITU-T Q.4016

Testing specification of call establishment procedures based on SIP/SDP and ITU-T H.248 for a real-time fax over IP service

Summary

Recommendation ITU-T Q.4016 contains the testing specification of call establishment procedures based on session initiation protocol (SIP)/session description protocol (SDP) and ITU-T H.248 for a real-time fax over IP service. The listed test requirements in this Recommendation have to be interpreted as "minimal requirements" for fax support between SIP-enabled devices for real-time fax over IP.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
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Fax, ITU-T G.711, ITU-T T.38, ITU-T V.152, SIP.

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FOREWORD

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Testing specification of call establishment procedures based on SIP/SDP and ITU-T H.248 for a real-time fax over IP service

1 Scope

This Recommendation contains the framework of session initiation protocol (SIP) signalling test requirements between SIP-enabled devices for real-time fax over IP (FoIP).

Three methods for the support of fax over IP have been identified:

1) Pseudo-VBDoIP emulation;

2) FoIP packet relay with ITU-T T.38
   - [ITU-T T.38], Annex D for call control signalling (SIP/session description protocol (SDP));
   - [ITU-T T.38], Annex E for gateway control signalling ITU-T H.248;

3) VBDoIP pass-through method as defined in [ITU-T V.152].

Where there are discrepancies among this Recommendation, [ITU-T T.38], [ITU-T V.151], [ITU-T V.152] and [ITU-T V.153], these other Recommendations take precedence over the text and procedures referred to by this Recommendation.

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 G.168] Recommendation ITU-T G.168 (2015), *Digital network echo cancellers*.

[ITU-T G.799.4] Recommendation ITU-T G.799.4 (2014), *Procedures for the control of de-jitter buffers used in PSTN-IP gateways carrying voiceband data*.

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

[ITU-T Q.3403] Recommendation ITU-T Q.3403 (2016), *IP multimedia call control protocol based on the session initiation protocol and the session description protocol. Basic call: Requirements for the user side and network side*.

[ITU-T Q.3629 v.1] Recommendation ITU-T Q.3629 v.1 (2016), *Interworking between the IP multimedia core network subsystem and circuit switched networks – Protocol specification*.

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

[ITU-T T.6] Recommendation ITU-T T.6 (1988), *Facsimile coding schemes and coding control functions for Group 4 facsimile apparatus*.

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

[ITU-T T.38] Recommendation ITU-T T.38 (2015), *Procedures for real-time Group 3 facsimile communication over IP networks*.

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

[ITU-T V.150.1] Recommendation ITU-T V.150.1 (2003), *Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs*.

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

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

[ITU-T V.153] Recommendation ITU-T V.153 (2009), *Interworking between Recommendation ITU-T T.38 and ITU-T V.152 using IP peering for real-time facsimile services*.

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

[ETSI TS 126 114] ETSI TS 126 114 V12.15.0 (2016), *Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction (3GPP TS 26.114 version 12.13.0 Release 12)*.

[ETSI TS 129 163] ETSI TS 129 163 V10.19.0 (2016-01), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 10.19.0 Release 10)*.

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

[IETF RFC 4734] IETF RFC 4734 (2006), *Definition of Events for Modem, Fax, and Text Telephony Signals*.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

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

3.1.2 pseudo-VBDoIP emulation service [ITU-T G.799.4]: A XoIP emulation service, trying to support voiceband data in audio mode (see clause 3.2.1 of [ITU-T V.152]), also known as non-V.152 VBDoIP service.

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

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

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

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

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

3.1.7 VBDoIP emulation service [ITU-T G.799.4]: A XoIP emulation service compliant to [ITU-T V.152].

3.1.8 voice-band data mode [ITU-T V.152]: The transport of voice-band data over a voice channel of a packet network with the encoding appropriate for modem signals as defined in clause 6 of [ITU-T V.152].

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

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

– Facsimile/modem: Gateway technologies for PSTN-to-IP interworking see e.g., [ITU-T V.152] for pass-through mode and [ITU-T T.38] as packet-relay mode;
– Text/modem: Gateway technologies for PSTN-to-IP interworking see e.g., [ITU-T V.152] for pass-through mode and [ITU-T V.151] as packet-relay mode; and

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

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

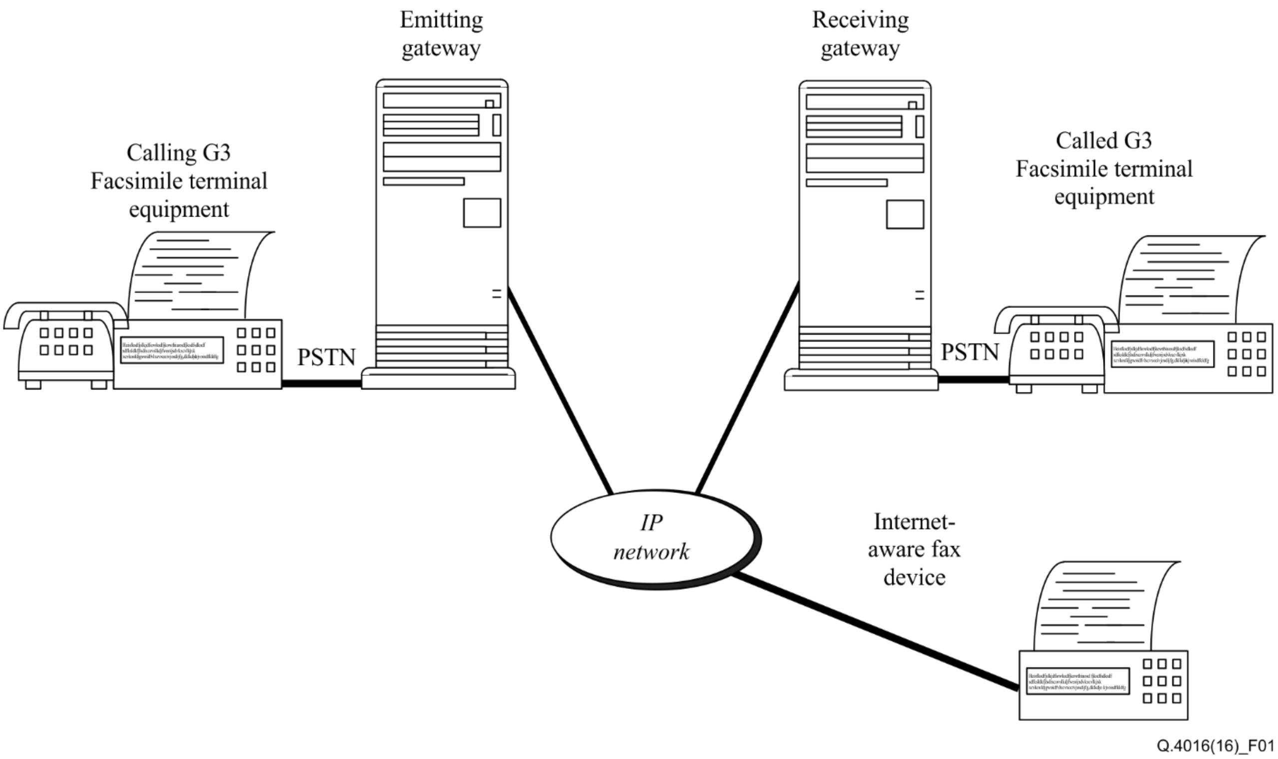
AGW	Access Gateway
ATA	Analogue Terminal Adapter
CED	Called station identification
CNG	Calling tone
CPE	Customer Premises Equipment
DTMF	Dual Tone Multi Frequency
ETSI	European Telecommunications Standards Institute
FD	Fax Device
FEC	Forward Error Correction
FoIP	Fax over IP

FOM	Figure of Merit
G3FE	Group 3 Facsimile Equipment
HGW	Home Gateway
IAD	Integrated Access Device
IAF	Internet-Aware Fax
IMS	IP Multimedia Subsystem
ITS	IP Telephony Services
MGW	Media Gateway
PSTN	Public Switched Telephone Network
PT	Payload Type
QoS	Quality of Service
RFC	Request for Comment
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
UNI	User Network Interface
VBD	Voiceband Data
VBDoIP	Voiceband Data over IP
VGW	Voice Gateway (SIP)
VoIP	Voice over IP
XoIP	X over IP (with 'X' as placeholder for "IP application protocol 'X'")

5 Media-type configurations for PSTN modem calls (from ETSI TR 183 072)

5.1 Overview

Any SDP *Offered Configuration List* (OCL; briefly "*Offered Codec. List*") for public switched telephone network (PSTN) modem calls provides *audio* support. There might also be packet relay support for "auxiliary information" like inband application control, modem signals, etc. (via [IETF RFC 4733] real-time transport protocol (RTP) packet types). Figure 1 (the same as Figure 1 of [ITU-T T.38]) illustrates a model for facsimile transmission over IP networks. A more complete view of *example* media-type/-format specific configurations is illustrated in Figure 2.



NOTE – This figure is the same as Figure 1 of [ITU-T T.38].

Figure 1 – Model for facsimile transmission over IP networks

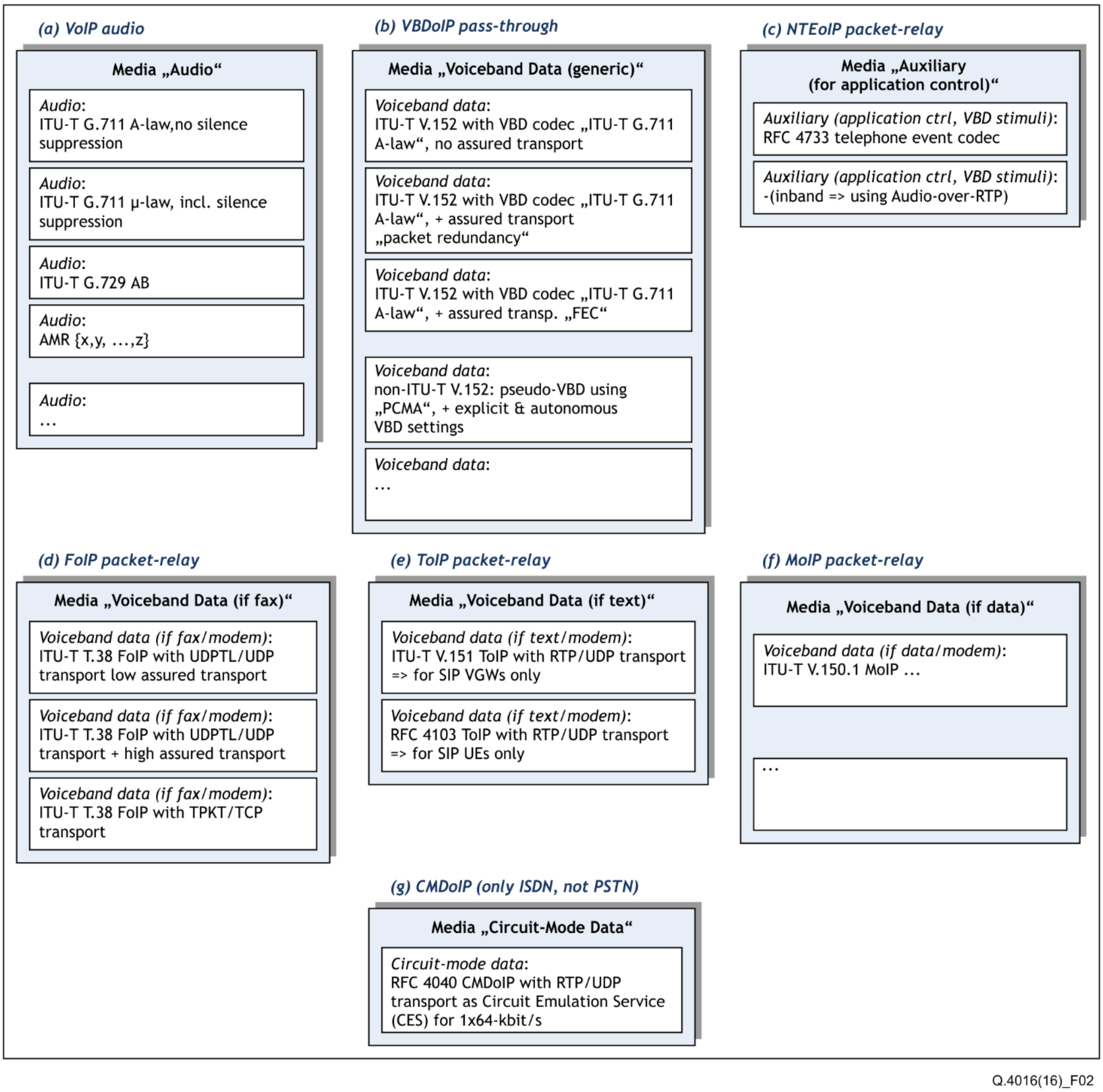


Figure 2 –Example media-type specific configurations

The possible scenario combinations depending on implemented standards are summarized in Table 1 (The numbers in the boxes refer to the scenarios explained above).

Table 1 – Mode transitioning support for ITU-T V.152 and ITU-T T.38 emulation services

<table>
<tr><th rowspan="2">Mode:</th><th colspan="2">ITU-T V.152</th><th colspan="2">ITU-T T.38</th></tr>
<tr><th>SIP-controlled ITU-T V.152 gateway</th><th>H.248-controlled ITU-T V.152 gateway</th><th>SIP-controlled ITU-T T.38 gateway (or Internet-aware fax (IAF))</th><th>H.248-controlled ITU-T T.38 gateway</th></tr>
<tr><td>Strict-controlled transitioning</td><td colspan="2">NO
Basically not addressed by [ITU-T V.152] (Note 1)</td><td>YES
(see clause D.2.2.4.2 of [ITU-T T.38])</td><td>YES
(see clause E.2.2.1 of [ITU-T T.38])</td></tr>
<tr><td>Autonomous transitioning</td><td>YES</td><td>YES</td><td>NO (for 2007 and prior versions of [ITU-T T.38]).
YES (for versions of [ITU-T T.38] after 2007 (Note 2);
see clause D.2.2.4.3 of [ITU-T T.38])</td><td>YES.
(see clause E.2.2.2 of [ITU-T T.38];
(Note 3))</td></tr>
<tr><td colspan="5">NOTE 1 – The revision of [ITU-T V.152] by ITU-T provides additional support of "strict-controlled transitioning", see clause 10.3 of [ITU-T V.152].
NOTE 2 – The latest revision of [ITU-T T.38] defines "autonomous transitioning" for SIP. This allows a facilitated operation between ITU-T T.38 endpoints with autonomous transitioning, independent of SIP or ITU-T H.248 control. SIP voice gateways (VGWs): autonomous transition in case of SIP-controlled ITU-T T.38 implies that both "controlling SIP UA" entities do support revised SDP Offer/Answer.
NOTE 3 – ITU-T H.248 GWs: autonomous transition in case of ITU-T H.248-controlled ITU-T T.38 implies support of the ITU-T H.248 ReserveGroup property. This signalling capability is not (yet) supported by all relevant ITU-T H.248 profiles from European Telecommunications Standards Institute (ETSI) TISPAN and 3GPP.
NOTE 4 – Table 1 is from [ETSI TR 183 072] Figure 5.</td></tr>
</table>

6 SIP signalling scenarios

6.1 SIP signalling scenarios structure

This clause provides the test suite structure (TSS) of this Recommendation.

Pseudo-VBDoIP emulation		
	Non-assured transport	
	1.1	ITU-T G.711 proposed and accepted, non-assured transport
	1.2	Successful changeover 1: ITU-T G.729 is accepted, ITU-T G.711 proposed; non-assured transport voiceband data over IP (VBDoIP) pass-through method
Autonomous transitioning with ITU-T V.152		
	Autonomous transitioning with ITU-T V.152, non-assured transport	
	2.1.1	Successful call with ITU-T V.152, non-assured transport
	2.1.2	Successful call with ITU-T V.152, assigned payload type (PT) with voiceband data (VBD), non-assured transport

	2.1.3	Successful call with ITU-T V.152, support of ITU-T V.152, ITU-T T.38, IETF RFC 3389 silence suppression, non-assured transport		
	2.1.4	Terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD codec. Non-assured transport.		
	2.1.5	The calling gateway A supports ITU-T V.152, ITU-T T.38, IETF RFC 3389 silence suppression, non-assured transport. The answering gateway prefers ITU-T T.38 above that of VBD for facsimile transmission. Non-assured transport.		
	2.1.6	The calling gateway A supports ITU-T V.152, ITU-T T.38. The answering gateway does supports ITU-T T.38 but does not support VBD. Non-assured transport.		
Autonomous transitioning with ITU-T V.152, assured transport packet redundancy				
	2.2.1	Successful call with ITU-T V.152, assured transport packet redundancy		
	2.2.2	Successful call with ITU-T V.152, assigned payload type with VBD, assured transport packet redundancy		
	2.2.3	Successful call with ITU-T V.152, support of ITU-T V.152, ITU-T T.38, IETF RFC 3389 silence suppression, assured transport packet redundancy		
	2.2.4	Terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD codec. Assured transport packet redundancy		
	2.2.5	The calling gateway A supports ITU-T V.152, ITU-T T.38, IETF RFC 3389 silence suppression, assured transport packet redundancy. The answering gateway prefers ITU-T T.38 above that of VBD for facsimile transmission.		
	2.2.6	The calling gateway A supports ITU-T V.152, ITU-T T.38. The answering gateway does supports ITU-T T.38 but does not support VBD. Assured transport packet redundancy.		
	ITU-T V.152 with VBD codec ITU-T G.711, assured transport "FEC"			
	2.3.1	Successful call with ITU-T V.152, assured transport "FEC"		
	2.3.2	Successful call with ITU-T V.152, assigned payload type with VBD, assured transport "FEC"		
	2.3.3	Successful call with ITU-T V.152, support of ITU-T V.152, ITU-T T.38, IETF RFC 3389 silence suppression, assured transport "FEC"		
	2.3.4	Terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD codec. Assured transport "FEC".		
	2.3.5	The calling gateway A supports ITU-T V.152, ITU-T T.38, IETF RFC 3389 silence suppression, assured transport "FEC". The answering gateway prefers ITU-T T.38 above that of VBD for facsimile transmission.		
	2.3.6	The calling gateway A supports ITU-T V.152, ITU-T T.38. The answering gateway does support ITU-T T.38 but does not support VBD. Assured transport "FEC".		
ITU-T T.38 autonomous transitioning method (ITU-T T.38 negotiated – auto switchover to ITU-T T.38)				
	3.1	Autonomous state transitioning between voice and facsimile; Fax is detected at the called party		
	3.2	Autonomous state transitioning between voice and facsimile; Fax is detected at the calling party		
Strict controlled transitioning (ITU-T T.38 – protocol-based switchover)				
		Fax detection at destination		
		4.1	Successful changeover	
			4.1.1	Successful changeover, only ITU-T G.711 offered and accepted, ITU-T T.38 proposed
			4.1.2	Successful changeover, ITU-T G.711 and ITU-T G.729 offered, ITU-T G.711 accepted, ITU-T T.38 proposed

			4.1.3	Successful changeover: ITU-T G.729 accepted, ITU-T T.38 and ITU-T G.711 proposed
			4.1.4	Successful changeover: ITU-T G.729 accepted, ITU-T T.38 only proposed
		4.2	Unsuccessful changeover	
				Unsuccessful changeover; case 1
			4.2.1	Unsuccessful changeover, case 1: ITU-T G.729 accepted, ITU-T G.711 and ITU-T T.38 proposed (200 OK response)
			Unsuccessful changeover, case 2	
			4.2.2	Unsuccessful changeover, case 2: ITU-T G.729 accepted, only ITU-T T.38 proposed (488 or 415 response / re-INVITE)
			4.2.3	Unsuccessful changeover, case 2: ITU-T G.711 accepted, only ITU-T T.38 proposed (488 or 415 response / re-INVITE)
			Unsuccessful changeover, case 3	
			4.2.4	Unsuccessful changeover, case 3: ITU-T G.711 accepted, ITU-T T.38 only proposed (488 or 415 response / re-INVITE)
		Fax detection at origin		
		4.3	Successful changeover	
			4.3.1	Successful changeover, case 1: ITU-T G.729 accepted, ITU-T T.38 proposed
			4.3.2	Successful changeover: ITU-T G.729 accepted, ITU-T T.38 and ITU-T G.711 proposed
			4.3.3	Successful changeover: ITU-T G.729 accepted, ITU-T T.38 only proposed
		4.4	Unsuccessful changeover	
				Unsuccessful changeover, case 1
			4.4.1	Unsuccessful changeover, case 1: ITU-T G.729 accepted, ITU-T G.711 and ITU-T T.38 proposed (200 OK response)
				Unsuccessful changeover, case 2
			4.4.2	Unsuccessful changeover, case 2: ITU-T G.729 accepted, only ITU-T T.38 proposed (488 or 415 response / re-INVITE)
			4.4.3	Unsuccessful changeover, case 2: ITU-T G.711 accepted, only ITU-T T.38 proposed (488 or 415 response / re-INVITE)
				Unsuccessful changeover, case 3
			4.4.4	Unsuccessful changeover, case 3: ITU-T G.711 accepted, ITU-T T.38 only proposed (488 or 415 response / re-INVITE)
		4.5	ITU-T T.38 offer in initial INVITE (IAF device)	

6.1.1 Configuration

The scope of the present Recommendation is to test the signalling and procedural aspects of the stage 3 requirements as described in [ITU-TQ.3403], [ITU-T T.38], [ITU-TV.152] and [ITU-T V.153]. The stage 3 description respects the requirements to several network entities and also to requirements regarding end devices. Therefore, several interfaces (reference points) are addressed to satisfy the testing of the different entities. Figure 3 illustrates the applicable Gm reference interface. Figure 4 illustrates the applicable Gm/Mw reference interfaces. Figure 5 illustrates the applicable configuration to test the user equipment as terminating and originating access device. Figure 6 illustrates applicable configuration to test the integrated access device (IAD), access gateway (AGW) and home gateway (HGW) as terminating and originating access device.

Therefore, to test the appropriate entities the configurations below are applicable:

IP multimedia subsystem network testing

In the case when the IP multimedia subsystem (IMS) network will be tested, the verification of several requirements is needed.

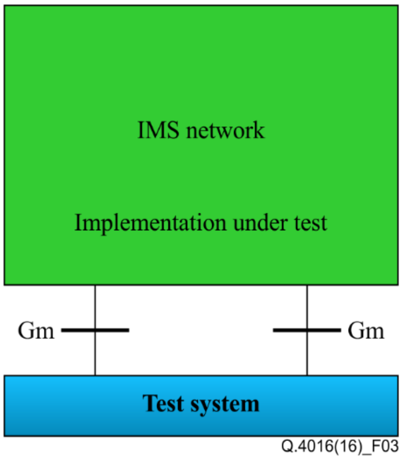
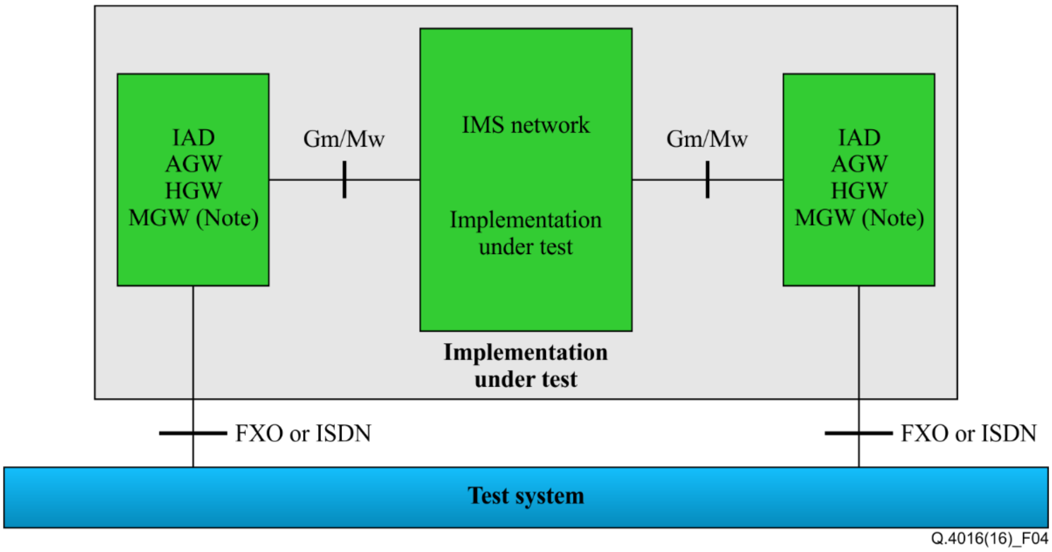


Figure 3 – Applicable Gm reference interface



NOTE – ITU-T H.248 residential gateway which provides here either the ITU-T T.38 or ITU-T V.152 interworking function.

Figure 4 – Applicable Gm/Mw reference interfaces

User terminal equipment testing

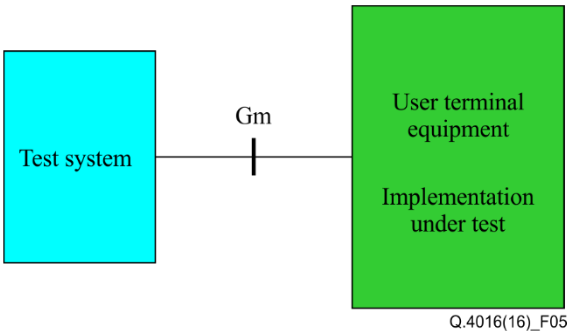


Figure 5 – Applicable configuration to test the user equipment as terminating and originating access device

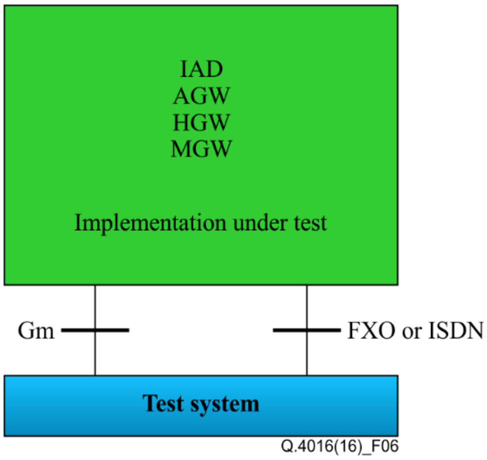


Figure 6 – Applicable configuration to test the IAD, AGW and HGW as terminating and originating access device

Possible "endpoints" are:

- the home gateway (HGW);
- the analogue terminal adapter (ATA);
- the integrated access device (IAD) for ISDN on IMS;
- the access gateway (AGW)/AGCF for PSTN replacement;
- the media gateway (MGW)/MGCF for PSTN/ISDN gateway;
- IP-PBXs for business trunking.

6.2 Pseudo-VBDoIP emulation

This clause provides the test cases for the Pseudo-VBDoIP emulation.

TSS	**1.1**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]	**Selection expression** PICS 1/2 AND 2/2
Scenario: ITU-T G.711 proposed and accepted, non-assured transport			
Signalling level: The *INVITE* contains SDP information regarding the proposed call voice. The terminating SIP user agent (UA) sends a *200 OK* response containing SDP information regarding the accepted ITU-T G.711 call parameters. **Media level:**			

Call with ITU-T G.711 Fax G3 options: G3 fax device (FD) to G3 FD ITU-T V.34 G3 FD to V.34 G3 FD ITU-T V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**
SDP header values:

A)		**INVITE SDP 1 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.>
	b)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> **a=maxmptime: PIXIT <list of packet times separated by space>** a=ptime: PIXIT
	c)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> a=maxmptime: PIXIT <list of packet times separated by space> a=maxmptime: PIXIT <list of packet times separated by space>
	d)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	f)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	g)	m=audio <port no.> RTP/AVP <PT assig.> <13> <PT assig.>
	h)	m=audio <port no.> RTP/AVP <PT assig.> <13> <PT assig.> a=ptime: PIXIT
B)		**INVITE SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] [13] [PT dyn] [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
Note: [optional parameters] < mandatory parameters **>**		

Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	SDP 1 (ITU-T G.711)	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	SDP 2 (ITU-T G.711)	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**1.2**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; **Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]**; Clause D.2.2.4.2of ITU-T T.38	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 PICS 9/2 PICS 10/2
Scenario: Successful changeover 1: ITU-T G.729 is accepted, ITU-T G.711 proposed; non-assured transport VBDoIP pass-through method			
Preconditions: The originating SIP GW supports ITU-T G.711			
Signalling level: In this scenario: – The originating SIP GW supports ITU-T G.729 and ITU-T G.711 – The call is set-up with a speech codec. (e.g., ITU-T G.729 and ITU-T G.711) – ITU-T G.729 is accepted			

– Fax is detected at called-party, initiating the re-INVITE proposing ITU-T G.711 – The originating SIP UA accepts ITU-T G.711, answers with 200 OK (ITU-T G.711) **Media level:** Call is switched from ITU-T G.729 mode to ITU-T G.711 mode Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**		
SDP header values:		
A)		**INVITE SDP 1 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <18> <13> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <18> <13> <PT assig.> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**INVITE SDP 2 (200 OK)**
		m=audio <port no.> RTP/AVP 18 [PT assig.] [13] <PT dyn> [a=maxmptime: PIXIT <list of packet times separated by space>] [a=ptime: PIXIT <list of packet times separated by space>] <a=rtpmap:<PT dyn.> telephone-event/8000> <a=fmtp:<PT dyn.> <events PIXIT >>
C)		**INVITE SDP 3 (INVITE)**
		m=audio <port no.> RTP/AVP <PT assig.> [18] [PT dyn] [a=ptime: PIXIT <list of packet times separated by space>] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
D)		**INVITE SDP 4 (200 OK)**
		m=audio <port no.> RTP/AVP <PT assig.> [18] [PT dyn] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=ptime: PIXIT <list of packet times separated by space>] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
Note: [optional parameters] < mandatory parameters >		

Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
Option A: Tone relay (see [IETF RFC 4733] – RTP payload for dual tone multi frequency (DTMF) digits, telephony tones and telephony signals)				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-63)	→	**RTP G.729**	→	
	←		←	**Data transmission detected (e.g., ANSam, /ANSam)**
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

Option B: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32kbit/s) over RTP/UDP/IP; Data transmission detected at the origination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-15)	→	**RTP G.729**	→	
Data transmission detected (e.g., CI, CT, calling tone (CNG))				
INVITE	→	**SDP 3**	→	**INVITE**
200 OK	←	**SDP 4**	←	**200 OK**
ACK	→		→	**ACK**
	→	**RTP G.711**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**
Option C: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32 kbit/s) over RTP/UDP/IP; Data transmission detected at the destination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-15)	→	**RTP G.729**	→	
				Data transmission detected (e.g., ANSam, /ANSam)
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

6.3 Autonomous transitioning with ITU-T V.152

6.3.1 Autonomous transitioning with ITU-T V.152, non-assured transport

This clause provides the test cases of autonomous transitioning with ITU-T V.152, non-assured transport.

<table>
<tr><td>TSS</td><td>TP 2.1.1</td><td colspan="2">Reference
Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1];
Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1];
Clause 7 of [ITU-T V.152];</td><td>Selection expression
PICS 1/2 AND 2/2 AND 3/2 AND 13/2;
PICS 9/2
PICS 10/2</td></tr>
<tr><td colspan="5">Scenario: Successful call with V.152, non-assured transport</td></tr>
<tr><td colspan="5">Preconditions:
The originating and terminating SIP GW supports V.152</td></tr>
<tr><td colspan="5">Signalling level:
In this scenario the terminating SIP UA accepts V.152, answers with 200 OK
Media level:
Fax G3 options:
G3 FD to G3 FD
V.34 G3 FD to V.34 G3 FD
V.34 G3 fallback to G3
The different QoS fax test-configurations with default parameters are contained in Table 5</td></tr>
<tr><td colspan="5">SDP header values:</td></tr>
<tr><td>A)</td><td></td><td colspan="3">SDP 1 (INVITE)</td></tr>
<tr><td></td><td>a)</td><td colspan="3">m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.>
a=maxmptime: PIXIT <list of packet times separated by space>
a=rtpmap: <PT dyn.> <encoding name>
a=gpmd: <PT dyn.> vbd=yes</td></tr>
<tr><td></td><td>b)</td><td colspan="3">m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.>
a=maxmptime: PIXIT <list of packet times separated by space>
a=rtpmap: <PT dyn.> telephone-event/8000
a=fmtp: <PT dyn.> <events PIXIT >
a=rtpmap: <PT dyn.> <encoding name>
a=gpmd: <PT dyn.> vbd=yes</td></tr>
<tr><td></td><td>c)</td><td colspan="3">m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.>
a=ptime: PIXIT
a=rtpmap: <PT dyn.> <encoding name>
a=gpmd: <PT dyn.> vbd=yes</td></tr>
<tr><td></td><td>d)</td><td colspan="3">m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.>
a=ptime: PIXIT
a=rtpmap: <PT dyn.> telephone-event/8000
a=fmtp: <PT dyn<events PIXIT >
a=rtpmap: <PT dyn.> <encoding name>
a=gpmd: <PT dyn.> vbd=yes</td></tr>
<tr><td></td><td>e)</td><td colspan="3">m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <13> <PT dyn.> <PT dyn.>
a=ptime: PIXIT
a=rtpmap: <PT dyn.> telephone-event/8000
a=fmtp: <PT dyn.> <events PIXIT >
a=rtpmap: <PT dyn.> <encoding name>
a=gpmd: <PT dyn.> vbd=yes</td></tr>
<tr><td>B)</td><td></td><td colspan="3">SDP 2 (200 OK)</td></tr>
<tr><td></td><td>a)</td><td colspan="3">m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.>
[a=ptime: PIXIT]
[a=maxmptime: PIXIT <list of packet times separated by space>]
[a=rtpmap: <PT dyn.> telephone-event/8000]
[a=fmtp: <PT dyn.> <events PIXIT >]
a=rtpmap: <PT dyn.> <encoding name>
a=gpmd: <PT dyn.> vbd=yes</td></tr>
</table>

Note: [optional parameters] < mandatory parameters >				
Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.1.2**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 13/2 PICS 9/2 PICS 10/2
Scenario: Successful call with V.152, assigned payload type with VBD, non-assured transport			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: In this scenario the terminating SIP UA accepts V.152, answers with 200 OK (V.152) The assigned payload type with VBD must not be used for carrying voice. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**INVITE SDP 1**	
	a)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> a=gpmd: <PT assig.> vbd=yes a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>]	
	b)	m=audio <port no.> RTP/AVP <PT assig.> [13] a=gpmd: PT assig. vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>]	
B)		**SDP 2 (200 OK)**	
	a)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> a=gpmd: <PT assig.> vbd=yes a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn. vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>]	
	b)	m=audio <port no.> RTP/AVP <PT assig.> [13] a=gpmd: <PT assig.> vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>]	
Note: [optional parameters] < mandatory parameters >			
Test configurations: AGW to AGW AGW to VGW VGW to VGW			
Comments			

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.1.3**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152];	PICS 1/2 AND 2/2 AND 3/2 AND 11/2 AND 13/2 PICS 9/2 PICS 10/2

Scenario: Successful call with V.152, support of V.152, T.38, IETF RFC 3389 silence suppression, non-assured transport

Preconditions:
The originating and terminating SIP GW supports V.152

Signalling level:
In this scenario the terminating SIP UA accepts V.152, answers with 200 OK (V.152)
A call is setup between gateway A and gateway B which supports V.152, T.38, IETF RFC 3389 silence suppression.
Media level:
Fax G3 options:
G3 FD to G3 FD
V.34 G3 FD to V.34 G3 FD
V.34 G3 fallback to G3
The different QoS fax test-configurations with default parameters are contained in Table 5

SDP header values:

A)		**INVITE SDP 1**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy
B)		**SDP 2 (200 OK)**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy

Note: [optional parameters]
< mandatory parameters >

Test configurations:
AGW to AGW
AGW to VGW
VGW to VGW

Comments

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	TP 2.1.4	Reference	Selection expression
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	PICS 1/2 AND 2/2 AND 3/2 AND 13/2 PICS 9/2 PICS 10/2
Scenario: Terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD codec. Non-assured transport.			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: The terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD coder, allowing subsequent bypass (passthrough) sessions if fax / modem signals are detected during the call. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			

SDP header values:

A)		**SDP 1 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >]
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> [13] <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >]a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes

Note: [optional parameters]
< mandatory parameters >

Test configurations:
AGW to AGW
AGW to VGW
VGW to VGW

Comments

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	TP 2.1.5	Reference	Selection expression
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	PICS 1/2 AND 2/2 AND 3/2 AND 13/2 PICS 9/2 PICS 10/2
Scenario: The calling gateway A supports V.152, T.38, IETF RFC 3389 silence suppression, non-assured transport. The answering gateway prefers T.38 above that of VBD for facsimile transmission. Non-assured transport.			
Preconditions: The originating SIP GW supports T.152. The terminating GW prefer T.38 above that of VBD for facsimile transmission.			
Signalling level: In this scenario the calling gateway A supports V.152, T.38 optional IETF RFC 3389 silence suppression, non-assured transport. The answering gateway prefers T.38 above that of VBD for facsimile transmission. A V.152 implementation that receives the above 'pmft' attribute, and is able to support both the relay mechanisms specified, shall include the same 'pmft' attribute in its response. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3			

The different QoS fax test-configurations with default parameters are contained in Table 5				
SDP header values:				
A)		**INVITE SDP 1**		
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy		
B)		**SDP 2 (200 OK)**		
	a)	a=group: FID 1 2 **a=pmft: T38** m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy		
Note: [optional parameters] < mandatory parameters >				
Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	TP 2.1.6	Reference	Selection expression
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	PICS 1/2 AND 2/2 AND 3/2 AND 13/2 PICS 9/2 PICS 10/2

Scenario: The calling gateway A supports V.152, T.38. The answering gateway does support T.38 but does not support VBD. Non-assured transport.

Preconditions:
The originating SIP GW supports T.152. The answering gateway does not support VBD.

Signalling level:
In this scenario the calling gateway A supports V.152, T.38 optional IETF RFC 3389 silence suppression, non-assured transport. The answering gateway does not support VBD.

Media level:
Fax G3 options:
G3 FD to G3 FD
V.34 G3 FD to V.34 G3 FD
V.34 G3 fallback to G3

The different QoS fax test-configurations with default parameters are contained in Table 5

SDP header values:

A)		**INVITE SDP 1**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=mid:1 a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy
B)		**SDP 2 (200 OK)**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=mid:1 m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy

Note: [optional parameters]
< mandatory parameters >

Test configurations:
AGW to AGW
AGW to VGW
VGW to VGW

Comments

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

6.3.2 Autonomous transitioning with ITU-T V.152, assured transport packet redundancy

This clause provides the test cases of autonomous transitioning with ITU-T V.152, assured transport packet redundancy.

TSS	**TP 2.2.1**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152];	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 14/2 PICS 9/2 PICS 10/2
Scenario: Successful call with V.152, assured transport packet redundancy			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: In this scenario the terminating SIP UA accepts V.152, answers with 200 OK **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**SDP 1 (INVITE)**	
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	
	b)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap: <PT dyn.> telephone-event/8000 a=fmtp: <PT dyn.> <events PIXIT > a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	
	c)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> a=ptime: PIXIT a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	
	d)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> <PT dyn.> a=ptime: PIXIT a=rtpmap: <PT dyn.> telephone-event/8000 a=fmtp: <PT dyn.> <events PIXIT > a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	
	e)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <13> <PT dyn.> <PT dyn.> <PT dyn.> a=ptime: PIXIT a=rtpmap: <PT dyn.> telephone-event/8000 a=fmtp: <PT dyn.> <events PIXIT >a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	
B)		**SDP 2 (200 OK)**	
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>]	

		[a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0		
Note: [optional parameters] < mandatory parameters >				
Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.2**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	PICS 1/2 AND 2/2 AND 3/2 AND 14/2 PICS 9/2 PICS 10/2
Scenario: Successful call with V.152, assigned payload type with VBD, assured transport packet redundancy			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: In this scenario the terminating SIP UA accepts V.152, answers with 200 OK (V.152) The assigned payload type with VBD must not be used for carrying voice. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**INVITE SDP 1**	
	a)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> <PT dyn.> a=gpmd: <PT assig.> vbd=yes a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	
	b)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> a=gpmd: PT assig. vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	
B)		**SDP 2 (200 OK)**	
	a)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> <PT dyn.> a=gpmd: <PT assig.> vbd=yes a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn. vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	

	b)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> a=gpmd: <PT assig.> vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0		
Note: [optional parameters] < mandatory parameters >				
Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.3**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	PICS 1/2 AND 2/2 AND 3/2 AND 11/2 14/2 PICS 9/2 PICS 10/2
Scenario: Successful call with V.152, support of V.152, T.38, IETF RFC 3389 silence suppression, assured transport packet redundancy			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: In this scenario the terminating SIP UA accepts V.152, answers with 200 OK (V.152) A call is setup between gateway A and gateway B which supports V.152, T.38, IETF RFC 3389 silence suppression. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**INVITE SDP 1**	
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0 m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy	

B)		**SDP 2 (200 OK)**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0 m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy
Note: [optional parameters] < mandatory parameters >		
Test configurations: AGW to AGW AGW to VGW VGW to VGW		
Comments		

INVITE	➔	**SDP 1**	➔	**INVITE**
180 Ringing	🡐		🡐	**180 Ringing**
200 OK (INVITE)	🡐	**SDP 2**	🡐	**200 OK (INVITE)**
ACK	➔		➔	**ACK**
BYE	➔		➔	**BYE**
200 OK (BYE)	🡐		🡐	**200 OK (BYE)**

TSS	**TP 2.2.4**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 14/2; PICS 9/2 PICS 10/2
Scenario: Terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD codec. Assured transport packet redundancy.			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: The terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD coder, allowing subsequent bypass (passthrough) sessions if fax / modem signals are detected during the call. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			

A)		SDP 1 (INVITE)
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >]
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0
Note: [optional parameters] < mandatory parameters >		

Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.5**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 14/2 PICS 9/2 PICS 10/2
Scenario: The calling gateway A supports V.152, T.38, IETF RFC 3389 silence suppression, assured transport packet redundancy. The answering gateway prefers T.38 above that of VBD for facsimile transmission.			
Preconditions: The originating SIP GW supports T.152. The terminating GW prefer T.38 above that of VBD for facsimile transmission.			
Signalling level: In this scenario the calling gateway A supports V.152, T.38 optional IETF RFC 3389 silence suppression, assured transport packet redundancy. The answering gateway prefer T.38 above that of VBD for facsimile transmission. A V.152 implementation that receives the above 'pmft' attribute, and is able to support both the relay mechanisms specified, shall include the same 'pmft' attribute in its response. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			

A)		**INVITE SDP 1**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=mid:1 a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0 m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy
B)		**SDP 2 (200 OK)**
	a)	a=group: FID 1 2 **a=pmft: T38** m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=mid:1 a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0 m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF

		a=T38FaxUdpEC:t38UDPRedundancy		
Note: [optional parameters] < mandatory parameters >				
Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.6**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 14/2 PICS 9/2 PICS 10/2
Scenario: The calling gateway A supports V.152, T.38. The answering gateway does supports T.38 but does not support VBD. Assured transport packet redundancy.			
Preconditions: The originating SIP GW supports T.152. The answering gateway does not support VBD.			
Signalling level: In this scenario the calling gateway A supports V.152, T.38, optional IETF RFC 3389 silence suppression, assured transport packet redundancy. The answering gateway does not support VBD. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**INVITE SDP 1**	
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0 m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy	
B)		**SDP 2 (200 OK)**	
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy	
Note: [optional parameters] < mandatory parameters >			

Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

6.3.3 Autonomous transitioning with ITU-T V.152, assured transport "FEC"

This clause provides the test cases of autonomous transitioning with ITU-T V.152, assured transport forward error correction (FEC).

TSS	**TP 2.2.1**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 15/2; PICS 9/2 PICS 10/2
Scenario: Successful call with V.152, assured transport "FEC"			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: In this scenario the terminating SIP UA accepts V.152, answers with 200 OK **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**SDP 1 (INVITE)**	
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx	
	b)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap: <PT dyn.> telephone-event/8000 a=fmtp: <PT dyn.> <events PIXIT > a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx	
	c)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> a=ptime: PIXIT a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0	
	d)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> <PT dyn.> a=ptime: PIXIT a=rtpmap: <PT dyn.> telephone-event/8000 a=fmtp: <PT dyn.> <events PIXIT > a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx	
	e)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <13> <PT dyn.> <PT dyn.> <PT dyn.> a=ptime: PIXIT	

		a=rtpmap: <PT dyn.> telephone-event/8000 a=fmtp: <PT dyn.> <events PIXIT > a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx
Note: [optional parameters] < mandatory parameters >		
Test configurations: AGW to AGW AGW to VGW VGW to VGW		
Comments		

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.2**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 15/2 PICS 9/2 PICS 10/2
Scenario: Successful call with V.152, assigned payload type with VBD, assured transport "FEC"			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: In this scenario the terminating SIP UA accepts V.152, answers with 200 OK (V.152) The assigned payload type with VBD must not be used for carrying voice. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			

A)		**INVITE SDP 1**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> <PT dyn.> a=gpmd: <PT assig.> vbd=yes a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx
	b)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> a=gpmd: PT assig. vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx
B)		**SDP 2 (200 OK)**

	a)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> <PT dyn.> a=gpmd: <PT assig.> vbd=yes a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn. vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap:<PT dyn.> red/8000 a=fmtp:<PT dyn.> 0/0/0/0
	b)	m=audio <port no.> RTP/AVP <PT assig.> [13] <PT dyn.> a=gpmd: <PT assig.> vbd=yes [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx
Note: [optional parameters] < mandatory parameters >		
Test configurations: AGW to AGW AGW to VGW VGW to VGW		
Comments		

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.3**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 15/2 PICS 9/2 PICS 10/2
Scenario: Successful call with V.152, support of V.152, T.38, IETF RFC 3389 silence suppression, assured transport "FEC"			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: In this scenario the terminating SIP UA accepts V.152, answers with 200 OK (V.152) A call is setup between Gateway A and gateway B which supports V.152, T.38, IETF RFC 3389 silence suppression. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**INVITE SDP 1**	
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy	

B)		SDP 2 (200 OK)
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy
Note: [optional parameters] < mandatory parameters >		
Test configurations: AGW to AGW AGW to VGW VGW to VGW		

Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.4**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 15/2 PICS 9/2 PICS 10/2
Scenario: Terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD codec, assured transport "FEC"			
Preconditions: The originating and terminating SIP GW supports V.152			
Signalling level: The terminating user on an INVITE with an SDP offer containing a codec without VBD, sends an SDP answer containing a VBD coder, allowing subsequent bypass (passthrough) sessions if fax / modem signals are detected during the call. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax fest-configurations with default parameters are contained in Table 5**			
SDP header values:			

A)		SDP 1 (INVITE)
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >]
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx
Note: [optional parameters] < mandatory parameters >		

Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.5**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 15/2 PICS 9/2 PICS 10/2
Scenario: The calling gateway A supports V.152, T.38, IETF RFC 3389 silence suppression, assured transport "FEC". The answering gateway prefers T.38 above that of VBD for facsimile transmission.			
Preconditions: The originating SIP GW supports T.152. The terminating GW prefers T.38 above that of VBD for facsimile transmission.			
Signalling level: In this scenario the calling gateway A supports V.152, T.38 optional IETF RFC 3389 silence suppression, assured transport packet redundancy. The answering gateway prefers T.38 above that of VBD for facsimile transmission. A V.152 implementation that receives the above 'pmft' attribute, and is able to support both the relay mechanisms specified, shall include the same 'pmft' attribute in its response. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			

A)		**INVITE SDP 1**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=mid:1 a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy
B)		**SDP 2 (200 OK)**
	a)	a=group: FID 1 2 **a=pmft: T38** m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=mid:1 a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy

Note: [optional parameters] < mandatory parameters >				
Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 2.2.6**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause 7 of [ITU-T V.152]	**Selection expression** PICS 1/2 AND 2/2 AND 3/2 AND 15/2 PICS 9/2 PICS 10/2
Scenario: The calling gateway A supports V.152, T.38. The answering gateway does supports T.38 but does not support VBD, assured transport "FEC".			
Preconditions: The originating SIP GW supports T.152. The answering gateway does not support VBD.			
Signalling level: In this scenario the calling gateway A supports V.152, T.38, optional IETF RFC 3389 silence suppression, assured transport packet redundancy. The answering gateway does not support VBD. **Media level:** Fax G3 options: G3 FD to G3 FD V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax fest-configurations with default parameters are contained in Table 5**			
SDP header values:			

A)		**INVITE SDP 1**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] a=rtpmap: <PT dyn.> <encoding name> a=gpmd: <PT dyn.> vbd=yes a=rtpmap: <PT unassig> parityfec/8000 a=fmtp: <PT unassig> <port no.> IN IP4 xxx.xxx.xxx.xxx m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy
B)		**SDP 2 (200 OK)**
	a)	a=group: FID 1 2 m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> [13] <PT dyn.> a=mid:1 [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap: <PT dyn.> telephone-event/8000] [a=fmtp: <PT dyn.> <events PIXIT >] m=image <port no.> udptl t38 a=mid:2 a=T38version: PIXIT a=T38FaxRateManagement:transferredTCF a=T38FaxUdpEC:t38UDPRedundancy
Note: [optional parameters] < mandatory parameters >		

Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

6.4 Autonomous state transitioning between voice and facsimile

This clause provides the test cases of autonomous state transitioning between voice and facsimile.

TSS	**TP 3.1**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.3 of ITU-T T.38; Clause E.2.2.2 of ITU-T T.38	**Selection expression** PICS 21/2
Scenario: Autonomous state transitioning between voice and facsimile; Fax is detected at the **called-party**			
Preconditions: Autonomous state transitioning shall only be possible under the following conditions: • local and remote side shall both support ITU-T T.38; and • local and remote side shall agree beforehand on the particular ITU-T T.38 configuration (e.g., ITU-T T.38 transport variant, ITU-T T.38 parameter settings). Both conditions may be easily addressed by the usage of *Revised* SDP Offer/Answer in SIP (see clause D.2.3.0 of ITU-T T.38): ITU-T T.38 shall be offered as *latent configuration*. A positive response by the *Answerer* side may permit then for autonomous state transitioning. Note – "ITU-T T.38 V.34G3" (see clause 3.5 of [ITU-T T.38 (2010)]) REQUIRES EXPLICIT support of ITU-T T.38 Version 4, which must be explicitly negotiated with SDP Offer/Answer at SIP level In this scenario, ITU-T T.38 capability is negotiated during connection setup using two "m" lines in SDP Offer/Answer. When defined T.38 stimuli are detected, autonomous switchover takes place. The redundancies, as well as FEC error-correction modes, are possible. In this case, the bandwidth can be smaller and the sensitivity to packet loss is lower.			
Signalling level: In this scenario: – The call is set-up with a speech codec (e.g., ITU-T G.729, ITU-T G.711) and ITU-T T.38 **Media level:** Call is switched from the voice mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			

		SDP 1 (INVITE)
	a)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> a=maxmptime: PIXIT <list of packet times separated by space>

		m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	c)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> a=ptime: PIXIT m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	d)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	e)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> a=ptime: PIXIT m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
		SDP 4 (200 OK)
	a)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT>] **[**a=maxmptime: PIXIT <list of packet times separated by space>] **[**a=ptime: PIXIT <list of packet times separated by space>] m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
Note: [optional parameters] < mandatory parameters >		
Test configurations: MGW-MGW		

Comments				
INVITE	→	**SDP 1 (G.711, G.729; T.38)**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2 (G.711, G.729; T.38)**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**Voice Mode**	→	
	←		←	**Fax detected**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

<table>
<tr><td>TSS</td><td>TP 3.2</td><td>Reference
Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1];
Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1];
Clause D.2.2.4.3 of [ITU-T T.38];
Clause E.2.2.2 of [ITU-T T.38]</td><td>Selection expression
PICS 21/2</td></tr>
<tr><td colspan="4">Scenario: Autonomous state transitioning between voice and facsimile; Fax is detected at the calling party</td></tr>
<tr><td colspan="4">Preconditions:
Autonomous state transitioning shall only be possible under the following conditions:
• local and remote side shall both support [ITU-T T.38]; and
• local and remote side shall agree beforehand on the particular ITU-T T.38 configuration (e.g., [ITU-T T.38] transport variant, [ITU-T T.38] parameter settings).
Both conditions may be easily addressed by the usage of Revised SDP Offer/Answer in SIP (see clause D.2.3.0 of ITU-T T.38): [ITU-T T.38] shall be offered as latent configuration. A positive response by the Answerer side may permit then for autonomous state transitioning.
Note – "T.38 V.34G3" (see clause 3.5 of [ITU-T T.38 (2010)]) REQUIRES EXPLICIT support of ITU-T T.38 Version 4, which must be explicitly negotiated with SDP Offer/Answer at SIP level.
In this scenario, T.38 capability is negotiated during connection setup using two "m" lines in SDP Offer/Answer. When defined T.38 stimuli are detected, autonomous switchover takes place. The redundancies, as well as FEC error-correction modes, are possible. In this case, the bandwidth can be smaller and the sensitivity to packet loss is lower.</td></tr>
<tr><td colspan="4">Signalling level:
In this scenario:
– The call is set-up with a speech codec (e.g., ITU-T G.729, ITU-T G.711) and ITU-T T.38
Media level:
Call is switched from the voice mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode.
Fax G3 options:
G3 FD to G3 FD
ITU-T V.34 G3 FD to ITU-T V.34 G3 FD
ITU-T V.34 G3 fallback to G3
The different QoS fax test-configurations with default parameters are contained in Table 5</td></tr>
<tr><td colspan="4">SDP header values:</td></tr>
</table>

		SDP 1 (INVITE)
	a)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150

	b)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> a=maxmptime: PIXIT <list of packet times separated by space> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	c)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> a=ptime: PIXIT m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	d)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	e)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> a=ptime: PIXIT m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
		SDP 4 (200 OK)
	a)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] <PT dyn.> [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT>] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=ptime: PIXIT <list of packet times separated by space>] m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150

Note: [optional parameters] < mandatory parameters >				
Test configurations: MGW-MGW				
Comments				
INVITE	→	**SDP 1 (G.711, G.729, T.38)**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2 (G.711, G.729, T.38)**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**Voice Mode**	→	
Fax detected	→		→	
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

6.5 ITU-T T.38 strict controlled transitioning (ITU-T T.38 – protocol-based switchover)

6.5.1 Fax detection at the destination

This clause provides the test cases for fax detection at the destination.

General comment:

"ITU-T T.38 V.34G3" (see clause 3.5 of [ITU-T T.38 (2010)]) REQUIRES EXPLICIT support of ITU-T T.38 Version 4, which must be explicitly negotiated with SDP Offer/Answer at SIP level.

TSS	**TP 4.1.1.1**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Successful changeover, only ITU-T G.711 offered and accepted, ITU-T T.38 proposed			
Preconditions: The originating SIP GW supports ITU-T T.38			
Signalling level: In this scenario: – The call is set-up with a speech codec (e.g., ITU-T G.711) – ITU-T G.711 is accepted – Fax is detected at **called-party**, initiating the re-INVITE proposing ITU-T T.38 – The originating SIP UA accepts ITU-T T.38, answers with 200 OK (ITU-T T.38) **Media level:** Call is switched from ITU-T G.711 mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 (see ˜7) **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			

A)		**INVITE SDP 1 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.>
	b)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> a=maxmptime: PIXIT <list of packet times separated by space> a=ptime: PIXIT
	c)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> a=maxmptime: PIXIT <list of packet times separated by space> a=maxmptime: PIXIT <list of packet times separated by space>
	d)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> a=ptime: PIXIT

		a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	f)	m=audio <port no.> RTP/AVP <PT assig.> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	g)	m=audio <port no.> RTP/AVP <PT assig.> <13> <PT assig.>
	h)	m=audio <port no.> RTP/AVP <PT assig.> <13> <PT assig.> a=ptime: PIXIT
B)		**INVITE SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [PT assig.] [13] [PT dyn] [a=ptime: PIXIT] [a=maxmptime: PIXIT <list of packet times separated by space>] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
		SDP 3 (INVITE)
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
C)		**SDP 4 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150

Note: [optional parameters] < mandatory parameters >				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**RTP G.711**	→	
	←		←	**Fax transmission detected (ITU-T V.21 flags)**
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.1.1.2**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Successful changeover, ITU-T G.711 and ITU-T G.729 offered, ITU-T G.711 accepted, ITU-T T.38 proposed			
Preconditions: The originating SIP GW supports ITU-T T.38			
Signalling level: In this scenario: – The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729) – ITU-T G.711 is accepted – Fax is detected at **called-party**, initiating the re-INVITE proposing ITU-T T.38 – The originating SIP UA accepts ITU-T T.38, answers with 200 OK (ITU-T T.38) **Media level:** Call is switched from ITU-T G.711 mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 (see Figure 7) **The different QoS fax test-configurations with default parameters are contained in Table 5**			

SDP header values:		
A)		**INVITE SDP 1**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [13] [18] [dyn.assign] [a=ptime: PIXIT] < a=rtpmap:<PT dyn.> telephone-event/8000> <a=fmtp:<PT dyn.> <events PIXIT >>
		SDP 3 (INVITE)

	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
C)		**SDP 4 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
Note: [optional parameters] < mandatory parameters >		

Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**RTP G.711**	→	
	←		←	**Fax transmission detected (ITU-T V.21 flags)**
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.1.2**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Successful changeover: ITU-T G.729 accepted, ITU-T T.38 and ITU-T G.711 proposed			
Preconditions: The originating SIP GW supports ITU-T T.38			

Signalling level: In this scenario: – The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729) – ITU-T G.729 accepted – Fax is detected at the called-party, initiating the re-INVITE proposing both ITU-T T.38 and ITU-T G.711 – The originating SIP UA accepts ITU-T T.38, answers with 200 OK (ITU-T T.38) **Media level:** Call is switched from ITU-T G.729 mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 (see Figure 7) **The different QoS fax test-configurations with default parameters are contained in Table 5**		
SDP header values:		
A)		**INVITE SDP 1**
	a)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <18> [13] [PT assig.] <dyn.assign> [a=ptime: PIXIT] <a=rtpmap:<PT dyn.> telephone-event/8000> <a=fmtp:<PT dyn.> <events PIXIT >
C)		**INVITE SDP 3**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18]
	b)	m=audio <port no.> RTP/AVP <PT assig.> [18] a=ptime: PIXIT
	c)	m=audio <port no.> RTP/AVP <PT assig.> [18] <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
D)		**SDP 4 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18] [dyn.assign] [a=ptime: PIXIT] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
E)		**SDP 5 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> [a=ptime: PIXIT] m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> [a=ptime: PIXIT]

		m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150		
F)		**SDP 6 (200 OK)**		
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150		
	b)	m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150		
Note: [optional parameters] < mandatory parameters >				
Comments				
Option A: Tone relay (see [IETF RFC 4733] – RTP payload for DTMF digits, telephony tones and telephony signals).				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-63)	→	**RTP G.729**	→	
	←		←	**Fax transmission detected (ITU-T V.21 flags)**
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**
ACK	←		←	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**
Option B: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32 kbit/s) over RTP/UDP/IP; Data transmission detected at the origination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-15)	→	**RTP G.729**	→	
Data transmission detected (e.g., CI, CT, CNG)				
INVITE	→	**SDP 3**	→	**INVITE**
200 OK	←	**SDP 4**	←	**200 OK**
ACK	→		→	**ACK**
	→	**RTP G.711**	→	

				Fax transmission detected (ITU-T V.21 flags)
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**
ACK	←		←	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**
Option C: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32 kbit/s) over RTP/UDP/IP; Data transmission detected at the destination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-15)	→	**RTP G.729**	→	
				Data transmission detected (e.g., ANSam, /ANSam)
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
				Fax transmission detected (ITU-T V.21 flags)
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**
ACK	←		←	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.1.3**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Successful changeover: G.729 accepted, T.38 only proposed			
Preconditions: The originating SIP GW supports T38			
Scenario **Signalling level:** In this scenario: The call is set-up with a speech codec. (e.g., ITU-T G.711 and ITU-T G.729) ITU-T G.729 accepted Fax is detected at **called-party**, initiating the re-INVITE proposing **only** ITU-T T.38 The originating SIP UA accepts ITU-T T.38, answers with 200 OK (ITU-T T.38) **Media level:** Call is switched from ITU-T G.711 mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 (see Figure 7) **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**INVITE SDP 1**	

	a)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <18> [13] [PT assig.] <dyn.assign> [a=ptime: PIXIT] <a=rtpmap:<PT dyn.> telephone-event/8000> <a=fmtp:<PT dyn.> <events PIXIT >
C)		INVITE SDP 3
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18]
	b)	m=audio <port no.> RTP/AVP <PT assig.> [18] a=ptime: PIXIT
	c)	m=audio <port no.> RTP/AVP <PT assig.> [18] <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
D)		**SDP 4 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18] [dyn.assign] [a=ptime: PIXIT] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
E)		**SDP 5 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
F)		**SDP 6 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150

	b)	m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150		
Note: [optional parameters] < mandatory parameters >				
Comments:				
Option A: Tone relay (see [IETF RFC 4733] – RTP payload for DTMF digits, telephony tones and telephony signals).				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-63)	→	**RTP G.729**	→	
	←		←	**Fax transmission detected (ITU-T V.21 flags)**
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**
ACK	←		←	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**
Option B: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using ITU-T G.711 or [ITU-T G.726] (32 kbit/s) over RTP/UDP/IP; Data transmission detected at the origination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-15)	→	**RTP G.729**	→	
Data transmission detected (e.g., CI, CT, CNG)				
INVITE	→	**SDP 3**	→	**INVITE**
200 OK	←	**SDP 4**	←	**200 OK**
ACK	→		→	**ACK**
	→	**RTP G.711**	→	
				Fax transmission detected (ITU-T V.21 flags)
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**
ACK	←		←	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**
Option C: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32 kbit/s) over RTP/UDP/IP; Data transmission detected at the destination				

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-15)	→	**RTP G.729**	→	
				Data transmission detected (e.g., ANSam, /ANSam)
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
				Fax transmission detected (ITU-T V.21 flags)
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**
ACK	←		←	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.2.1**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Unsuccessful changeover, case 1: ITU-T G.729 accepted, ITU-T G.711 and ITU-T T.38 proposed (200 OK response)			
Preconditions: The originating SIP UA does not support ITU-T T.38			
Signalling level: In this scenario: The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729) ITU-T G.729 accepted Fax is detected at called-party, initiating the re-INVITE proposing ITU-T G.711 and ITU-T T.38 The originating SIP UA rejects the ITU-T T.38 offer, answers with 200 OK ITU-T G.711 **Media level:** ITU-T T.30 negotiations and image transmission are performed in ITU-T T.30 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 (see Figure 7) **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**SDP 1 (INVITE)**	

	a)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >

B)	**SDP 2 (200 OK)**	
	a)	m=audio <port no.> RTP/AVP <18> [13] [PT assig.] <dyn.assign> [a=ptime: PIXIT] <a=rtpmap:<PT dyn.> telephone-event/8000> <a=fmtp:<PT dyn.> <events PIXIT >>
C)		INVITE SDP 3
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18]
	b)	m=audio <port no.> RTP/AVP <PT assig.> [18] a=ptime: PIXIT
	c)	m=audio <port no.> RTP/AVP <PT assig.> [18] <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
D)		SDP 4 (200 OK)
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18] [dyn.assign] [a=ptime: PIXIT] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
E)		**SDP 5 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
F)		**SDP 6 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT>
	b)	**m=audio <port no.> RTP/AVP <PT assig.>**

Note: [optional parameters] < mandatory parameters >				
Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
Option A: Tone relay (see [IETF RFC 4733] – RTP payload for DTMF digits, telephony tones and telephony signals).				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-63)	→	**RTP G.729**	→	
	←		←	**Fax transmission detected (ITU-T V.21 flags)**
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**
Option B: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32 kbit/s) over RTP/UDP/IP; Data transmission detected at the **origination;**				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-15)	→	**RTP G.729**	→	
Data transmission detected (e.g., CI, CT, CNG)				
INVITE	→	**SDP 3**	→	**INVITE**
200 OK	←	**SDP 4**	←	**200 OK**
ACK	→		→	**ACK**
	→	**RTP G.711**	→	
				Fax transmission detected (ITU-T V.21 flags)
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**
Option C: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32 kbit/s) over RTP/UDP/IP; Data transmission detected at the destination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-15)	→	**RTP G.729**	→	
				Data transmission detected (e.g., ANSam, /ANSam)
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
				Fax transmission detected (ITU-T V.21 flags)
INVITE	←	**SDP 5**	←	**INVITE**
200 OK	→	**SDP 6**	→	**200 OK**

ACK	←		←	**ACK**
	→	**RTP G.711**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

<table>
<tr><td colspan="2">TSS</td><td>TP 4.2.2</td><td>Reference
Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1];
Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1];
Clause D.2.2.4.2 of [ITU-T T.38]</td><td>Selection expression
PICS 1/2
PICS 2/2
PICS 3/2
PICS 9/2
PICS 10/2
PICS 22/2</td></tr>
<tr><td colspan="5">Scenario: Unsuccessful changeover, case 2: ITU-T G.729 accepted, only ITU-T T.38 proposed (488 or 415 response / re-INVITE)</td></tr>
<tr><td colspan="5">Preconditions:
The originating SIP UA does not support ITU-T T.38</td></tr>
<tr><td colspan="5">Signalling level:
In this scenario:
- The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729)
- ITU-T G.729 accepted
- Fax is detected at called-party, initiating the re-INVITE proposing ITU-T T.38
- The originating SIP UA answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type)
- The called party sends a re-INVITE G.711 in order to fallback to ITU-T G.711 for fax transmission
Media level:
ITU-T T.30 negotiations and image transmission are performed in T.30 mode.
Fax G3 options:
G3 FD to G3 FD
ITU-T V.34 G3 FD to ITU-T V.34 G3 FD
ITU-T V.34 G3 fallback to G3
The different QoS fax test-configurations with default parameters are contained in Table 5</td></tr>
<tr><td colspan="5">SDP header values:</td></tr>
<tr><td>A)</td><td></td><td colspan="3">SDP 1 (INVITE)</td></tr>
<tr><td></td><td>a)</td><td colspan="3">m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=maxmptime: PIXIT <list of packet times separated by space>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>b)</td><td colspan="3">m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>c)</td><td colspan="3">m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>d)</td><td colspan="3">m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>e)</td><td colspan="3">m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
</table>

B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <18> [13] [PT assig.] <dyn.assign> [a=ptime: PIXIT] <a=rtpmap:<PT dyn.> telephone-event/8000> <a=fmtp:<PT dyn.> <events PIXIT >>
C)		**SDP 3 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
D)		**SDP 4 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT>
	b)	m=audio <port no.> RTP/AVP <PT assig.>
E)		**SDP 5 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT>
	b)	m=audio <port no.> RTP/AVP <PT assig.>
Note: [optional parameters] < mandatory parameters >		
Test configurations: AGW to AGW AGW to VGW VGW to VGW		
Comments		
Tone relay (see [IETF RFC 4733] – RTP payload for DTMF digits, telephony tones and telephony signals).		

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-63)	→	**RTP G.729**	→	
	←		←	**Fax transmission detected (ITU-T V.21 flags)**
INVITE	←	**SDP 3**	←	**INVITE**
488 (or 415)	→		→	**488 (or 415)**
ACK	←		←	**ACK**
INVITE	←	**SDP 4**	←	**INVITE**
200 OK (INVITE)	→	**SDP 5**	→	**200 OK (INVITE)**
ACK	←		←	**ACK**
		RTP G.711		
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

<table>
<tr><td>TSS</td><td>TP 4.2.3</td><td>Reference
Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1];
Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1];
Clause D.2.2.4.2 of [ITU-T T.38]</td><td>Selection expression
PICS 1/2
PICS 2/2
PICS 3/2
PICS 9/2
PICS 10/2
PICS 22/2</td></tr>
<tr><td colspan="4">Scenario: Unsuccessful changeover, case 2: ITU-T G.711 accepted, only ITU-T T.38 proposed (488 or 415 response / re-INVITE)</td></tr>
<tr><td colspan="4">Preconditions:
The originating SIP UA does not support ITU-T T.38</td></tr>
<tr><td colspan="4">Signalling level:
In this scenario:
– The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729)
– ITU-T G.711 accepted
– Fax is detected at called-party, initiating the re-INVITE proposing ITU-T T.38
– The originating SIP UA answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type)
– The called party sends a re-INVITE G.711 in order to fallback to G.711 for fax transmission
Media level:
ITU-T T.30 negotiations and image transmission are performed in ITU-T T.30 mode.
Fax G3 options:
G3 FD to G3 FD
ITU-T V.34 G3 FD to ITU-T V.34 G3 FD
ITU-T V.34 G3 fallback to G3
The different QoS fax test-configurations with default parameters are contained in Table 5</td></tr>
<tr><td colspan="4">SDP header values:</td></tr>
<tr><td>A)</td><td></td><td colspan="2">SDP 1 (INVITE)</td></tr>
<tr><td></td><td>a)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.>
a=maxmptime: PIXIT <list of packet times separated by space>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>b)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>c)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>d)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>e)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td>B)</td><td></td><td colspan="2">SDP 2 (200 OK)</td></tr>
<tr><td></td><td>a)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> [13] [18] [dyn.assign]
[a=ptime: PIXIT]
[a=rtpmap:<PT dyn.> telephone-event/8000]
[a=fmtp:<PT dyn.> <events PIXIT >]</td></tr>
<tr><td>C)</td><td></td><td colspan="2">SDP 3 (INVITE)</td></tr>
<tr><td></td><td>a)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT>
m=image <port no.> udptl t38
b=AS:46
a=T38FaxVersion: PIXIT
a=T38FaxRateManagement:PIXIT
a=T38MaxBitRate: PIXIT
a=T38FaxUdpEC:PIXIT
a=T38FaxMaxBuffer:1800
a=T38FaxMaxDatagram:150</td></tr>
<tr><td></td><td>b)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.>
m=image <port no.> udptl t38
b=AS:46
a=T38FaxVersion: PIXIT
a=T38FaxRateManagement:PIXIT
a=T38MaxBitRate: PIXIT
a=T38FaxUdpEC:PIXIT</td></tr>
</table>

		a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
D)		**SDP 4 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT>]
	b)	m=audio <port no.> RTP/AVP <PT assig.>
E)		**SDP 5 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT>]
	b)	m=audio <port no.> RTP/AVP <PT assig.>
Note: [optional parameters] < mandatory parameters >		
Test configurations: AGW to AGW AGW to VGW VGW to VGW		
Comments		

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**RTP G.711**	→	
	←		←	**Fax transmission detected (ITU-T V.21 flags)**
INVITE	←	**SDP 3**	←	**INVITE**
488 (or 415)	→		→	**488 (or 415)**
ACK	←		←	**ACK**
INVITE	←	**SDP 4**	←	**INVITE**
200 OK (INVITE)	→	**SDP 5**	→	**200 OK (INVITE)**
ACK	←		←	**ACK**
		RTP G.711		
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.2.4**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Unsuccessful changeover, case 3: ITU-T G.711 accepted, ITU-T T.38 only proposed (488 or 415 response / re-INVITE)			
Preconditions: The originating SIP UA does not support ITU-T T.38			
Scenario **Signalling level:** – The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729) – ITU-T G.711 accepted – Fax is detected at called-party, initiating the re-INVITE proposing ITU-T T.38 – The originating SIP UA answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type) – The called party sends a ACK in order to fallback to previously negotiated PT assig. **Media level:** ITU-T T.30 negotiations and image transmission are performed in ITU-T T.30 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**SDP 1 (INVITE)**	

	a)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [13] [18] [dyn.assign] [a=ptime: PIXIT] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
C)		**SDP 3 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
Test configurations: AGW to AGW AGW to VGW VGW to VGW		
Comments		

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**RTP G.711**	→	
	←		←	**Fax transmission detected (ITU-T V.21 flags)**
INVITE	←	**SDP 3**	←	**INVITE**
488 (or 415)	→		→	**488 (or 415)**
ACK	←		←	**ACK**
		RTP G.711		
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

6.5.2 Fax detection at origin

The transition from modem over IP (MoIP) to FoIP occurs when a gateway (in MoIP mode) detects and verifies the presence of an [ITU-T T.30] facsimile signal such as ITU-T V.21 Channel 2 high-level data link control (HDLC) encoded flags or an ITU-T V.8 CM signal on the telephony link to the gateway (Annex F of [ITU-T T.38]).

<table>
<tr><td colspan="2">TSS</td><td colspan="2">TP 4.3.1</td><td>Reference
Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1];
Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1];
Clause D.2.2.4.2 of [ITU-T T.38]</td><td>Selection expression
PICS 1/2
PICS 2/2
PICS 3/2
PICS 9/2
PICS 10/2
PICS 22/2</td></tr>
<tr><td colspan="6">Scenario: Successful changeover, ITU-T G.711 accepted, ITU-T T.38 proposed</td></tr>
<tr><td colspan="6">Preconditions:
The SIP GW at the destination supports ITU-T T.38</td></tr>
<tr><td colspan="6">Signalling level:
In this scenario:
– The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729)
– ITU-T G.711 is accepted
– Fax is detected at calling -–party, initiating the re-INVITE proposing ITU-T T.38
– The called SIP UA accepts ITU-T T.38, answers with 200 OK (ITU-T T.38)
Media level:
Call is switched from ITU-T G.711 mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode.
Fax G3 options:
ITU-T V.34 G3 FD to ITU-T V.34 G3 FD
ITU-T V.34 G3 fallback to G3
The different QoS fax test-configurations with default parameters are contained in Table 5</td></tr>
<tr><td colspan="6">SDP header values:</td></tr>
<tr><td>A)</td><td></td><td colspan="4">INVITE SDP 1</td></tr>
<tr><td></td><td>a)</td><td colspan="4">m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.>
a=maxmptime: PIXIT <list of packet times separated by space>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>b)</td><td colspan="4">m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>c)</td><td colspan="4">m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>d)</td><td colspan="4">m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>e)</td><td colspan="4">m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td>B)</td><td></td><td colspan="4">SDP 2 (200 OK)</td></tr>
<tr><td></td><td>a)</td><td colspan="4">m=audio <port no.> RTP/AVP <PT assig.> [13] [18] [dyn.assign]
[a=ptime: PIXIT]
[a=rtpmap:<PT dyn.> telephone-event/8000]
[a=fmtp:<PT dyn.> <events PIXIT >]</td></tr>
<tr><td></td><td></td><td colspan="4">SDP 3 (INVITE)</td></tr>
<tr><td></td><td>a)</td><td colspan="4">m=audio <port no.> RTP/AVP <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT>
m=image <port no.> udptl t38
b=AS:46
a=T38FaxVersion: PIXIT
a=T38FaxRateManagement:PIXIT
a=T38MaxBitRate: PIXIT
a=T38FaxUdpEC:PIXIT
a=T38FaxMaxBuffer:1800
a=T38FaxMaxDatagram:150</td></tr>
</table>

	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
		SDP 4 (200 OK)
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
Note: [optional parameters] < mandatory parameters >		

Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**RTP G.711**	→	
Fax detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)				
INVITE	→	**SDP 3**	→	**INVITE**
200 OK	←	**SDP 4**	←	**200 OK**
ACK	←		←	**ACK**
		T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP		
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

<table>
<tr><td>TSS</td><td>TP 4.3.2</td><td>Reference
Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1];
Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1];
Clause D.2.2.4.2 of [ITU-T T.38]</td><td>Selection expression
PICS 1/2
PICS 2/2
PICS 3/2
PICS 9/2
PICS 10/2
PICS 22/2</td></tr>
<tr><td colspan="4">Scenario: Successful changeover: ITU-T G.729 accepted, ITU-T T.38 and ITU-T G.711 proposed</td></tr>
<tr><td colspan="4">Preconditions:
The SIP GW at the destination supports ITU-T T.38</td></tr>
<tr><td colspan="4">Signalling level:
In this scenario:
– The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729)
– ITU-T G.729 accepted
– Fax is detected at calling - party, initiating the re-INVITE proposing both ITU-T T.38 and ITU-T G.711
– The called SIP UA accepts ITU-T T.38, answers with 200 OK (ITU-T T.38)
Media level:
Call is switched from ITU-T G.729 mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode.
Fax G3 options:
ITU-T V.34 G3 FD to ITU-T V.34 G3 FD
ITU-T V.34 G3 fallback to G3 (see Figure 7)
The different QoS fax test-configurations with default parameters are contained in Table 5</td></tr>
<tr><td colspan="4">SDP header values:</td></tr>
<tr><td>A)</td><td></td><td colspan="2">INVITE SDP 1</td></tr>
<tr><td></td><td>a)</td><td colspan="2">m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=maxmptime: PIXIT <list of packet times separated by space>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>b)</td><td colspan="2">m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>c)</td><td colspan="2">m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>d)</td><td colspan="2">m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>e)</td><td colspan="2">m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td>B)</td><td></td><td colspan="2">SDP 2 (200 OK)</td></tr>
<tr><td></td><td>a)</td><td colspan="2">m=audio <port no.> RTP/AVP <18> [13] [PT assig.] <dyn.assign>
[a=ptime: PIXIT]
<a=rtpmap:<PT dyn.> telephone-event/8000>
<a=fmtp:<PT dyn.> <events PIXIT >></td></tr>
<tr><td>C)</td><td></td><td colspan="2">INVITE SDP 3</td></tr>
<tr><td></td><td>a)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> [18]</td></tr>
<tr><td></td><td>b)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> [18]
a=ptime: PIXIT</td></tr>
<tr><td></td><td>c)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> [18] <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td>D)</td><td></td><td colspan="2">SDP 4 (200 OK)</td></tr>
<tr><td></td><td>a)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> [18] [dyn.assign]
[a=ptime: PIXIT]
[a=rtpmap:<PT dyn.> telephone-event/8000]
[a=fmtp:<PT dyn.> <events PIXIT >]</td></tr>
<tr><td>E)</td><td></td><td colspan="2">SDP 5 (INVITE)</td></tr>
<tr><td></td><td>a)</td><td colspan="2">m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT>
[a=ptime: PIXIT]</td></tr>
</table>

		m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> [a=ptime: PIXIT] m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
F)		**SDP 6 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
Note: [optional parameters] < mandatory parameters >		
Comments		
Option A: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32 kbit/s) over RTP/UDP/IP; Data transmission detected at the **origination**;		

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-15)	→	**RTP G.729**	→	
Data transmission detected (e.g., CI, CT, CNG)				
INVITE	→	**SDP 3**	→	**INVITE**
200 OK	←	**SDP 4**	←	**200 OK**
ACK	→		→	**ACK**
	→	**RTP G.711**	→	
Fax transmission detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)				
INVITE	→	**SDP 5**	→	**INVITE**
200 OK	←	**SDP 6**	←	**200 OK**
ACK	→		→	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

Option B: Tone relay (see [IETF RFC 4733] – RTP payload for DTMF digits, telephony tones and telephony signals). Data transmission detected at the destination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-63)	→	**RTP G.729**	→	
				Data transmission detected (e.g., ANSam, /ANSam)
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
Fax detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)				
INVITE	→	**SDP 5**	→	**INVITE**
200 OK	←	**SDP 6**	←	**200 OK**
ACK	→		→	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.3.3**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	**Selection expression** PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Successful changeover: ITU-T G.729 accepted, ITU-T T.38 only proposed			
Preconditions: The SIP GW at the destination supports ITU-T T38			
Scenario **Signalling level:** In this scenario: The call is set-up with a speech codec. (e.g., ITU-T G.711 and ITU-T G.729) ITU-T G.729 accepted Fax is detected at **calling-party**, initiating the re-INVITE proposing **only** ITU-T T.38 The called SIP UA accepts ITU-T T.38, answers with 200 OK (ITU-T T.38) **Media level:** Call is switched from ITU-T G.711 mode to ITU-T T.38 mode. ITU-T T.30 negotiations and image transmission are performed in ITU-T T.38 mode. Fax G3 options: ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**INVITE SDP 1**	

	a)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <18> [13] [PT assig.] <dyn.assign> [a=ptime: PIXIT] <a=rtpmap:<PT dyn.> telephone-event/8000> <a=fmtp:<PT dyn.> <events PIXIT > >
C)		**INVITE SDP 3**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18]
	b)	m=audio <port no.> RTP/AVP <PT assig.> [18] a=ptime: PIXIT
	c)	m=audio <port no.> RTP/AVP <PT assig.> [18] <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
D)		SDP 4 (200 OK)
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18] [dyn.assign] [a=ptime: PIXIT] [a=rtpmap:<PT dyn.> telephone-event/8000] **[a=fmtp:<PT dyn.> <events PIXIT >]**
E)		**SDP 5 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
F)		**SDP 6 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150

	b)	m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
Note: [optional parameters] < mandatory parameters >		

Option A: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32kbit/s) over RTP/UDP/IP; Data transmission detected at the origination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-15)	→	**RTP G.729**	→	
Data transmission detected (e.g., CI, CT, CNG)				
INVITE	→	**SDP 3**	→	**INVITE**
200 OK	←	**SDP 4**	←	**200 OK**
ACK	→		→	**ACK**
	→	**RTP G.711**	→	
Fax transmission detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)				
INVITE	→	**SDP 5**	→	**INVITE**
200 OK	←	**SDP 6**	←	**200 OK**
ACK	→		→	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

Option C: Tone relay (see [IETF RFC 4733] – RTP payload for DTMF digits, telephony tones and telephony signals). Data transmission detected at the destination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-63)	→	**RTP G.729**	→	
				Data transmission detected (e.g., ANSam, /ANSam)
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
Fax transmission detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)				
INVITE	→	**SDP 5**	→	**INVITE**
200 OK	←	**SDP 6**	←	**200 OK**
ACK	→		→	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

<table>
<tr><td>TSS</td><td>TP 4.4.1</td><td>Reference
Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1];
Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1];
Clause D.2.2.4.2 of [ITU-T T.38]</td><td>Selection expression
PICS 1/2
PICS 2/2
PICS 3/2
PICS 9/2
PICS 10/2
PICS 22/2</td></tr>
</table>

<table>
<tr><td colspan="3">Scenario: Unsuccessful changeover, case 1: ITU-T G.729 accepted, ITU-T G.711 and ITU-T T.38 proposed (200 OK response)</td></tr>
<tr><td colspan="3">Preconditions:
The SIP UA at the destination doesn't support ITU-T T.38</td></tr>
<tr><td colspan="3">Signalling level:
In this scenario:
– The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729)
– ITU-T G.729 accepted
– Fax is detected at calling-party, initiating the re-INVITE proposing ITU-T G.711 and ITU-T T.38
– The called SIP UA rejects the ITU-T T.38 offer, answers with 200 OK ITU-T G.711
Media level:
ITU-T T.30 negotiations and image transmission are performed in ITU-T T.30 mode.
Fax G3 options:
ITU-T V.34 G3 FD to ITU-T V.34 G3 FD
ITU-T V.34 G3 fallback to G3
The different QoS fax test-configurations with default parameters are contained in Table 5</td></tr>
<tr><td colspan="3">SDP header values:</td></tr>
<tr><td>A)</td><td></td><td>SDP 1 (INVITE)</td></tr>
<tr><td></td><td>a)</td><td>m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=maxmptime: PIXIT <list of packet times separated by space>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>b)</td><td>m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>c)</td><td>m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>d)</td><td>m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td></td><td>e)</td><td>m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td>B)</td><td></td><td>SDP 2 (200 OK)</td></tr>
<tr><td></td><td>a)</td><td>m=audio <port no.> RTP/AVP <18> [13] [PT assig.] <dyn.assign>
[a=ptime: PIXIT]
<a=rtpmap:<PT dyn.> telephone-event/8000 >
<a=fmtp:<PT dyn.> <events PIXIT >></td></tr>
<tr><td>C)</td><td></td><td>INVITE SDP 3</td></tr>
<tr><td></td><td>a)</td><td>m=audio <port no.> RTP/AVP <PT assig.> [18]</td></tr>
<tr><td></td><td>b)</td><td>m=audio <port no.> RTP/AVP <PT assig.> [18]
a=ptime: PIXIT</td></tr>
<tr><td></td><td>c)</td><td>m=audio <port no.> RTP/AVP <PT assig.> [18] <PT dyn.>
a=ptime: PIXIT
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT ></td></tr>
<tr><td>D)</td><td></td><td>SDP 4 (200 OK)</td></tr>
<tr><td></td><td>a)</td><td>m=audio <port no.> RTP/AVP <PT assig.> [18] [dyn.assign]
[a=ptime: PIXIT]
[a=rtpmap:<PT dyn.> telephone-event/8000]
[a=fmtp:<PT dyn.> <events PIXIT >]</td></tr>
</table>

E)		**SDP 5 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
F)		**SDP 6 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT>
	b)	m=audio <port no.> RTP/AVP <PT assig.>
Note: [optional parameters] < mandatory parameters >		
Test configurations: AGW to AGW AGW to VGW VGW to VGW		
Comments		
Option A: Tone pass through: The tone is sent in-band using a lower compression algorithm such as the one used for voiceband data (VBD), e.g., encoded using [ITU-T G.711] or [ITU-T G.726] (32kbit/s) over RTP/UDP/IP; Data transmission detected at the **origin**;		

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-15)	→	**RTP G.729**	→	
Data transmission detected (e.g., CI, CT, CNG)				
INVITE	→	**SDP 3**	→	**INVITE**
200 OK	←	**SDP 4**	←	**200 OK**
ACK	→		→	**ACK**
	→	**RTP G.711**	→	
Fax transmission detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)				
INVITE	→	**SDP 5**	→	**INVITE**
200 OK	←	**SDP 6**	←	**200 OK**
ACK	→		→	**ACK**
	→	**T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

Option B: Tone relay (see [IETF RFC 4733] – RTP payload for DTMF digits, telephony tones and telephony signals). Data transmission detected at the destination				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event Codes (0-63)	→	**RTP G.729**	→	
				Data transmission detected (e.g., ANSam, /ANSam)
INVITE	←	**SDP 3**	←	**INVITE**
200 OK	→	**SDP 4**	→	**200 OK**
ACK	←		←	**ACK**
	→	**RTP G.711**	→	
Fax transmission detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)				
INVITE	→	**SDP 5**	→	**INVITE**
200 OK	←	**SDP 6**	←	**200 OK**
ACK	→		→	**ACK**
	→	**RTP G.711**	→	
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.4.2**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	**Selection expression** PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Unsuccessful changeover, case 2: ITU-T G.729 accepted, only ITU-T T.38 proposed (488 or 415 response / re-INVITE)			
Preconditions: The SIP UA at the destination doesn't support ITU-T T.38			
Signalling level: In this scenario: – The call is set-up with a speech codec (e.g., ITU-T G.711 or ITU-T G.729) – ITU-T G.729 accepted – Fax is detected at the calling-party, initiating the re-INVITE proposing ITU-T T.38 – The called SIP UA answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type) – The calling party sends a re-INVITE ITU-T G.711 in order to fallback to ITU-T G.711 for fax transmission **Media level:** ITU-T T.30 negotiations and image transmission are performed in ITU-T T.30 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			

A)		**SDP 1 (INVITE)**

	a)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <18> <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <18> <PT assig.> <13> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <18> [PT assig.] [13] <dyn.assign> [a=ptime: PIXIT] <a=rtpmap:<PT dyn.> telephone-event/8000> <a=fmtp:<PT dyn.> <events PIXIT >>
C)		**SDP 3 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
D)		**SDP 4 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT>
	b)	m=audio <port no.> RTP/AVP <PT assig.>
E)		**SDP 5 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT>
	b)	m=audio <port no.> RTP/AVP <PT assig.>
Note: [optional parameters] < mandatory parameters >		

Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
Tone relay (see [IETF RFC 4733] – RTP payload for DTMF digits, telephony tones and telephony signals).				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted IETF RFC 4733 Event codes (0-63)	→	**RTP G.729**	→	
Fax transmission detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)				
INVITE	→	**SDP 3**	→	**INVITE**
488 (or 415)	←		←	**488 (or 415)**
ACK	→		→	**ACK**
INVITE	→	**SDP 4**	→	**INVITE**
200 OK (INVITE)	←	**SDP 5**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
		RTP G.711		
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.4.3**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	**Selection expression** PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Protocol based switchover to ITU-T T.38, Fax detection at origin; Unsuccessful changeover, case 2: ITU-T G.711 accepted, only ITU-T T.38 proposed (488 or 415 response / re-INVITE)			
Preconditions: The SIP UA at the destination doesn't support ITU-T T.38			
Signalling level: In this scenario: – The call is set-up with a speech codec (e.g., ITU-T G.711 or ITU-T G.729) – ITU-T G.711 accepted – Fax is detected at calling-party, initiating the re-INVITE proposing ITU-T T.38 – The called SIP UA answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type) – The calling party sends a re-INVITE G.711 in order to fallback to G.711 for fax transmission **Media level:** ITU-T T.30 negotiations and image transmission are performed in ITU-T T.30 mode. Fax G3 options: ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**SDP 1 (INVITE)**	

	a)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18] [13] [dyn.assign] [a=ptime: PIXIT] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
C)		**SDP 3 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
D)		**SDP 4 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT>
	b)	m=audio <port no.> RTP/AVP <PT assig.>
E)		**SDP 5 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT>
	b)	m=audio <port no.> RTP/AVP <PT assig.>
Note: [optional parameters] < mandatory parameters >		

Test configurations: AGW to AGW AGW to VGW VGW to VGW				
Comments				
INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**RTP G.711**	→	
Fax transmission detected (ITU-T V.21 Channel 2 HDLC encoded flags or an ITU-T V.8 CM signal)	→		→	
INVITE	→	**SDP 3**	→	**INVITE**
488 (or 415)	←		←	**488 (or 415)**
ACK	→		→	**ACK**
INVITE	→	**SDP 4**	→	**INVITE**
200 OK (INVITE)	←	**SDP 5**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
		RTP G.711		
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

TSS	**TP 4.4.4**	**Reference**	**Selection expression**
		Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]; Clause D.2.2.4.2 of [ITU-T T.38]	PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: Unsuccessful changeover, case 3: ITU-T G.711 accepted, ITU-T T.38 only proposed (488 or 415 response / re-INVITE)			
Preconditions: The SIP UA at the destination doesn't support ITU-T T.38			
Scenario **Signalling level:** In this scenario: - The call is set-up with a speech codec (e.g., ITU-T G.711 and ITU-T G.729) - ITU-T G.711 accepted - Fax is detected at calling-party, initiating the re-INVITE proposing ITU-T T.38 - The called SIP UA answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type) - The calling party sends a ACK in order to fallback to previously negotiated PT assig. **Media level:** T.30 negotiations and image transmission are performed in T.30 mode. Fax G3 options: V.34 G3 FD to V.34 G3 FD V.34 G3 fallback to G3 **The different QoS fax test-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**SDP 1 (INVITE)**	

	a)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=maxmptime: PIXIT <list of packet times separated by space> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	b)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	c)	m=audio <port no.> RTP/AVP <PT assig.> <18> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	d)	m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
	e)	m=audio <port no.> RTP/AVP <PT assig.> <13> <18> <PT dyn.> a=ptime: PIXIT a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT >
B)		**SDP 2 (200 OK)**
	a)	m=audio <port no.> RTP/AVP <PT assig.> [18] [13] [dyn.assign] [a=ptime: PIXIT] [a=rtpmap:<PT dyn.> telephone-event/8000] [a=fmtp:<PT dyn.> <events PIXIT >]
C)		**SDP 3 (INVITE)**
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
	b)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150
Test configurations: AGW to AGW AGW to VGW VGW to VGW		
Comments		

INVITE	→	**SDP 1**	→	**INVITE**
180 Ringing	←		←	**180 Ringing**
200 OK (INVITE)	←	**SDP 2**	←	**200 OK (INVITE)**
ACK	→		→	**ACK**
Fax emitted	→	**RTP G.711**	→	
Fax transmission detected (ITU-T V.8 signals CI/CM)	→		→	
INVITE	→	**SDP 3**	→	**INVITE**
488 (or 415)	←		←	**488 (or 415)**
ACK	→		→	**ACK**
		RTP G.711		
BYE	→		→	**BYE**
200 OK (BYE)	←		←	**200 OK (BYE)**

6.6 Scenario 5 – ITU-T T.38 offer in initial INVITE (IAF device)

This clause provides the test cases for the scenario 5 – ITU-T T.38 offer in initial INVITE Internet aware fax device (IAF device).

TSS	**TP 4.5**	**Reference** Clause 5 of ETSI TS 124 229 V10.20.0 [ITU-T Q.3403 v.1]; Annex K of ETSI TS 129 163 [ITU-T Q.3629 v.1]	**Selection expression** PICS 1/2 PICS 2/2 PICS 3/2 PICS 9/2 PICS 10/2 PICS 22/2
Scenario: ITU-T T.38 offer in initial INVITE			
Preconditions: An IAF device is defined as an IP fax terminal connected to the voice over IP (VoIP) network using one port access gateway			
Scenario: An IAF device is defined as an IP fax terminal connected to the VoIP network using one port access gateway. Sometimes IAF as digital SIP terminal without modem part can send only ITU-T T.38 offer in the initial INVITE. **Signalling level:** Initial offer contains ITU-T T.38 media **Media level:** Call should be completed in ITU-T T.38 mode. Fax G3 options: G3 FD to G3 FD ITU-T V.34 G3 FD to ITU-T V.34 G3 FD ITU-T V.34 G3 fallback to G3 **The different QoS fax fest-configurations with default parameters are contained in Table 5**			
SDP header values:			
A)		**SDP 1 (INVITE)**	
	a)	m=audio <port no.> RTP/AVP <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150	
	b)	m=audio <port no.> RTP/AVP <PT assig.> <PT dyn.> a=rtpmap:<PT dyn.> telephone-event/8000 a=fmtp:<PT dyn.> <events PIXIT> a=ptime: PIXIT m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150	
	c)	m=audio <port no.> RTP/AVP <PT assig.> m=image <port no.> udptl t38 b=AS:46 a=T38FaxVersion: PIXIT a=T38FaxRateManagement:PIXIT a=T38MaxBitRate: PIXIT a=T38FaxUdpEC:PIXIT a=T38FaxMaxBuffer:1800 a=T38FaxMaxDatagram:150	

<table>
<tr><td>B)</td><td></td><td colspan="3">SDP 2 (200 OK)</td></tr>
<tr><td></td><td>a)</td><td colspan="3">m=audio <port no.> RTP/AVP <PT dyn.>
a=rtpmap:<PT dyn.> telephone-event/8000
a=fmtp:<PT dyn.> <events PIXIT>
m=image <port no.> udptl t38
b=AS:46
a=T38FaxVersion: PIXIT
a=T38FaxRateManagement:PIXIT
a=T38MaxBitRate: PIXIT
a=T38FaxUdpEC:PIXIT
a=T38FaxMaxBuffer:1800
a=T38FaxMaxDatagram:150</td></tr>
<tr><td></td><td>b)</td><td colspan="3">m=image <port no.> udptl t38
b=AS:46
a=T38FaxVersion: PIXIT
a=T38FaxRateManagement:PIXIT
a=T38MaxBitRate: PIXIT
a=T38FaxUdpEC:PIXIT
a=T38FaxMaxBuffer:1800
a=T38FaxMaxDatagram:150</td></tr>
<tr><td colspan="5">Test configurations:
AGW to AGW
AGW to VGW
VGW to VGW</td></tr>
<tr><td colspan="5">Comments</td></tr>
<tr><td>INVITE</td><td>→</td><td>SDP 1 (T.38)</td><td>→</td><td>INVITE</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>200 OK (INVITE)</td><td>←</td><td>SDP 2 (T.38)</td><td>←</td><td>200 OK (INVITE)</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>T.38 / UDPTL/UDP; RTP/UDP; TPKT/TCP</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>200 OK (BYE)</td><td>←</td><td></td><td>←</td><td>200 OK (BYE)</td></tr>
</table>

7 FoIP transmission parameters

This clause provides default ITU-T T.38 configurations, ITU-T T.38 parameter/value settings and generic formats.

7.1 ITU-T T.38 configuration

Table 2 illustrates the ITU-T T.38 parameter/value settings; generic format, i.e., signalling protocol independent and ITU-T T.38 transport unconditional. Table 3 illustrates recommended configurations for ITU-T T.38 UDPTL-based FoIP according to Table L.1 of [ETSI TS 126 114].

Table 2 – ITU-T T.38 configurations (ITU-T T.38 parameter/value settings; generic format, i.e., signalling protocol independent and ITU-T T.38 transport unconditional)

No.	Parameter	Value
0	**T38 Transport Mode**	**UDPTL/UDP** | RTP/UDP | TPKT/TCP
1	**T38FaxVersion**	**0** | 1 | 2 | 3 | 4
2	**T38MaxBitRate**	9600 | **14400** | 33600
3	**T38FaxFillBitRemoval**	**FALSE** | TRUE
4	**T38FaxTranscodingMMR**	**FALSE** | TRUE
5	**T38FaxTranscodingJBIG**	**FALSE** | TRUE
6	**T38FaxRateManagement**	localTCF | **transferredTCF**
7	**T38FaxMaxBuffer**	… | **1800** | …
8	**T38FaxMaxDatagram**	… | **150** | …

Table 2 – ITU-T T.38 configurations (ITU-T T.38 parameter/value settings; generic format, i.e., signalling protocol independent and ITU-T T.38 transport unconditional)

No.	Parameter	Value
9	**T38FaxMaxIFP**	… | **40** | …
10	**T38FaxUdpEC**	t38UDPFEC | **t38UDPRedundancy** | t38UDPNoEC
11	**T38FaxUdpECDepth**	*minred*:… | **1** | … *maxred*:… | ***none*** | …
12	**T38FaxUdpFECMaxSpan**	… | **3** | …
13	**T38VendorInfo**	$ … $ | ***parameter omitted***
14	**T38ModemType**	**t38G3FaxOnly** | t38G3AndV34G3
NOTE – Table 2 is the same as Table H.1 of [ITU-T T.38].		

Table 3 –Recommended configurations for ITU-T T.38 UDPTL-based FoIP according to Table L.1 of ETSI TS 126 114

SDP attributes	Value
T38FaxVersion	2 (or higher) (Note 1)
T38MaxBitRate	14400 bps
T38FillBitRemoval	N/A (Note 2)
T38FaxTranscodingMMR	N/A (Note 2)
T38FaxTranscodingJBIG	N/A (Note 2)
T38FaxRateManagement	'transferredTCF'
T38FaxMaxBuffer	1800 bytes
T38FaxMaxDatagram	At least150 bytes
T38FaxMaxIFP	40 bytes (Note 3)
T38FaxUdpEC	't38UDPRedundancy'
T38FaxUdpECDepth	'minred:1', 'maxred:2' (Note 3)
T38FaxUdpFECMaxSpan	3 (Note 3)
T38ModemType	't38G3FaxOnly' (Note 3)
NOTE 1 – Some SDP attributes listed here apply only to newer versions. NOTE 2 – Support not required. NOTE 3 – Only applicable when fax version 4 is supported. NOTE 4 – See [ITU-T T.38], Annex D, Table D.1 for a complete description. NOTE 5 – Table 3 is the same as Table L.1 of [ETSI TS 126 114].	

7.2 DTMF/telephony events

This clause provides dual tone multi frequency (DTMF) and data-related additions to the [IETF RFC 4733] telephony event registry. Table 4 illustrates different DTMF and data-related additions options to the [IETF RFC 4733] telephony event registry.

Table 4 – DTMF and data-related additions to the IETF RFC 4733 telephony event registry

Event Code	Event Name
DTMF Events	

Table 4 – DTMF and data-related additions to the IETF RFC 4733 telephony event registry

Event Code	Event Name
0	DTMF digit "0"
1	DTMF digit "1"
2	DTMF digit "2"
3	DTMF digit "3"
4	DTMF digit "4"
5	DTMF digit "5"
6	DTMF digit "6"
7	DTMF digit "7"
8	DTMF digit "8"
9	DTMF digit "9"
10	*
11	#
12	A
13	B
14	C
15	D
Data-related additions to IETF RFC 4733 telephony event registry	
23	CRdSeg: second segment of ITU-T V.8 bis CRd
24	CReSeg: second segment of ITU-T V.8 bis CRe
25	MRdSeg: second segment of ITU-T V.8 bis MRd
26	MReSeg: second segment of ITU-T V.8 bis MRe
27	V32AC: A pattern of bits modulated at 4800 bits/s, emitted by a ITU-T V.32/ ITU-T V.32bis answering terminal upon detection of the AA pattern.
28	V8bISeg: first segment of initiating ITU-T V.8 bis signal
29	V8bRSeg: first segment of responding ITU-T V.8
30	V21L300: 300 bits/s low channel ITU-T V.21 indication
31	V21H300: 300 bits/s high channel ITU-T V.21 indication
32	ANS (V.25 Answer tone). Also known as called station identification (CED) (ITU-T T.30 called tone).
33	/ANS (V.25 Answer tone after phase shift). Also known as /CED (ITU-T T.30 called tone after phase shift)
34	ANSam (ITU-T V.8 amplitude modified Answer tone)
35	/ANSam (ITU-T V.8 amplitude modified Answer tone after phase shift)
36	CNG (ITU-T T.30 calling tone)
37	ITU-T V.21 channel 1 (low channel), '0' bit
38	ITU-T V.21 channel 1, '1' bit. Also used for ESiSeg (second segment of ITU-T V.8 bis ESi signal.
39	ITU-T V.21 channel 2, '0' bit
40	ITU-T V.21 channel 2, '1' bit. Also used for ESrSeg (second segment of ITU-T V.8 bis ESr signal).

Table 4 – DTMF and data-related additions to the IETF RFC 4733 telephony event registry

Event Code	Event Name
49	CT (ITU-T V.25 Calling Tone)
52	ANS2225: 2225-Hz indication for text telephony
53	CI (ITU-T V.8 Call Indicator signal preamble)
54	ITU-T V.21 preamble flag (T.30)
55	V21L110: 110 bits/s ITU-T V.21 indication for text telephony
56	B103L300: Bell 103 low channel indication | IETF RFC 4734 for text telephony
57	V23Main: ITU-T V.23 main channel indication for text telephony
58	V23Back: ITU-T V.23 back channel indication for text telephony
59	Baud4545: 45.45 bits/s Baudot indication for text telephony
60	Baud50: 50 bits/s Baudot indication for text telephony
61	VBDGen: Tone patterns indicative of use of an unidentified modem type
62	XCIMark: A pattern of bits modulated in the ITU-T V.23 main channel, emitted by a ITU-T V.18 calling terminal.
63	V32AA: A pattern of bits modulated at 4800

7.3 Different QoS fax test-configurations with default parameters

This clause provides different quality of service (QoS) fax test-configurations with default parameters. Table 5 illustrates different QoS fax test-configurations with default parameters. Table 6 illustrates the definition of figure of merit. Table 7 illustrates the image quality categories. Table 8 illustrates the RTP payload types for standard audio and video encodings. Figure 7 illustrates the structure description language (SDL) diagram of MoIP fallback and transition to ITU-T T.30 fax by payload according to Figure F.4 of [ITU-T T.38].

Table 5 – Different QoS fax test-configurations with default parameters

No.	**From**				**PT assig.**	**To**			
	G.711 Modem Type	ECM	Page resolution	IP Gateway		IP Gateway	ECM	Page resolution	G.711 Modem Type
1.	V.34 Data rate 33.6 kbit/s	True	200x100 dpi	AGW / MSAN/ VGW	G.711, VBD with G711, T.38	AGW / MSAN/ VGW	True	200x100 dpi	V.34 Data rate 33.6 kbit/s
2	V.17 Data rate 14.4 kbit/s	True	200x100 dpi	AGW / MSAN / VGW	G.711, VBD with G711, T.38	AGW / MSAN/ VGW	True	200x100 dpi	V.17 Data rate 14.4 kbit/s
3	V.17 Data rate 14.4 kbit/s	False	200x100 dpi	AGW / MSAN/ VGW	G.711, VBD with G711, T.38	AGW / MSAN/ VGW	False	200x100 dpi	V.17 Data rate 14.4 kbit/s

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 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 – Definition of figure of merit

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. NOTE 3 – The definition of figure of merit is taken from [ITU-T E.458].			

Table 7 – Image quality categories

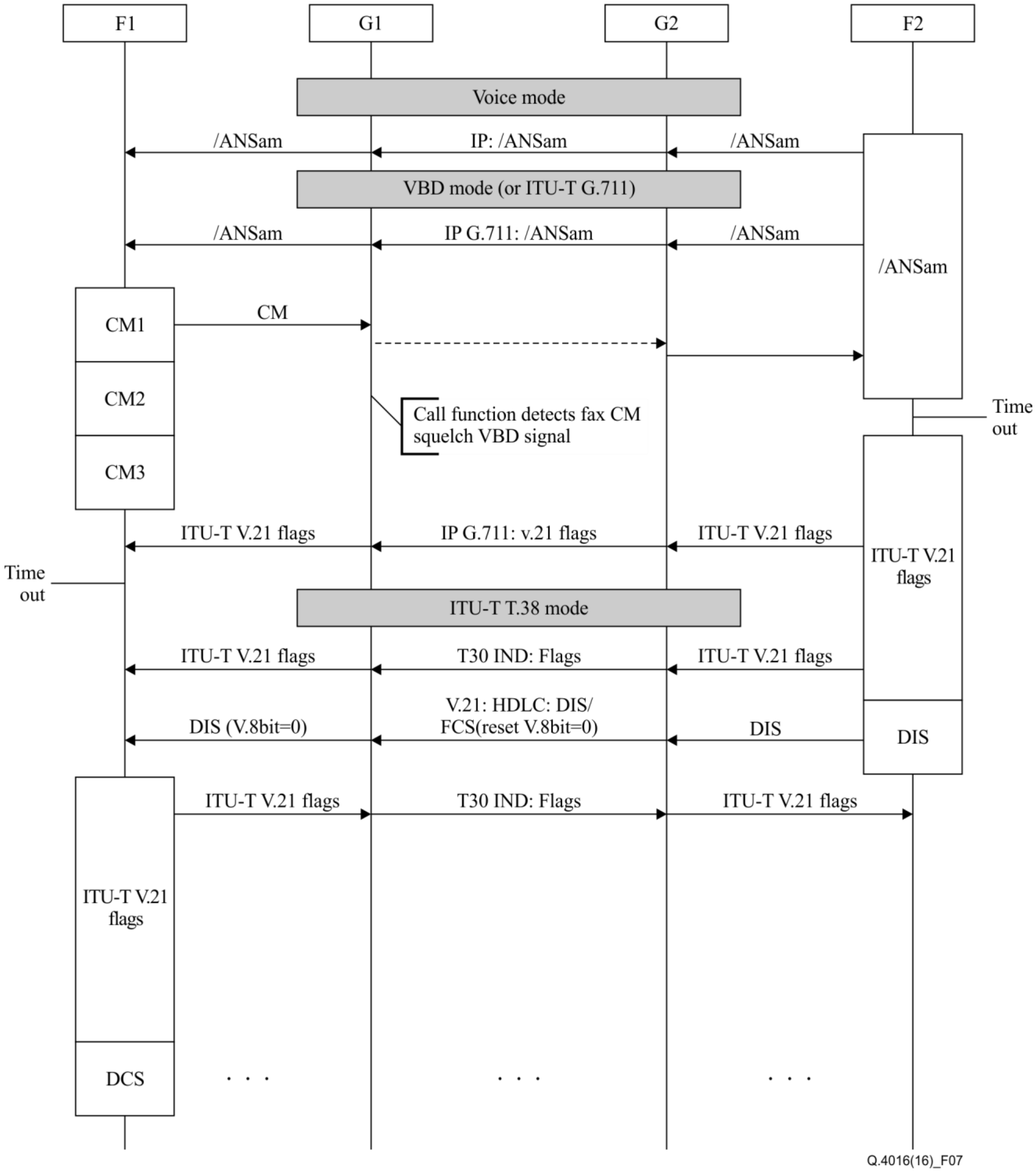
error-free page	No degradation by network impairments
errored page	Information conveyed
severely-errored page	Part of information missing
NOTE – Image quality categories are taken from [ITU-T E.453].	

Table 8 – RTP payload types for standard audio and video encodings

ITEM	**Payload types**	**Encoding Name**	**Audio/Video (A/V)**	**Clock Rate (Hz**	**Channels**
0	0	PCMU	A	8000	1
1	1	Reserved			
2	2	Reserved			
3	3	GSM	A	8000	1

Table 8 – RTP payload types for standard audio and video encodings

ITEM	Payload types	Encoding Name	Audio/Video (A/V)	Clock Rate (Hz	Channels
4	4	G723	A	8000	1
5	5	DVI4	A	8000	1
6	6	DVI4	A	16000	1
7	7	LPC	A	8000	1
8	8	PCMA	A	8000	1
9	9	G722	A	8000	1
10	10	L16	A	44100	2
11	11	L16	A	44100	1
12	12	QCELP	A	8000	1
13	13	CN	A	8000	1
14	14	MPA	A	90000	
15	15	G728	A	8000	1
16	16	DVI4	A	11025	1
17	17	DVI4	A	22050	1
18	18	G729	A	8000	1
19	19	Reserved	A		
20	20	Unassigned	A		
21	21	Unassigned	A		
22	22	Unassigned	A		
23	23	Unassigned	A		
24	24	Unassigned	V		
25	25	CelB	V	90000	
26	26	JPEG	V	90000	
27	27	Unassigned	V		
28	28	nv	V	90000	
29	29	Unassigned	V		
30	30	Unassigned	V		
31	31	H261	V	90000	
32	32	MPV	V	90000	
33	33	MP2T	AV	90000	
34	34	H263	V	90000	
35	35-71	Unassigned	?		
36	72-76	Reserved for RTCP conflict avoidance			
37	77-95	Unassigned	?		
36	96-127	dynamic	?		



NOTE: This figure is the same as Figure F.4 of [ITU-T T.38].

Figure 7 –SDL diagram of MoIP fallback and transition to ITU-T T.30 fax by payload

Annex A

PICS

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

A.1 Roles

Table A.1 – Roles

Item	Is the implementation ...	Reference	Status	Support
1	User		o.11	
2	Network		o.11	
NOTE: o.11: It is mandatory to support at least one of these items.				

A.2 Connection types

Table A.2 – Connection types

Item	Is the exchange able to ...	Reference	Status	Support
1	support the media type "audio" and media format 0		o.21	
2	support the media type "audio" and media format 8		o.21	
3	support the media type "audio" and media format 18 ?		o.21	
4	AMR-WB		o.21	
5	support the media type "image" and media format t38?		o.22	
6	support the dynamic assignment for PT assig.?			
7	use the transport protocol udptl?		o.23	
8	use the transport protocol tcptl?		o.23	
9	Use of telephone-event a=fmtp: <PT dyn.> 0-15		o	
10	Use of telephone-event a=fmtp: <PT dyn.> 0-15, 32,33,34,35,36,37, 38,39,40,49,53,54		o	
11	Support of silent suppression, media format 13		o	
12	Support transport packet redundancy with PT assig. G.711		o	
13	Supported VBD with PT assig. G.711 (V.152)		o	
14	Supported VBD with PT assig. G.711 (V.152) with assured transport packet redundancy		o	
15	Supported VBD with Codec G.711 (V.152) assured transport "FEC" with Codec G.711		o	
16	Non-assured transport with Codec G.726		o	
17	Support transport packet redundancy with Codec G.726		o.21	
18	Supported VBD with Codec G.726 (V.152)		o.21	
19	Supported VBD with Codec G.726 (V.152) with assured transport packet redundancy		o.21	
20	Supported VBD with Codec 726 assured transport "FEC" with Codec G.726		o.21	
21	Supported **SIP-controlled state transitioning between voice and facsimile**		o	
22	Supported T.38 strict controlled transitioning		o	
NOTE: o.21: It is mandatory to support at least one of these items. o.23: It is mandatory to support at least one of these items.				

Annex B

PIXIT

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

Table B.1 – Packetization size

Item	Packetization size	Reference	Status	Support
1	a=ptime: 10 ms		o.31	
2	a=ptime: 20 ms		o.31	
3	a=ptime: 30 ms		o.31	
Note: o.31 It is mandatory to support at least one of these items				

Table B.2 – ITU-T T.38 configuration

No.	Parameter	Value	Status	Support
0	T38 Transport Mode	UDPTL/UDP | RTP/UDP | TPKT/TCP	M	
1	T38FaxVersion	0 | 1 | 2 | 3 | 4	M	
2	T38MaxBitRate	9600 | 14400 | 33600	O	
3	T38FaxFillBitRemoval	FALSE | TRUE	O	
4	T38FaxTranscodingMMR	FALSE | TRUE	O	
5	T38FaxTranscodingJBIG	FALSE | TRUE	O	
6	T38FaxRateManagement	localTCF | transferredTCF	M	
7	T38FaxMaxBuffer	… | 1800 | …	M	
8	T38FaxMaxDatagram	… | 150 | …	M	
9	T38FaxMaxIFP	… | 40 | …	M	
10	T38FaxUdpEC	t38UDPFEC | t38UDPRedundancy | t38UDPNoEC	M	
11	T38FaxUdpECDepth	minred:… | 1 | … maxred:… | none | …	M	
12	T38FaxUdpFECMaxSpan	… | 3 | …	o.41	
13	T38VendorInfo	$ … $ | parameter omitted	o.41	
14	T38ModemType	t38G3FaxOnly | t38G3AndV34G3	o.41	
O.41 Only applicable when fax version 4 is supported				

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