Technical Specification for Fax Over IP Service

Workstream "Technical Aspects"

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Executive Summary

Problems with Fax over IP (FoIP) are experienced by all carriers and service providers in the world. While Voice over IP connections are usually set up without problems, fax connections often fail or connections are prematurely disconnected.

On the basis of two surveys that were launched between i3 forum carrier members in 2009 and 2010, it was possible to identify fax call phase and the section of the connection chain where most of the faults appear, as well as the versions of necessary standards implemented in carrier networks. Fax over IP sessions most often fail during call setup phase i.e.,, during fax discrimination, voice-fax switchover, or capability negotiations. Problems appear in gateway – gateway communication and between FoIP servers and gateways.

Information about the version of ITU-T T.38, ITU-T V.152 and other FoIP-related standards implemented in existing networks was the main assumption used in this document. Taking into account these results, this document is intended to deliver guidelines for reliable FoIP call setup in the following three areas:

[1] Specification of basic prerequisites for successful setup of fax calls over an IP link. These general prerequisites, such as necessary bandwidth, redundancy level and required network QoS, echo control, protocol stack and necessary gateway resources have been specified together with initial recommendations

[2] Guidelines for existing networks in order to take into consideration different standard version implementation. It has been considered that the significant number of different in-field implementations can make it unrealistic to define a minimum set of requirements and building a profile that could assure reliable FoIP sessions in each interconnection configuration. Moreover, fax connections are originated and terminated in service providers' networks which are not controlled by carriers. In this situation the guidelines for existing carrier's network are to:

   a. Identify the possible scenarios of FoIP connection setup
   b. Show how to search if recognised interoperability problems appear
   c. Recommend how to solve these problems. For each scenario, there are the proposals of dedicated tests that should find and eliminate all known bugs before a new interconnection link will be commercially used

[3] Analysis of the latest version of T.38, as well as related standards versions that are planned to be issued in the near future and outline their effect on the problems identified in the previous section

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1 Scope and objectives of this document

This document is intended to gather the most complete set of the best practices and technical guidelines for set up of reliable Fax over IP (FoIP) interconnections. The guidelines are prepared to help carriers take appropriate measures in their networks, taking into account that fax connections are originated and terminated out of their networks.

These measures are intended to assure reliable interconnections if originating and terminating endpoints are able to negotiate a common media format. If endpoints are not able to connect, then slight implementation changes may be proposed to the carrier's customers (service providers).

The structure of this document is as follows:

1. Network prerequisites defining the basic requirements concerning:
   - Bandwidth
   - Delay
   - Packet loss
   - Gateway configuration at TDM side and IP side
   - Gateway dimensioning

2. Detailed guidelines for existing carrier networks:
   This part is based on the status of implementation of FoIP-related standards as declared by carriers in two surveys launched in the years 2009 and 2010. The method proposed for reliable FoIP interconnection contains:
   - Identification of possible call configurations
   - Exchange of necessary information between carrier and service provider
   - Agreement of possible configuration details and parameters
   - Identification of possible FoIP scenarios
   - End-to-end tests
   - Searching and removing known bugs

   The next step is reference-configuration definition and description of:
   - Possible scenarios of G3 connections
   - Recommendations and tests for G3 connections
   - Possible scenarios of SG3/V.34 connections
   - Recommendations and tests for SG3/V.34 connections

   A summary test list is presented at the end.

3. An outline of the new standards and their impact on identified problems

2 Acronyms

ATA	Analogue Terminal Adapter
AGW	Access Gateway
CAC	Call Admission Control
COS	Class of Service
CPU	Central Processor Unit
CRTP	Compressed Real-Time Transport Protocol
DSD	Data Signal Detection
DSP	Digital Signal Processor
DTMF	Dual Tone Multi Frequency
ECM	Error Correction Mode
FEC	Forward Error Correction
FXS	Foreign eXchange Subscriber Interface
G3	Group 3
GW	Gateway
HDX	Half Duplex
IAF	Internet Aware Fax Device
MGC	Media Gateway Controller
MGCP	Media Gateway Control Protocol
NSE	Named Signalling Event
NSF	Non Standard Facilities
PSTN	Public Switched Telephone Network
PT	Passthrough
QoS	Quality of Service
RGW	Residential Gateway
RSVP	Resource Reservation Protocol
RTD	Round-Trip Delay
RTP	Real-Time Protocol
SDP	Session Description Protocol
SG3	Super Group 3 (V.34 fax)
SIP	Session Initiation Protocol
SIP-I	Session Initiation Protocol (Q.1912.5)
SRTP	Secure Real-Time Protocol
SSE	State Signaling Events
TCP	Transmission Control Protocol
TDM	Time Division Multiplex
TPKT	Transport Protocol Data Unit Packet
UCM	Universal Call Manager
UDP	User Datagram Protocol
UDPTL	Facsimile UDP Transport Layer (protocol)
URI	Universal Resource Identifier
VAD	Voice Activity Detection
VBD	Voice Band Data
VoIP	Voice over IP Protocol

3 References

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[27] IETF RFC 4566 ”SDP: Session Description Protocol”, July 2006
[28] IETF RFC 5939 “Session Description Protocol (SDP) Capability Negotiation”, September 2010
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[30] ETSI TR 183 072 (TISPAN); “Emulation Services for PSTN Modem Calls, V0.0.7”, 2010-06
[31] ETSI EG 202 057-2 V1.1.1 “Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 2: Voice telephony, Group 3 fax and modem data services”, (2002-09)
[32] ITU-T Recommendation G.711 “Appendix II: A comfort noise payload definition for ITU-T G.711 use in packet-based multimedia communication systems”, 02/2000
[33] IETF RFC 4734 “Definition of Events for Modem, Fax, and Text Telephony Signals”, December 2006
[34] “Fax, Modem, and Text for IP Telephony”, David Hanes, Gonzalo Salgueiro, Cisco Press, June 2008

4 Basic definitions

Fax pass-through - is a transport method of fax modulated data over an IP network where the waveform is digitized and transmitted using a lossless voice codec.

Fax relay – in this method, the modulated waveform is decoded. The decoded data is transmitted over an IP segment using a relay protocol.

VBD - fax pass-through method as defined in ITU-T V.152 [15]. This mode may be used when V.152 is supported by all gateways.

Pseudo VBD – fax pass-through method that allows transporting audio and VBD signals with the same media (codec) configuration. Pseudo VBDoIP, therefore, typically uses G.711 [8] without silence suppression, without adaptive Jitter Buffer control, without gain control, without noise reduction, overlayed by a G.168 [21]-compliant echo canceller (EC). It is used by pre-V.152 gateways. Definition follows ETSI TR 183 072 [30].

Emitting gateway - the gateway where the calling terminal is connected

Receiving gateway - the gateway where the called terminal is connected

G3 fax – standard fax terminal group 3

SG3 fax – fax terminal with V.34 capability

5 General prerequisites

This section is the list of basic requirements that should be met to assure reliable Fax-over-IP connections. The list contains basic but essential information, and is intended to be a checklist for the carrier personnel who configure interconnection links. General prerequisites contain network parameters and a gateway configuration guide.

5.1 Network prerequisites

5.1.1 Bandwidth

To set-up a successful call of any type it is necessary to assure necessary bandwidth for signalling layer and for media layer.

Signalling for fax is similar to voice call signalling and does not require additional bandwidth. In the media layer, the maximum bandwidth necessary for a single fax connection depends on transmission speed, packetisation period and redundancy used. Also, encryption of fax connections can increase the bandwidth used.

The maximum bandwidth necessary for selected cases has been calculated below:

Transmission	Packetisation period [ms]	Fax speed [kbit/s]	Redundancy level	Bandwidth [kbit/s]	
				IP	Ethernet
PT G.711	10	Any	Level 0 (RFC 2198)	96,0	110,4
PT G.711	10	Any	Level 1 (RFC 2198)	164	178,4
PT G.711	20	Any	Level 0 (RFC 2198)	80,0	87,2
PT G.711	20	Any	Level 1 (RFC 2198)	146	153,2
G3 / T.38	10	14,4	Level 0	43,2	57,6
G3 / T.38	10	14,4	Level 1	64,0	78,4
G3 / T.38	10	14,4	Level 2	83,2	97,6
G3 / T.38	20	14,4	Level 0	28,8	36
G3 / T.38	20	14,4	Level 1	46,4	53,6
G3 / T.38	20	14,4	Level 2	63,2	70,4
SG3 / T.38	10	33,6	Level 0	62,4	76,8
SG3 / T.38	10	33,6	Level 1	102,4	116,8
SG3 / T.38	20	33,6	Level 0	48,0	55,2
SG3 / T.38	20	33,6	Level 1	84,8	92,0

Note: Ethernet calculation without preamble. Calculations assume no encryption.

Table 1. Maximum bandwidth for selected FoIP connection modes (during facsimile transmission)

The total maximum bandwidth in the link required by fax connections should be calculated by multiplying the maximum bandwidth for a single call (during image transmission) by the expected number of simultaneous busy-hour fax connections for all supported types of fax connections. Bandwidth for a single FoIP call can be estimated using a simple Excel bandwidth calculator:

E:\d\DANE\IPIP\
Forum_2\T38 Rec\Do

5.1.2 Packet loss

Packet loss should be generally low because fax connections are very sensitive to it. In pass-through mode, when redundancy is applied a fax connection can be sustained with up to 1 percent of random packet loss [34] p.221. It is considered that for pass-through mode, level 1 of redundancy is a good compromise between necessary bandwidth and expected packet loss rate. In T.38, different redundancy levels are possible. When UDPTL transport is used, it is possible to use a different redundancy level for T.30 control messages (low speed) and for image transmission (high speed). In this case, a good compromise between bandwidth and reliability would be to use level 4 for low speed and level 1 for high speed as recommended in [25] p. 37. The decision of redundancy level used is usually taken according to service-provider policy.

5.1.3 Delay

In this case, two kinds of delay should be taken into consideration: transmission delay and signalling-processing delay. Transmission delay is a time of signal propagation in the IP network. Signalling delay is the total of transmission delay and the delay caused by signalling message processing by signalling equipment and terminals.

- **Transmission delay**
  Delay introduced in typical VoIP network is not a problem for fax connections. For voice, it should be kept below 150 msec. while, for fax delay, values of several seconds are usually acceptable in some, but not all, relay implementations.

  In case high delay occurs, it is possible that echo suppressors and echo cancellers in a PSTN call leg re-enable after having been disabled by ANSam, CED, or echo protection tone (EPT). It should be checked if such a situation happens and, if so, echo cancellers and echo suppressors should be appropriately adjusted.

- **SIP Signalling delay**
  Signalling delay is end-to-end SIP message-processing delay. If this delay is so long that T4 timer (as defined in ITU-T T.30 sec.5.4.2. [7]) expires, then the call will be disconnected. T4 typical value is 3 sec. ± 15% but some terminals reset T4 timer after receiving fax flags while the other wait for first message, so different waiting times can occur. High delay values may also cause message collisions. SIP signalling delay can be different for the same network and in different conditions e.g.,, for different traffic.

5.2 Gateway configuration

5.2.1 TDM side adjustments

In the case of TDM-to-IP media gateway, the following TDM-side parameters should be appropriately set:

- T38 fax signal volume
- Sending time of fax/modem signals. (2,6 s to 4s)
- Threshold of V.21 detection and threshold of modem detection

Other parameters adjustment may be also necessary depending on the gateway architecture and on vendor operation manual recommendations.

5.2.2 Voice Band Data (VBD) configuration

This mode means that fax-signalling and image data are transmitted through IP networks using a voice codec. VBD requirements are described in ITU-T V.152. The V.152 compliant gateways enter this mode when [*a=gpmd: vbd=yes parameter]* has been mutually agreed and appropriate VBD stimuli are detected.

Non-V.152-compliant gateways can also use VBD mode as a result of automatic speed increase after VBD stimuli are detected or can send a *Re-INVITE* using pseudo VBDoIP configuration. VBD and pseudo VBD mode assume that the following requirements are met:

a. G.711 A-law or G.711 µ-law is to be used (G.726-32 is also acceptable)
b. VAD, comfort noise and CRTP are disabled (provided DSD capability is supported as described in Amendment 1 to V.152 Annex B)
c. Jitter buffer is set to a fixed value according to expected jitter
d. Echo suppressors as described in ITU-T G.164 are disabled
e. Echo cancellers as described in ITU-T G.168 are enabled in case of V.21 and G3 image transmission and disabled in case of V.34 transmission

As target solution echo cancellers should be sensitive to V.21 preamble as described in Amendment 1 to V.152, Annex C [16].

5.2.3 Redundancy

Fax calls are very sensitive to packet loss, especially in T.30 control phase and high-packet-loss rate can cause fax call failure.

The appropriate redundancy level for a given packet loss value and various loss burst ratios cannot be easy calculated. Such an algorithm could be very complex because of packet loss burst nature. The general recommendation is to configure a high level of redundancy for low-speed data to protect fax-control messages and lower redundancy level for high-speed data, i.e.,, for image transmission. A good compromise between bandwidth and reliability is to use level 4 for low speed and level 1 for high speed as recommended in [25] p. 37.

5.2.4 COS marking

Packet marking for fax connections is not critical. The good practice is to use the same class as for voice but may also be other.

5.2.5 Media gateway buffers

Media gateways switch their buffer mode depending on the type of the transferred payload. The “TDM-to-IP” buffer is usually small and adaptive for voice, whereas it is greater and fixed for fax transmission.

Jitter-buffer delay is usually not critical and for fax it should be set to a fixed value according to expected jitter.

5.2.6 Echo control

It is possible that Echo Cancellers or Echo Suppressors affect fax-call announcement tones, thus affecting fax start-up phase, or it might even affect fax-image transmission.

Echo Cancellers and Echo Suppressors should be enabled/disabled while switching from audio to fax session as follows:

	Echo suppressors G.164	Echo cancellers G.165, G.168
G3 fax passthrough	Disabled	Enabled
G3 fax relay (T.38)	Disabled	Enabled
SG3 fax passthrough	Disabled	Disabled
SG3 fax relay (T.38)	Disabled	Disabled

Table 2. Echo suppressors/cancellers setting

Echo suppressors if they are present in TDM part may be re-enabled in case of big round-trip delay and impact the fax call. If such an effect appears it should be fixed and eliminated.

5.2.7 Transport

There are three following transport stacks that can be used in fax-relay mode as defined in ITU-T T.38: UDPTL/UDP, TPKT/TCP, RTP/UDP.

The UDPTL/UDP transport stack is most often used and is supported by all gateways so it seems to be the best choice for fax relay.

5.3 Media gateway dimensioning

The media gateways should be properly configured depending on their architecture in order to avoid overload of gateway resources by fax connections.

The amount of gateway resources depend on installed hardware and its capacity. The proper configuration of resources requires the following values to be taken into consideration:

- Maximum number of call attempts per second
- Maximum number of simultaneous connections
- DSP pool and number of calls per DSP
- CPU capacity and memory amount
- Number of E1/T1 ports
- Maximum load of E1/T1 boards
- IP bearer capacity

A fax pass-through connection requires less DSP capacity than a fax relay.

According to practical experience, a T.38 G3 call requires similar DSP capacity as G.729a VoIP call. CPU capacity and memory amount necessary for T.38 calls is usually greater than for normal VoIP call.

It is recommended to determine the gateway capacity for fax connections before it is used in a new link (see section 7.1 "Initial tests"). Then during operation it is recommended to supervise if the level of CPU load is in the range allowed by the vendor.

6 Detailed technical guidelines

6.1 Current situation in existing networks

In existing carrier networks different versions of FoIP-related standards are implemented. In this situation, it is impossible for a carrier to choose a protocol profile that can guarantee that each fax call will be successful.

The standards that define a media configuration for Fax over IP (T.38 and V.152) still contain ambiguous fragments. Although they are continuously improving, there is still a wide variety of implementations in the industry. The latest standards versions are rarely implemented in carrier networks and the same situation is likely to exist in service-provider networks where different gateways and ATAs can be used.

In the signalling layer, the existing legacy SDP protocol capabilities do not always allow the SIP peers to negotiate the media configuration during one Offer/Answer. If Offer/Answer dialogue lasts longer than six seconds, the session will likely fail if a media reinvite is attempted.

The proposed way to have reliable FoIP interconnection in existing carrier networks is to check and eliminate all known failures and interoperability problems that have been so far identified, and to introduce the necessary changes and check if FoIP connections are reliably set up. It can be performed in the following steps:

- Identify possible call configurations
- Exchange necessary information between carrier and service provider
- Agree all possible configuration details and parameter values

– Identify possible scenarios for FoIP
– Perform end-to-end tests

6.2 Interconnection configurations

Calling and called fax terminals are located in interconnected service-provider networks. Fax calls can be initiated in TDM or VoIP network and terminated in TDM or VoIP network as shown in i3 forum reference configuration (see Fig. 1 below). Connections initiated/terminated in MNO's networks are out of scope in this version of the document.

The best case with relative small amount of identified problems is when IP interconnected fax call is initiated and terminated in TDM networks (e.g.,, TDM1 - Carrier A – Carrier B – TDM2). In this configuration, trunking media gateways are usually under carrier control. In practice, fax connections in this configuration rarely fail.

If VoIP networks are interconnected, then a fax call can be initiated and/or terminated by:

a) Normal fax terminal connected using an ATA or Access/Residential Gateway (PSTN Emulation)
b) Internet Aware Fax device connected directly to VoIP network (PSTN Simulation)

Such configurations require information exchange with interconnected Service Providers. On the basis of this information, the Border Gateways/Border Gateway Controllers can be appropriately configured.

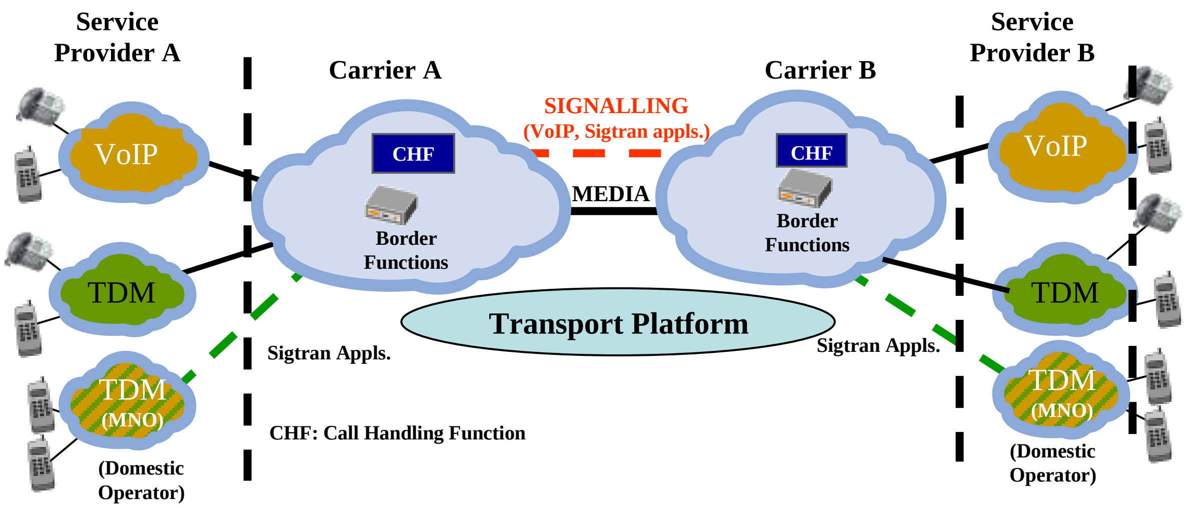

Fig. 1 General Reference Configuration

6.3 Information exchange and initial settings

Before the commercial start-up of a new interconnection link, it is recommended to exchange the following information with the Service Provider and make appropriate settings.

6.3.1 Call Control Protocol

It is assumed that for interconnect purposes SIP/SIP-I will be used as call control protocol with legacy SDP for session description.

6.3.2 Fax Tones Transport

Fax tones transport method is an important setting. These tones disable/enable echo suppressors and echo cancellers and may also be used for remote switchover triggering.

A fax call is initially set up as voice call and it may happen that the bandwidth assigned to the call will not be sufficient for in-band tone transport. In this case, the call may fail in the very beginning or during image transmission. Tone transport method should be agreed between carriers and service providers.

It is strongly recommended to negotiate and use DTMF/telephony events relay method as described in RFC 2833[12] /RFC 4733 [13]/RFC 4734 for V.21, V.8 and T.30 tones. In this case fax signalling tones are transmitted as defined events and not using the audio codec negotiated. The following events should be supported:

ANS (CED)	*32*
/ANS	*33*
ANSam	*34*
/ANSam	*35*
Calling tone (CNG)	*36*
V.21 channel 2, "0" bit	*39*
V.21 channel 2, "1" bit	*40*
V.21 preamble flag	*54 (if RFC 4734 [33] is supported*

In the gateways, this transport method should be configured as follows:
-Transfer of fax tones to the packet side
-Fax tones sent using RFC2833/4733/4734 to the TDM side
-Alternatively for fax tones transport pseudo VBD method can be used.
-Usually media gateways are set to switchover from voice to fax mode, triggered by local fax terminal signals but it is also possible that media gateways switchover can be triggered by remote fax terminal signal. In this latter case, using RFC2833/4733/4734 can essentially improve reliability

6.3.3 Fax training mode

Training mode is a key parameter which must always be considered.

It is recommended to use a transferred mode of training. In this mode the training signal is passed end-to-end.

Local training mode is recommended only for reliable transport such as TCP when both G3 devices are identified via DIS/DCS exchange as IAF devices (T.38 [5] sec.8,) or when end-to-end delays are high.

6.3.4 ECM

The use of Error Correction Mode assures the fidelity of images sent. However, when packet loss in the network is high, ECM can cause many retransmissions and redials. For that reason, UDPTL with redundancy is preferred.

Since carriers declare that in existing carrier networks packet loss rate is very low, ECM can be enabled on carrier's gateways to let the terminals negotiate the use of ECM.

If ECM is enabled it is recommended to set redundancy for high-speed (image) not greater than 1.

6.3.5 Non-Standard Facilities

Some gateways allow the carrier to elect whether the gateway should transparently transmit NSF, if present, or remove it (via "spoofing") from the transaction. In some cases, NSF can cause the protocol being used by the endpoint terminals to be so "non-standard" that the T.38 implementations are unable to handle it. Since this is rare, it is recommended that the *gateways be configured to transparently pass NSF, but the option to remove it could be exercised in some cases.*

6.3.6 T.38 fax call parameters

In T.38 fax relay, the appropriate SDP parameters are necessary. Some parameters and parameter-default values are not precisely defined in currently used T.38 versions and, consequently, their interpretation may be different in different implementations.

Recommended default values for UDPTL/UDP transport and proposed meaning are as follows:

(M= Mandatory, O=Optional)

Parameter	*Meaning*		*Default value*
Negotiated parameters:			
T38FaxVersion	*0, 1 = ASN.1 syntax 1998 supported 2 ASN.1 syntax 2002 supported 3 = ASN.1 syntax 2002 + V.34*****	*M*	*0*
T38FaxUdpEC	*Error correction used: Redundancy, FEC or without error correction*	*M*	*t38UDPRedundancy*
T38FaxFillBitRemoval	*Removal and reinsert of fill bits applied*	*O*	*NO,* *****
*T38FaxTranscodingMMR ***	*Ability to transcode MH/MR from/to a facsimile endpoint to MMR data between the T.38 gateways*	*O*	*NO,* *****
*T38FaxTranscodingJBIG ***	*Ability to transcode send JBIG data between T.38 gateways*	*O*	*NO,* *****
Declarative parameters:			
T38MaxBitRate	*Max. bit rate for image transmission*	*O*	*14400*
T38FaxRateManagement	*Fax training method: transferred or local*	*M*	*transferredTCF*
*T38FaxMaxBuffer ***	*Maximum single UDPTL,(RTP or TPKT) payload that the endpoint can accept.*	*O*	*1800*
*T38FaxMaxDatagram ****	*Maximum IFP primary message size the endpoint is prepared to receive.*	*O*	*150*

Note: * No means that parameter is not mentioned at all. Parameter=NO does not appear in SDP
***** Use of these parameters is not clear (SIP Forum Problem Statement)***
****** Definition of this parameter is ambiguous (SIP Forum Problem Statement)***
******* V.34 support is commonly assumed though not always implemented. (SIP Forum Problem Statement)***

Table 3. T.38 Fax Call Parameters

The parameter values are determined by endpoints located in service provider networks and carriers usually approve what is used by their customers. Carrier's networks should transmit this information transparently.

If possible the carrier may suggest interconnected service providers to change used parameter values and use. It is recommended to use the profile described in PacketCable™ document PKT-SP-CODEC-MEDIA-I09-100527 "Codec and Media Specification, 2.0" [25], section 7.4.2.5 – 7.4.2.7.

When T.38 session is active the parameters cannot be changed. If any endpoint sends reINVITE with new session parameters, the other endpoint should accept the offer without trying to change their values. If the offer is rejected the call could be disconnected.

6.3.7 T.38: support of string "user=phone"

It could happen that a call agent implementation drops fax calls, if "From" or "To" fields contain the string "user=phone", within SIP re-INVITEs for T.38 fax.

The string "user=phone" in "From" and "To" headers inside SIP re-INVITE message means only that the URI should be interpreted as tel-URI based on E.164 telephone numbers. The correct MGC behaviour should be checked by appropriate test (see section 7.1 "Initial tests"). If this problem exists then SIP implementation should be corrected.

6.4 Possible scenarios for G3 fax in existing networks

6.4.1 Status of necessary standard implementation

The surveys performed among the i3 Forum carriers showed that the status of implementation of the standards necessary to set-up successful fax calls in carrier networks gateways is as follows:

1. T.38 standard version implemented is Edition 1 from June 1998 [1] or Edition 2 from March 2002 [2].
2. V.152 implementation is very poor. Only a small percent of carriers have it implemented on their equipment.
3. T.38 Edition 2 from March 2002 [2] which enables autonomous transitioning method to T.38 fax relay is very rarely implemented.
4. RFC 2833 [12] is mostly implemented, however fax events are sometimes not supported.
5. ECM is supported by almost all carriers.
6. Cisco proprietary NSE* switchover method is almost not used and can be neglected.

* Named Signalling Events

6.4.2 G3 scenarios description

Taking into account the status of FoIP-related standards implementation, the following connection setup scenarios are possible:

1. <u>Automatic speed increase – use of pseudo VBD</u>
   This simple scenario can be used in all networks that have neither V.152 nor T.38 standards implemented. It is based on the capability of the gateways to increase speed to pseudo VBDoIP mode after detection of FoIP or MoIP tones or receiving packets with appropriate payload.
   Auto speed increase to VBD mode after detection of continuous 2100 Hz tone can also be used. If initial voice call has been set up using e.g., G.729 codec, then the gateway switches to pseudo VBD when it detects continuous 2100 Hz tone.

   Required standards: RFC 4733 [13] /2833 [12] /4734 [33] may be useful to transmit fax/modem events to the gateways if triggering tone is configured to come from remote end.

2. <u>V.152 VBD negotiated – auto switchover</u>
   This scenario assumes that emitting and receiving gateways are both V.152-compliant gateways and appropriate transparency is assured throughout the connection chain. In this scenario, VBD is negotiated during connection setup. When defined VBD stimuli appear, autonomous switchover takes place as defined in ITU-T T V.152 [15]. Redundancy according to RFC 2198 [18] is possible. The bandwidth used is greater, but redundancy makes the call less sensitive to packet loss.

   Required standards: ITU-T V.152 mandatory, RFC 4733/2833 and RFC 2198 optional.

3. <u>T.38 negotiated – auto switchover to T.38</u>
   It is possible to use this scenario for H.248 controlled gateways provided that T.38 standard version from 2002 is implemented in emitting and receiving gateways and

appropriate transparency is assured throughout the connection chain. In this scenario, T.38 capability is negotiated during the connection setup using two “m” lines in SDP Offer/Answer. When defined T.38 stimuli are detected, an autonomous switchover takes place. The redundancy 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. As autonomous switchover is described in ITU-T T.38 [5] only for H.248 controlled gateways, its use with SIP-controlled voice gateways is unclear and not recommended.
Required standards: T.38 version 2002 + Amendment 1 (Version 2) mandatory.

4. <u>T.38 – protocol-based switchover</u>
   This scenario assumes that at least T.38 standard version 0 from 1998 [1] is implemented in emitting and receiving gateways and appropriate transparency is assured throughout the connection chain. In this scenario, after setting up a voice call and detection of fax tones the protocol-based switchover takes place. The receiving gateway sends re-INVITE and initiates switchover to T.38 relay mode. Both redundancy and FEC error-correction modes are possible. Terminals can be only G3. If one or both terminals are SG3 then gateways should be able to perform fallback to G3 procedure. In this case the bandwidth can be smaller and the sensitivity to packet loss is lower.

   Protocol-based switchover is most often used, and many problems that can appear during this scenario were identified. Most of them are specified in “SIP Forum – Fax over IP Task Group – Problem and Recommendation Statement”. On the basis of this document some checks and tests are proposed below. Required standards: T.38 version 1998 [1] mandatory.

5. <u>T.38 protocol-based switchover rejected -> pseudo VBD</u>
   This scenario assumes that the gateway sending re-INVITE supports any version of ITU-T T.38 and the opposite gateway does not. In this case the receiving gateway initiates switchover to T.38 sending re-INVITE after CED or V.21 preamble detection. The emitting gateway should answer with 415 error code and appropriate accept header containing possible session parameters. Then the receiving gateway sends re-INVITE with SDP offer using received configuration proposal. A similar situation can appear when an emitting gateway initiates a switchover to T.38 after detection of CNG (what is rarely used).Required standards: any ITU-T T.38 version implementation on the gateway initiating protocol based switchover to T.38.

6. <u>T.38 request in initial offer</u>
   In this scenario IAF initiates the connection. This device is capable to offer T.38 connection in initial INVITE (fax only connection as described in T.38 [5] D.2.2.3 and D.2.4.1). If the receiving gateway has any version of T.38 implemented and the receiving terminal is fax or if the receiving terminal is IAF the connection will be set up. If the receiving device is a gateway then the problem may appear. According to “SIP Forum – Fax Over IP Task Group – Problem and Recommendation Statement” [24] it is possible that some gateways may be unable to set up any connection if audio stream is not offered.

   SIP Forum recommends that IAF should never offer T.38 only, but always together with an audio stream that can be “recvonly”. It is assumed that in this case IAF should be able to offer audio media stream as well.

7. <u>G3 fax initiates connection to IAF</u>
   In this scenario, audio connection is offered to IAF. If IAF does not support audio it should answer with 415 error code and an appropriate accept header with T.38 as the

only possible media configuration. If the gateway has any version of T.38 implemented then the connection should be set-up.

6.5 G3 scenario identification

The possible scenario combination depending on implemented standards can be summarized in the following table:

Emitting gateway /terminal	Receiving gateway/terminal				
	Automatic speed increase	V.152	T.38 June 1998	T.38 March 2002	IAF
Automatic speed increase	1	1 (note 1)	5 (note 1)	5 (note 1)	(note 2)
V.152	1(note 1)	2	5 (note 1)	5 (note 1)	(note 2)
T.38 June 1998	1(note 1)	1 (note 1)	4	4	7
T.38 March 2002	1(note 1)	1(note 1)	4	3, 4	3,7
IAF	(note 2)	(note 2)	6	6	6

Note 1: If automatic speed increase implemented on both gateways
Note 2: Connections may probably fail. IAF are very rarely used and its features are not well defined. Their behaviours are to be described in more detail in the next version of this document.

Table 4. Possible scenario combinations in existing networks

6.6 Recommendations and testing for G3 connections

According to the capabilities of all the gateways in the connection chain, the possible scenarios should be identified and the gateways properly configured.

6.6.1 Testing configuration

For new link testing carriers should have the reference testing network as shown below. It allows checking if FoIP problems are detected in interconnection offered to a service provider.

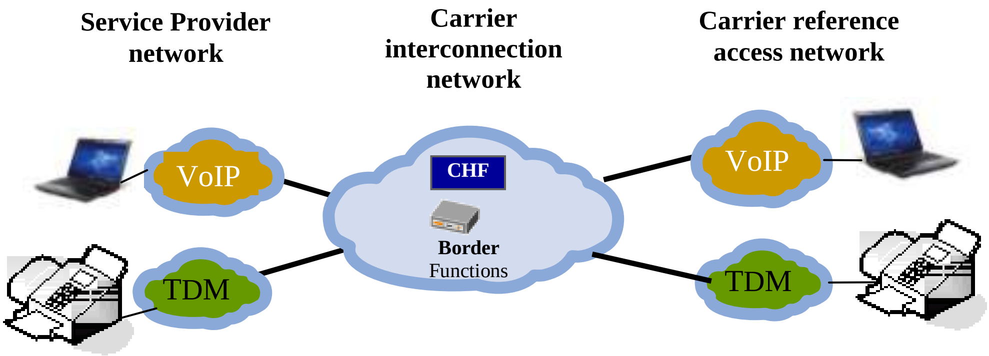

Fig. 2 - Testing Configuration

6.6.2 Scenario 1 - Automatic speed increase

Recommendations:

It is recommended that the triggering tones which can be used by gateways as well as their source (local or remote end, in band or out of band transmission) are identified and checked:

a) *All possible speed-increase triggering tones should be analysed.*
b) *If detection of triggering tones on the IP side is used, the use of RFC 2833 [12]/4733 [13]/4734 [33] is recommended*
c) *SBC should transparently transmit the new RTP payload.*
d) *SBC should not disconnect the call when it detects that bandwidth used is greater than negotiated*.

Tests:

a) Echo suppressors test (optional: if echo suppressors are installed in TDM part).
   Set-up TDM to VoIP call. Play CNG, CED and ANSam; echo suppressors should not mute any direction.
b) Echo cancellation test.
   Track internal gateway processes. Check if echo cancellers are enabled during G3 fax call. Check if echo cancellers are disabled during SG3 fax call.
c) Automatic speed increase test (disable of VAD, fixed jitter buffer)
   In the scenarios using pseudo VBDoIP check if upspeed has been activated after agreed fax tones and if all VBD requirements are met. Track internal gateway processes. Test scenario depends on the gateway type.
d) Fax tones and training tones transfer.
   Track the call and check if RFC 2833 [12] payload appears. If not then G.711 should be applied. Check if fax tones are correctly played to, and recognized by, the end terminal. Check if training is performed correctly.
e) Check of SBC transparency to auto payload change.
   Track the packets sent from SBC to AGW/RGW and MGW. SBC should transparently transmit new payload.
f) Check of SBC transparency to bandwidth increase without signalling.
   Track the call, after payload change the call should not be disconnected. SBC should not disconnect the call if after auto speed increase from audio to VBD the bandwidth used in the call is greater.

6.6.3 Scenario 2 – V.152 negotiated VBD

Recommendations:

a) *ITU-T V.152 [15] should be supported by emitting and receiving gateway.* SBC should transparent.
b) *If detection of triggering tones on IP side is used then RFC 2833 [12]/4733 [13] is recommended to be used*.

Tests:

a) Fax tones transfer through an IP segments.
   Track the call and check if RFC 2833 [12] payload appears. If not G.711 codec should be applied. Check if fax tones are correctly played to and recognized by the end terminal.
b) V.152 switchover track.
   Tracking the switchover in testing configuration and verify if it is performed correctly.

6.6.4 Scenario 3 - auto switchover to T.38

Recommendations:

Use of this scenario is recommended for the TDM-IP-TDM interconnection when H.248 controlled MGW are used. For IP-IP or IP-TDM interconnection where SIP-controlled voice *gateways or AGW/RGW are involved, this scenario is not recommended because of the lack of a clear standardization.*

Tests:

a) Autonomous T.38 switchover track.
   Track the call. Both INVITE offer and 200 OK answer should contain two "m" lines: audio and T.38 with nonzero port numbers. During the call, the switchover to T.38 should take place just after detection of V.21 preamble (or other trigger set) by receiving gateway. The call should be correctly completed in T.38 mode.

6.6.5 Scenario 4 – protocol-based switchover to T.38

Most of the identified problems occur in this widely used scenario. The following guidelines prepared on the basis of "SIP Forum – Fax Over IP Task Group – Problem and Recommendation Statement" [24] should help to improve its reliability.

Recommendations:

a) T.38 switchover triggers.

   There are many possibilities to initiate the switchover from audio to T.38 fax relay. It can be initialised by emitting gateway when it detects CNG generated on TDM side or by the receiving gateway when it detects CNG on the packet side or the local CED, as well as V.21 preamble generated locally or remotely. It may happen that both gateways simultaneously send the signalling messages trying to initiate the switchover and the call fails. In most cases the receiving gateway initiates the switchover when it detects fax tones coming from the local end terminal.

   Carriers and Service Providers should identify if switchover triggers do not cause conflicts and eliminate it in cooperation with service providers.

b) Signalling processing delay.

   Carriers and Service Providers should consider and check the maximum total delay between the receiving gateway's 200OK to initial INVITE and the subsequent re-INVITE to T.38. The tests performed by SIP forum showed that a relationship exists between failed fax relay calls and this delay. If this delay was shorter than 5 seconds then most of the calls were successful. When delay was longer it could cause T.30 timers to expire. This problem is to be further investigated by SIP Forum and i3 Forum.

   It is recommended to keep the delay between receiving gateway's 200OK to initial INVITE and the subsequent re-INVITE to T.38 as short as possible and to check if longer delay causes the fax calls failure.

c) Muting voice channel during switchover.

   After detection of the fax tone which is used to trigger the voice to fax switchover, the receiving gateway should mute the existing audio channel in both directions. The suppression of the voice channel should be applied before the terminal starts sending NSF/CSI/DIS. If the terminals start to exchange DIS/DCS in audio mode and the gateway controllers simultaneously negotiate T.38 in signalling layer then the call may fail in the switchover phase.

   SIP Forum recommends that in case of protocol based T.38 switchover, the suppression of audio channel in both directions should be applied in 800 milliseconds after detection of V.21 preamble if this preamble is used to trigger the switchover.

d) Media Stream Configuration after T.38 switchover.

   After the switchover to T.38, the media stream configuration can be different. "SIP Forum Problem Statement" [24] specifies four possible configurations. The call begins usually with an active audio media stream which is sometimes accompanied with an inactive T.38 stream. After the switchover, a new T.38 stream appears or the existing inactive one is activated. An existing audio stream can disappear, become inactive or a new audio stream is created when the existing one becomes inactive. If the endpoints use different configurations it may cause problems.

It is recommended to check if the media configuration after the switchover is correct and if only one media stream is active. Border gateways and SBCs should be transparent to different configuration changes.

Tests:

a) Protocol-based switchover-triggering test for conflicts.
Recommended trigger is V.21 preamble detected by the receiving gateway. In any case, it is recommended to track the call and check if only one gateway sends re-INVITE with T.38 offer. If requester-INVITE is sent by both gateways, it means that different triggering events are used. The correct triggering event should be agreed with service providers.
b) Measurement of maximum delay between fax discrimination tone and sending re-INVITE by receiving gateway.
Track internal gateway processes and measure maximum delay.
c) Measurement of maximum total delay between the receiving gateway's 200OK to initial INVITE and the subsequent re-INVITE to T.38.
Track the call for different network load and measure the delay.
d) Measurement of the delay between V.21 preamble and muting voice channel by receiving gateway.
Test scenario depends on the gateway type. Track internal gateway processes and measure maximum delay. It should be below 800 ms if V.21 is used as switchover trigger. Track the call and check if the receiving gateway did not send complete DIS before the flow of audio packets is stopped.
e) Test of media streams configuration after switchover to T.38.
Track SIP signalling and check the media streams configuration after switchover. Only one media stream can be active in the same time period.
f) Check the correct reaction to “user=phone” string in re-INVITE to T.38.
If this parameter is present in re-INVITE it should not cause fax call disconnection.

6.6.6 Scenario 5 – T.38 protocol-based switchover rejected

Recommendations:

In this case, the offer should include pseudo VBD mode i.e., G.711 codec with no VAD as in scenario 1. As it is not possible to disable VAD and set fixed jitter buffer using SDP parameters it is recommended that default implementation of G.711 is without VAD and that the gateway sets jitter buffer to fixed value after rejection of T.38 offer.

a) Correct re-INVITE rejection should be with 415 error code and with appropriate accept header as defined in IETF RFC 3261
b) New media configuration should be G.711 codec without VAD as described in 5.2.2.

Tests:

a) Re-INVITE rejection error code check.
Track SIP signalling and check the error code as well as accept header.
b) Test of the media configuration after call set up: pseudo VBD.
Track SIP signalling and check final re-INVITE SDP offer. Track internal gateway processes and check if pseudo VBD has been activated. Test scenario depends on the gateway type.

6.6.7 Scenario 6 – T.38 request in initial offer (Internet Aware Fax Device)

Recommendations:

Internet Aware Fax device (IAF) is a normal fax terminal connected to the VoIP network using one port Access Gateway. In this case, the connection is always initially set up as a voice call and then switched over to T.38. If, however, an IAF is a digital SIP terminal without modem part, it can send only T.38 offer in the initial INVITE. This type of call is described in ITU-T T.38 [5] as a “fax only” call. Some gateways are unable to accept such a call before they check if the end terminal has appropriate capabilities. In this case, the call will not be set up. If Service Provider’s customers use IAFs it is recommended to test this scenario.

According to SIP Forum Problem Statement the IAF should always send an audio offer together with T.38 offer to avoid call rejection. The audio stream should be marked as “recvonly” because IAF may not be able to send audio.

Tests:

a) T.38 offer in initial INVITE test
   Set up a fax call using IAF sending T.38 offer in initial INVITE. Track the call and check if the setup is performing correctly in case of IAF as originating terminal and a gateway as the receiving device. If necessary, an audio stream should be added in the initial offer.

6.6.8 Scenario 7 – G3 fax initiates connection to IAF

Recommendations:

If IAF does not support audio it should reject the call with 415 error code and appropriate accept header indicating T.38 capabilities. If the emitting gateway is T.38 capable, then the call will be set up. Otherwise the call will be disconnected. If IAF supports audio then audio call will be set up and IAF should send re-INVITE with T.38 offer before sending first DIS in audio mode.

G3 gateway should always send audio media stream together with T.38 media stream in initial offer. IAF can answer with audio + T.38 or T.38 only. In both cases the call should be set up.

Tests:

a) G3 to IAF test
   Check call setup when G3 fax initiates connection to IAF supporting and not supporting audio. Check if audio is offered together with T.38.

6.7 Possible scenarios for SG3 fax in existing networks

Assuming that the status of standard implementation status is the same as described in 6.4.1 different scenarios are possible depending on fax terminals and gateways capabilities.

6.7.1 Call setup between G3 and SG3 terminals

If one of the terminals trying to connect is a G3 only terminal, then the call will fall back to normal G3 connections regardless of the gateways capabilities in the connection chain. The fallback procedure is described in ITU-T.30 [7] clause 6 p.80.

6.7.2 SG3 scenarios description

If calling and receiving terminals are both V.34 capable SG3, then the call handling depends on gateway capabilities. The following scenarios are possible:

Fax pass-through scenarios

1. Automatic speed increase – use of pseudo VBD.
   If the gateways have auto speed increase implemented then the scenario for SG3 fax is the same as for G3 fax.

2. V.152 VBD negotiated – auto switchover
   If V.152 is negotiated at the call setup then the scenario for SG3 fax is the same for G3 fax.

Fax relay scenarios
The possible scenarios for T.38 fax relay depend on emitting and receiving gateway capabilities. They are specified in T.38 as follows:

Emitting gateway V.34 HDX capable	Receiving gateway V.34 HDX capable	Comment
No	No	Standard T.38
No	Yes*	Fallback to Standard T.38
Yes*	No	Fallback to Standard T.38
Yes*	Yes*	V.34 HDX T.38 procedures used

* V.34 HDX capable gateway should have at least T.38 version from 2002 with Amendment 1 implemented.

Table 5. Possible scenarios for T.38 / V.34

Taking into account the possibility of forced fallback, there are three additional fallback scenarios:

3. Fallback when emitting gateway is not V.34 HDX capable
   In this case emitting gateway will not recognize CM and will not transmit it to receiving gateway. If the receiving terminal will not receive CM, then it will fallback to G3 and send V.21 preamble. Receiving gateway will recognize this signal and will re-INVITE to T.38 and initiate a protocol-based switchover (see scenario 4 and 5 for G3 fax).

   Required standards: T.38 version not supporting SG3 fax, RFC 4733 [13] /2833 [12] /4734 [33] may be useful to transmit fax/modem events.

4. Fallback when receiving gateway is not V.34 HDX capable
   False alarm for ANSam as CED generated by receiving gateway's CED detector will cause sending CED to emitting gateway. CED transmitted to emitting SG3 terminal should make it to fallback to G3. Receiving SG3 terminal will not receive CM and after timeout it will fallback to G3 and send V.21 preamble and initiate protocol-based switchover (see scenario 4 and 5 for G3 fax).

   Required standards: T.38 version not supporting SG3 fax, RFC 4733 [13] /2833 [12] /4734 [33] may be useful to transmit fax/modem events.

5. Forced fallback
   It is also possible to force fallback procedure as follows:
   - The communication is established in voice mode with the codec defined by configuration.
   - Called terminal sends ANSam. The receiving GW and the emitting GW shall switch to modem passthrough mode (i.e., the GW shall adapt the G.711 codec to the passthrough mode).
   - If CM V8 command indicates a "facsimile" call function (Table 3/V.8) then the emitting GW must block transmission of this command over IP.
   - After time out, the called terminal will send V.21 preamble and then DIS command to the receiving gateway. To prevent a possible return to V.8 negotiation the

receiving gateway may reset the V.8 bit of the DIS message (bit 6, first octet as defined in T.30 [7] table 2/T30. p.52).

- Upon detection of DIS, the receiving GW shall try to switch from voice or modem (G.711 payload) communication to fax T.38 communication. The switch shall be operated by signalling messages exchange between the two MGCs (by sending a SIP Re-INVITE for example).
- If switch to T.38 relay mode is not possible, the connection should fall back to fax passthrough mode.

Required standards: T.38 version supporting SG3 fax, RFC 4733 [13] /2833 [12] /4734 [33] may be useful to transmit fax/modem events.

If emitting and receiving gateways are both V.34 HDX capable and appropriate transparency of interconnection network is assured then a full SG3 scenario will take place:

6. Normal SG3 V.34 connection

When both terminals are SG3 terminals and emitting and receiving gateway are V.34 HDX capable, then after the voice call is set-up V.8 procedures are used for fax switchover as described in T.38 [5].

Required standards: T.38 version supporting SG3 fax, RFC 4733 [13] /2833 [12] /4734 [33] may be useful to transmit fax/modem events.

6.8 SG3 fax recommendations and testing

According to the capabilities of all the gateways in the connection chain, the possible SG3 scenarios should be identified and the gateways properly configured.

6.8.1 Fax pass-through scenarios

For pass-through scenarios, it is important that carriers SBCs and gateways support the automatic speed increase or V.152 all over the connection route.

Recommendations: as for Scenario 1 and Scenario 2 for G3 fax.

Tests: as for Scenario 1 and Scenario 2 for G3 fax.

6.8.2 Fax relay scenarios

Fallback scenarios

Recommendations:

a) In case of forced fallback emitting gateway should have the capability to block CM if the facsimile bit is set.

b) In case of forced fallback emitting and receiving gateway should have T.38 version 2002 [2] with Amendment 1 or later implemented

c) In case of forced fallback receiving gateway should have the capability to reset the V.8 bit of the DIS message

d) Carrier's networks should be transparent to V.8 and V.34.

Tests:

a) SG3 to G3 fallback procedures verification.

Track the call and check if fallback to G3 takes place. If yes, verify if it is performed according to the description in ITU-T T.38 [5] Appendix F (F3 Fig. F4/T.38)

Normal SG3 V.34 connection

Recommendations:

- *a) Emitting and receiving gateway should have T.38 version 2002 [2] with Amendment 1 or later implemented.*
- *b) AWG/RGW in Service Provider networks are recommended to use RFC 2833 [12]/ 4733 [13] / 4734 [33] with fax events.*
- *c) When ANSam is first detected, the calling device must reply with CM within 2,6 till 4 seconds to setup SG3 fax connection. If the delay is longer, the answering machine will fall back to G3 fax and begin to send DIS sequences*.

Tests:

- a) Echo cancellers test with ANSam.
  Track internal gateway processes. Check if echo cancellers are disabled during SG3 fax call. Track incoming signal – echo should be visible.
- b) Fax tones transfer through IP segments.
  Track the call and check if RFC 2833/4733 payload appears. If not G.711 payload should be applied. Check if fax tones are correctly played to, and recognized by, the end terminal.

7 Summary test list

7.1 Initial tests

- a) Testing T.30 terminals by PSTN call
  Set up PSTN fax call as G3 fax and SG3 fax to check T.30 implementation in terminals.
- b) Testing VoIP connections – normal voice calls.
  Normal voice calls should be set up to check if the interconnection works correctly.
- c) IP tests, ping, RTD, jitter, packet loss.
- d) Terminal time out for command – answer delay (optional).
  During PSTN call to block DCS sending by calling terminal (after DIS from receiving terminal) and measure the delay after which the terminal will disconnect the call. Usually after T4 expiration DIS is repeated 3 times,
- e) Resource capacity testing – maximum number of fax call which does not cause gateway capacity overflow.
  This test should be performed in a laboratory for a given type and configuration of the gateway (AGW or SBC). It is necessary to determine the maximum number of simultaneous fax calls which can be set up by the gateway.
- f) Redundancy level testing for network QoS degradation.
  Set-up a fax call and increase packet loss until image quality begins to be unacceptable.

7.2 End-to-end tests

- a) Echo suppressors test optional: if echo suppressors are installed in TDM part (6.6.2)
- b) Echo cancellation test (6.6.2)
- c) Automatic speed increase test (disable of VAD, fixed jitter buffer) (6.6.2)
- d) Fax tones and training tones transfer (6.6.2)
- e) Check of SBC transparency to auto payload change (6.6.2)
- f) Check of SBC transparency to bandwidth increase without signalling (6.6.2)
- g) V.152 switchover track (6.6.3)
- h) Autonomous T.38 switchover track (6.6.4)
- i) Protocol-based switchover triggering test for conflicts (6.6.5)
- j) Measurement of maximum delay between fax discrimination tone and sending re-INVITE by receiving gateway (6.6.5)

k) Measurement of maximum total delay between the receiving gateway's 200OK to initial INVITE and the subsequent re-INVITE to T.38 (6.6.5)
l) Measurement of the delay between V.21 preamble and muting voice channel by receiving gateway (6.6.5)
m) Test of media streams configuration after switchover to T.38 (6.6.5)
n) Check of correct reaction to “user=phone” string in re-INVITE to T.38 (6.6.5)
o) Re-INVITE rejection error code check (6.6.6)
p) Test of the media configuration after call setup: pseudo VBD (6.6.6)
q) T.38 offer in initial INVITE test (6.6.7)
r) G3 to IAF test (6.6.68)
s) Echo cancellers test with ANSam (6.8.2)
t) SG3 to G3 fallback procedure verification (6.8.2)

7.3 Final verification tests

a) Switchover procedures testing according to chosen scenario(s).Tracking call setup to check if scenario runs properly.
b) Facsimile transmission testing. Test of image transfer quality. Set-up a fax call, send the pattern and check the quality according to ETSI EG 202 057-2 [31]
c) Facsimile transmission speed registration
d) Check of gateway and SBC behaviour in case of T.38 session parameters changed during T.30 session. This test is based on the case described in “SIP Forum Problem Statement” [24]. Session parameters change during active T.38 call should normally not appear. However, if it appeared it should be accepted without changing real session parameters.

8 Impact of the new standard versions

It is expected that in the near future the new versions of following standards will appear:

1. ITU-T T.38
2. ITU-T V.152
3. IETF RFC 5939
4. IETF draft SDPMediaCapNeg will become valid RFC and will be implemented

The most important identified problems which are expected to be solved by implementation of the above-mentioned standards are as follows:

1. ITU-T V.152
– Problems with reenabling of echo cancellers in some cases of SG3 to G3 fallback

2. ITU-T T.38
- Simultaneous triggering of T.38 switchover by different fax tones
- Different interpretation of call parameters and its use
- Different parameter’s default values
- Late muting voice channel during T.38 switchover

3. RFC 5939 and SDPMediaCapNeg
– High delay of SIP signalling message processing
– Handling of T. 38 only in initial offer
– media configuration after T.38 switchover

These standards, together with the existing version of RFC 4733 [13] and RFC 4734 [28], will be progressively implemented in carrier’s and service provider’s networks. It is also assumed that ECM will be finally supported for FoIP in all IP networks.

Full implementation of the new standard is expected to solve all existing FoIP problems. The impact of the new standard implementation and possible new problems, if they appear, are to be described in the next release of this document.