INTERNATIONAL INTERCONNECTION FORUM
FOR SERVICES OVER IP

(i3 FORUM)

(www.i3forum.org)

Source: Workstream "Technical Aspects"

Technical Specification for Fax over IP service

(Release 3.0, May 2013)

The guidelines and conclusions as well as target solution proposal were elaborated in close cooperation between i3 Forum and SIP Forum (www.sipforum.org). Two testing session carried out by i3 Forum carriers were possible thanks to Copia International and Commetrex who provided, their software CopiaFacts and Bladeware to testing participants. i3 forum acknowledges the cooperation of SIP Forum Fax over IP Task Group in analyzing the testing results.

Revision History

Date	Rel.	Subject/Comment
October 2010	1.0	First release of Fax over IP document
May 14th 2012	2.0	Second release based on testing with SIP Forum
May 12th 2013	3.0	Third release of based on full IP testing session

Executive Summary

Problems with fax over IP are experienced by all carriers and service providers. While voice-over-IP connections are usually set up without problems, fax connections often fail or connections are prematurely disconnected.

This document is intended to deliver guidelines for reliable FoIP call setup. It contains the following parts:

[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 have been specified, together with initial recommendations;

[2] description of common problems identified during i3 forum and SIP Forum testing campaigns in real carrier production environments. These problems seem to be the reason of different FoIP calls failures in mixed TDM/IP interconnection configurations.

[3] conclusions drawn from these observations are as follows:
- most of failures appear when IP and TDM interconnections segments are mixed along the route.
- end-to-end IP route with end-to-end SIP signalling should let us to avoid these failures.
- fax call should be identified before the route will be selected and this information should be used to route the call appropriately.

[4] proposal of two possible target solutions. Both solutions are based on fax call identification and on the methods of intelligent routing of identified fax call. Usually a FoIP call is set up as a normal voice call and routing entities are not aware that this particular call will be a fax call. In the proposed smart-routing solutions, the information about a fax call’s nature is known to routing entities before call set up starts. This information is then used to route a fax call via a fax-capable route.

[5] Appendix A with a description of i3 forum and SIP Forum testing configuration and results.

[6] Appendix B with recommendations concerning Fax-over-IP call set up.

Table of Contents

1 Scope and Objective of the document

This document is intended to gather the most complete set of the best practices and technical guidelines for the setup of reliable Fax-over-IP (FoIP) interconnections. The proposed guidelines take into account the complexity of carrier and service-provider networks that contain PSTN and VoIP segments. Prepared on the basis of testing campaigns, the guidelines should help carriers and service providers to introduce appropriate measures in their networks to significantly increase the success rate of FoIP calls.

The structure of this document is as follows:

1. Network prerequisites defining the basic requirements concerning:
   - Bandwidth
   - Delay
   - Packet loss
   - COS marking
   - Echo control

2. A detailed description of failure reasons identified during common testing campaigns performed by i3 forum and SIP Forum in real international interconnect environments. The description compares reference configuration with existing interconnection networks and lists the problems resulting from:
   - Breaking SIP signalling into independent sections separated by TDM/C7 sections.
   - Delays in signalling-message processing.
   - Analog signal discontinuity resulting in slips and distortions.
   - Incorrect recognition of training and V.21 events.

   In the end of this part main conclusions are listed.

4. Target solution proposals. Two solutions using early fax call identification and smart routing are presented.

5. Appendices describing testing details and Fax ove IP call setup recommendations.

2 Acronyms

ATA	Analogue Terminal Adapter
AGW	Access Gateway
CAC	Call Admission Control
CFR	Confirmation to Receive (T.30 response)
COS	Class of Service
CPU	Central Processor Unit
CRTP	Compressed Real-Time Transport Protocol
DCS	Digital Command Signal (T.30 command)
DIS	Digital Identification Signal (T.30 command)
DSD	Data Signal Detection
DSP	Digital Signal Processor
DTMF	Dual Tone Multi Frequency
ECM	Error Correction Mode
EOP	End of Procedure (T.30 command)
FEC	Forward Error Correction
FTT	Failure to Train (T.30 response)
FXS	Far eXchange Subscriber Interface
G3	Group 3
GW	Gateway
HDX	Half Duplex
IAF	Internet Aware Fax Device
MCF	Message Confirmation (T.30 response)
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
RTN	Retrain negative (T.30 response)
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

[1] ITU-T Recommendation T.38 "Procedures for real-time Group 3 facsimile communication over IP networks", 1998
[2] ITU-T Recommendation T.38 "Procedures for real-time Group 3 facsimile communication over IP networks", 2002
[3] ITU-T Recommendation T.38 "Procedures for real-time Group 3 facsimile communication over IP networks", 2004
[4] ITU-T Recommendation T.38 "Procedures for real-time Group 3 facsimile communication over IP networks", 2005
[5] ITU-T Recommendation T.38 "Procedures for real-time Group 3 facsimile communication over IP networks", 2007
[6] ITU-T Recommendation T.38 "Procedures for real-time Group 3 facsimile communication over IP networks", 2010
[7] ITU-T Recommendation T.30 Procedures for document facsimile transmission in the general switched telephone network 09/2005.
[8] ITU-T Recommendation G.711 "Pulse Code Modulation of Voice Frequencies", 1988
[9] T.38 Amendment 1: "Support for half-duplex V.34 and V.150.1 interworking", 07/2003
[10] T.38 Amendment 2: „Support for optional RTP encapsulation, clarification of version negation procedures and modification of "no-signal"", 04/2004
[11] T.38 Amendment 3: "Procedure for real-time group 3 facsimile communication over ip networks: T.38 implementation guidelines", 01/2004
[12] IETF RFC 2833 "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals", 2000
[13] IETF RFC 4733 "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals" December 2006
[14] ITU-T Recommendation V.150.1 "Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs", 2003
[15] ITU-T Recommendation V.152. "Procedures for supporting voice-band data over IP networks", 2005.
[16] ITU-T Recommendation V.152 "Amendment 1". 2009.
[17] V.152 "Procedures for supporting voice-band data over IP networks": Proposed revisions
[18] IETF RFC 2198 "RTP Payload for Redundant Audio Data", 1997
[19] ITU-T Recommendation G.164 "Echo Suppressors", 1988
[20] ITU-T Recommendation G.165 "Echo Cancellers", 1993
[21] ITU-T Recommendation G.168 "Digital network echo cancellers" 04/2000
[22] ITU-T Recommendation H.248 "Gateway control protocol", Version 3", 2005
[23] ITU-T Recommendation E.164 "The international public telecommunication numbering plan", 1997
[24] "SIP Forum – Fax Over IP Task Group – Problem and Recommendation Statement. T.38: Problems related to SIP/SDP Negotiation", 21.04.2010
http://www.sipforum.org/component/option,com_docman/task,doc_view/gid,412/Itemid,261/
[25] PKT-SP-CODEC-MEDIA-I10-120412 "Codec and Media Specification, 2.0", PacketCable™
http://www.cablelabs.com/specifications/PKT-SP-CODEC-MEDIA-I10-120412.pdf
[26] IETF RFC 3261 "SIP: Session Initiation Protocol", June 2002.
[27] IETF RFC 4566 "SDP: Session Description Protocol", July 2006
[28] ITU-T Recommendation T.4 "Standardization of Group 3 facsimile terminals for document transmission." 07/2003
[29] ETSI TR 183 072 (TISPAN); "Emulation Services for PSTN Modem Calls, V0.0.7", 2010-06
[30] 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)
[31] 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
[32] IETF RFC 4734 "Definition of Events for Modem, Fax, and Text Telephony Signals", December 2006
[33] "Fax, Modem, and Text for IP Telephony", David Hanes, Gonzalo Salgueiro, Cisco Press, June 2008
[34] ITU-T Recommendation V.8 "Procedures for starting sessions of data transmission over the public switched telephone network", 11/2000

[35] IETF RFC 3840, "Indicating User-Agent Capabilities in the Session Initiation Protocol." 08/2004
[36] IETF RFC 3841 "Caller Preference for the SIP" 80/2004
[37] IETF RFC 6913 "Indicating Fax over IP Capability in the Session Initiation Protocol (SIP)". /03/2013
[38] IETF RFC 3326, "The Reason Header Field for the Session Initiation Protocol (SIP)." 12/2002
[39] IETF RFC 4856, "Media Type Registration of Payload Formats in the RTP Profile for Audio and Video Conferences." 03/2007

4 Basic Definitions

Fax pass-through - is a transport method of modulated fax data over an IP network where the waveform is digitized and transmitted using a lossless voice codec such as G.711.

Fax relay – in this method, the modulated waveform is decoded by an "emitting" gateway. The decoded data is then transmitted over an IP segment using a relay protocol. The relayed data is then remodulated by the "receiving" gateway and transmitted to the called endpoint terminal over a TDM network.

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, adaptive Jitter Buffer control, gain control, or 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 [29].

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 Problem statement

Problems with fax over IP are experienced by all telecom customers. While voice-over-IP connections are usually set up without problems, fax connections often fail or connections are prematurely disconnected.

The reasons of Fax over IP failures can be divided in two groups:

1. Failures caused by the faulty implementation of T.30 and T.38 standards in IP fax terminals and servers.
2. Failures caused by a network that is used to set up and transmit fax call.

After several years of related standard development and implementation improvement, Fax over IP works almost perfectly in local enterprise networks that are fully controlled by a single owner. Most of the failures appear in complex configurations when two local networks are interconnected with one or more TDM and IP interconnection network segments. Elimination of network-related failures should make this service sufficiently reliable.

6 Network basic 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. The prerequisites concern mainly IP segments of the network but some of them (as delay and echo control) are common for IP and TDM segments.

6.1 Bandwidth

To setup a successful call of any type it is necessary to assure necessary bandwidth for the signalling and media layers.
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, packetization 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	Packetization 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 image 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

6.2 Packet loss

Packet loss (caused either by misrouting or by late arrival) should be generally low because fax connections are very sensitive to it. Within high-quality pay-for-service networks, especially those purchased for business use, this is unlikely to be an issue. However, where routes include the open Internet performance is not guaranteed and can vary widely. In pass-through mode, when redundancy is applied, a fax connection can be sustained with up to 1-percent of random packet loss [33] p.221.For pass-through mode on routes that include the open Internet, level 1 of redundancy is a good compromise between necessary bandwidth and expected packet loss rate, however businesses that purchase IP service are likely to find even level 1 redundancy prohibitively expensive in bandwidth.

In T.38 mode, 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 on routes that include the open Internet would be to use level 4 for low speed and level 1 for high speed as recommended in [25] p.37. On high-quality routes, level 1 for low speed and level 0 for high speed is sufficient. The decision of redundancy level used is usually taken according to service-provider policy. FoIP pass-through calls, even with redundancy implemented, are susceptible to burst packet loss. (During the open Internet tests one or two 30-msec breaks in a packet stream would kill a G.711 fax call.)

6.3 Delay

Two kinds of delay should be taken into consideration: transmission delay and signalling-processing delay. Transmission delay is the 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 networks is not a problem for fax connections. For voice, it should be kept below 150 msec., while for fax, delay values of up to 3-4 seconds are usually acceptable in some, but not all, relay implementations. During testing, T.38 and G.711 fax calls over the open Internet were successfully setup while RTD was over 330 ms.

- **SIP Signalling delay**
  Signalling delay is end-to-end SIP message-processing delay. If this delay is so long that the 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 others wait for the 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.

Especially critical is the maximum total delay between the receiving gateway's 200OK to the initial INVITE and the subsequent re-INVITE to T.38. If this delay is long enough to let the calling terminal hear a DIS message from the called terminal and the calling terminal starts to send DCS, it is too late to change the session to T.38 mode. If this occurs, a re-INVITE to T.38 usually kills the session because the T.30 signalling has reached the “no-return point”. The tests performed by the SIP forum showed that a relationship exist 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 over 6 seconds most of the calls failed.

6.4 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.

6.5 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 segments, may be re-enabled in case of a large round-trip delay and affect the fax call. If this occurs, it should be fixed and eliminated.

7 Interconnection reference configuration

Calling and called fax terminals are located in interconnected service-provider networks. Fax calls can be initiated in TDM or VoIP networks and terminated in TDM or VoIP networks as shown in i3 forum reference configuration (see Fig. 1 below).

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

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

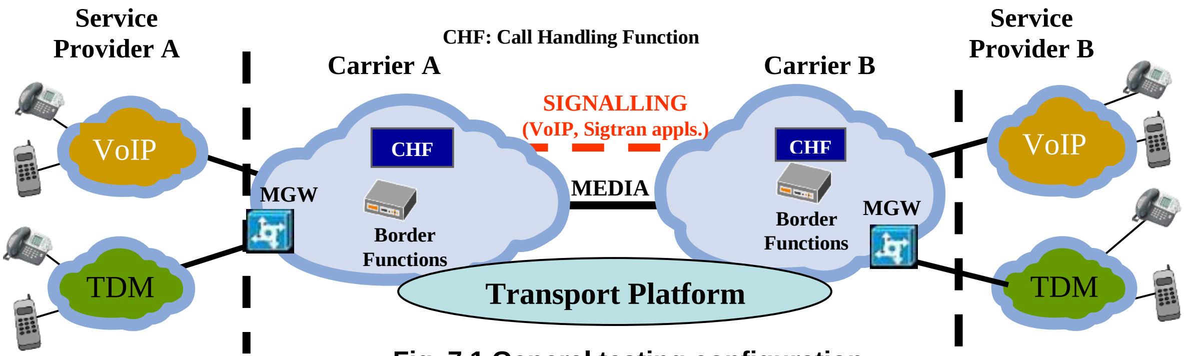

Fig. 7.1 General testing configuration

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 (ITU-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. In most cases a media re-INVITE is necessary. If the delay between 200 OK to initial INVITE and re-INVITE is too long fax call may fail.

8 Real interconnection configurations

Real interconnection configurations are usually different from reference configuration. Carriers delivering interconnection services use Least Cost Routing systems that choose the route between two destinations basing on interconnection rates. To maximize profit usually the cheapest routes are chosen. The resulting configuration may contain more than two carrier networks using IP or TDM technology.

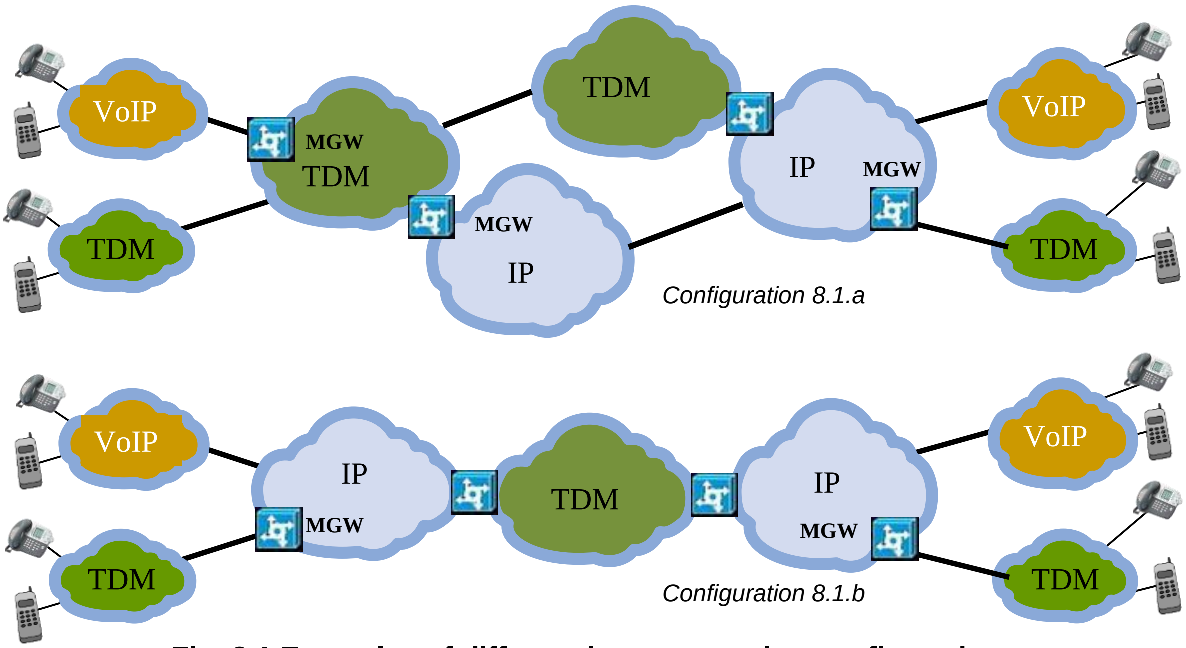

Fig. 8.1 Examples of different interconnection configurations

9 Problems identified in real interconnection configurations

Problems described below concern mainly the configuration when originating and terminating IP fax endpoints are connected to domestic VoIP service provider networks, which, in turn, are interconnected by mixed IP/TDM carrier networks.

9.1 Different call capabilities negotiated in SIP and in T.30.

In this case, the call starts as a normal VoIP call using G.711 mode. Then, the receiving endpoint re-INVITEs to T.38 with T38MaxBitRate=14400. As there is a TDM segment in the interconnection route, the re-INVITE is not propagated to the calling endpoint. The receiving Media Gateway Control Function answers OK with T38MaxBitRate=14400. After some time, the emitting Media Gateway Control Function sends re-INVITE to the calling endpoint and receives OK answer with T38MaxBitRate=9600. Finally, between calling endpoint and its border functions the call is set up with lower maximum speed and other modulation than between called endpoint and its border functions.

This can be a reason for buffer overflow of the endpoint that declared lower speed because the other side may send data with greater speed.

Afterwards, both endpoints start T.30 negotiations using media layer. The called endpoint sends DIS offering all types of modulation with maximum speed V.17/14400 bps. The calling endpoint sends DCS and V.17/14400 bps offer. Finally, on media layer the call is setup with V.17/14400 that offers greater speed than negotiated by SIP/SDP.

It can also be the reason of failures during training and during image transmission as well as during post-image T.30 message exchange. Probably wrong demodulator type (V.27 instead of V.29) is used somewhere on the route.

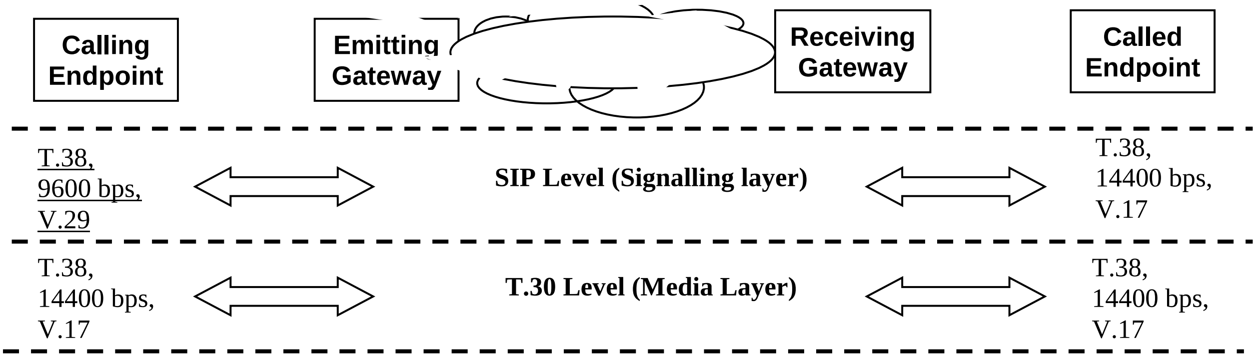

Fig. 9.1 Different call parameters negotiated on different levels.

9.2 Different FoIP modes are used by each endpoint (G.711 and T.38).

Please refer to Figure 8.1b and 9.1. When there is a TDM segment on the route (the cloud in figure 9.1), the re-INVITE generated by the called endpoint or receiving gateway is not forwarded to the calling terminal. If the emitting gateway does not re-INVITE the Calling Endpoint to T.38, the calling side remains in G.711 mode while called terminal and receiving gateway switch to T.38. In the first call phase, T.30 negotiations can reach “no return point” and, when called side switches to T.38 mode, its T.30 machine starts negotiations again while the calling endpoint has finished it and is ready to send image.

It may result in the break on media layer in any segment of the route. Then called endpoint tries to send DIS but it cannot reach the calling endpoint. The call fails.

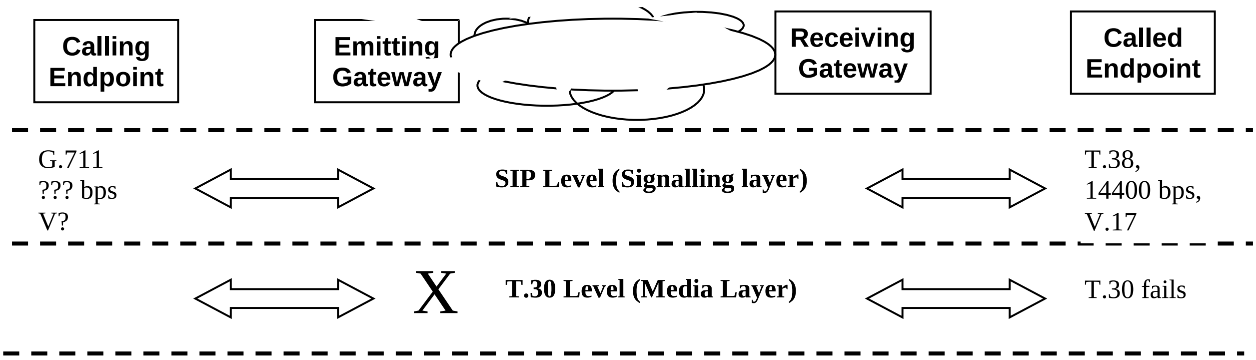

Fig. 9.2 Break in signalling path. Different FoIP modes on both ends. No media flow.

9.3 Media path not set up after re-INVITE.

This failure appears relatively often. Between correct calls only some number of such calls looks to be properly set up on signalling level but media path is not set up end-to-end or is set up in one direction only. Of course such call must fail. Such problem may be caused by very long delay between initial INVITE and 200 OK on the calling side. Often called endpoint re-INVITES to T.38 mode before calling endpoint receives 200 OK to its initial INVITE. It may be caused by long time of SIP to C7 and back to SIP conversion. During this time both T.30 machines can be in different state.

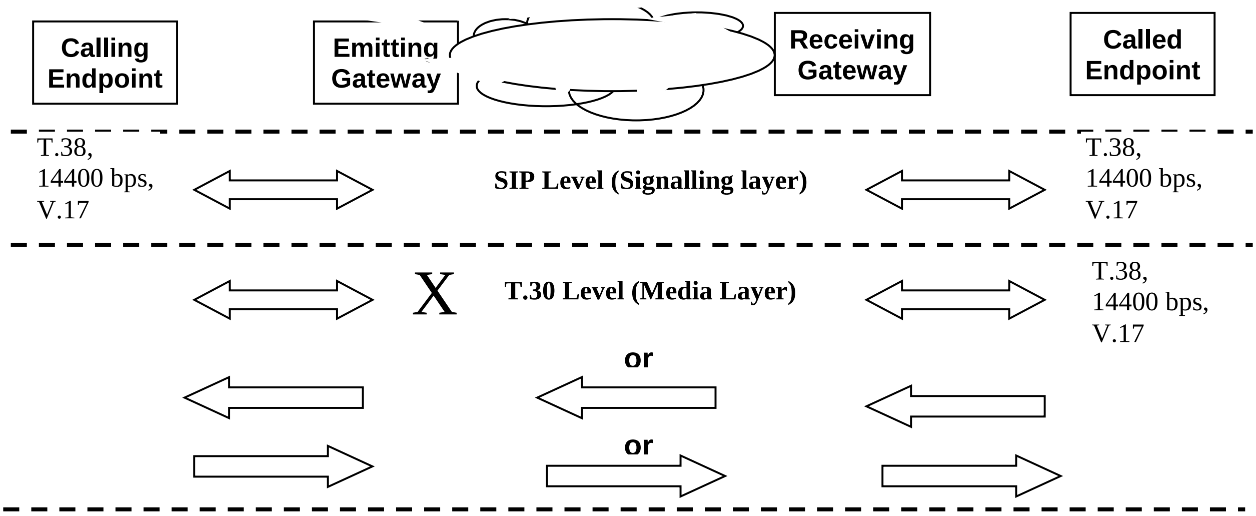

Fig. 9.3 No media flow or only one direction flow.

9.4 Training failures and V.21 events recognition problems

These failures seem to appear because of incorrect media conversion from the analog signal in a TDM segment to a digital signal in an IP segment. These kinds of errors can lead to bad recognition of training signal or T.30 messages like CFR, FTT, EOF and MFC.

9.5 Low-rate codec used in one interconnection segment.

In the interconnection configuration presented below, it is possible that in an intermediate interconnection IP segment a low-rate codec may be used. As endpoint-to-endpoint signalling is split into several segments, any endpoint may not be aware that the media channel is not able to transmit fax signals correctly.

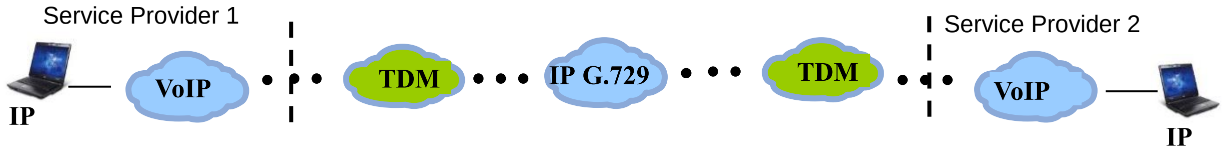

Fig. 9.4 Problem with low-rate codec.

Test calls during common test campaign have been sent using mainly IP fax terminals but also PSTN fax terminals. Test calls were exchanged between pairs of carriers that registered call flows on both sides using Wireshark. CopiaFacts server logs were also very useful in tracing the problems.

9.6 Problem summary

FoIP call failures appear in all four fax call phases:

1. Initial T.30 signalling.
   - SIP signalling on both sides OK but media channel not properly set up
   - T.30 signalling fails.
2. Training.
   - Training signal not recognized
   - No answer to training
   - Training fails.
3. Image transmission.
   - Transmission errors (appear usually in audio mode)
4. Post-image T.30 signalling.
   - Post -mage T.30 signalling fails

Most frequent were the failures during initial T.30 signalling. Next three error types appeared less frequently.

In interconnection configurations as presented on fig. 8.1 appear several disadvantageous problems. These problems are not taken into consideration in related standards. Some of them can introduce additional failure possibilities that make fax-over-IP interconnection unreliable. Most dangerous mechanisms are as follows:

1. Different call capabilities negotiated in SIP and T.30 caused by split of end-to-end signalling in two (or more) independent sessions – not SIP end-to-end.
2. Different FoIP modes used by each endpoint caused by split of end-to-end signalling in two (or more) independent sessions – not SIP end-to-end.
3. Large signalling delay because of SIP/CCS7/SIP conversion, even 10 seconds between initial INVITE and 200 OK that can lead to incorrect media channel setup or T.30 signalling failures.
4. Several TDM/IP/TDM conversions along a route that can lead to V.21 events recognition problems or training failures.
5. Analog signal continuity breaks that can lead to slips and incorrect TDM to IP conversion.
6. Low-rate codecs used in one of route segments that cannot be "seen" by endpoints.

10 Target technical recommendations for G3 FoIP connections

10.1 Testing Conclusions

On the basis of described above disadvantageous effects in real-life mixed TDM/IP interconnection the following conclusions can be drawn:

1. Network elements that introduce failures are:
   - Signalling gateways (SIP-to-C7-and-back long delay)
   - Media gateways (V.21, V.17 recognition, time slips etc.)
2. To avoid these failures, fax calls should be routed via all IP route.
3. If all IP route is not possible it should minimize the number of TDM/IP conversions.
4. The chosen IP route should guarantee QoS to eliminate transmission errors.

These conclusions lead to recommended solutions of FoIP calls reliability problem. Both solutions use the same principle but in different ways.
As a fax call is usually set up as a normal voice call, all routing entities, when choosing the route, are unaware that finally a fax will be transmitted. To assure appropriate routing:

1. All routing entities must know that the call that they are to set up will be a fax call.
2. They must be able to recognize and use this information to route the call appropriately.

General difference between two solutions is how the information that a call is a fax call is delivered to routing entities:

1. Smart routing based on calling UA fax capability.
   In this solution the information that the call to be set up will be a fax call is announced by originating endpoint by adding +sip.fax media feature tag in Contact header as specified in RFC 6913 [37]. RFC 6913 was submitted to the IETF by the SIP Forum FoIP Task Group as a result of the testing described here. This solution will use the fax media feature tag with RFC 3840 and RFC 3841 to implement "smart FoIP routing."

2. Smart routing using self-learning fax discovery.
   In this solution, new functionality is added to Border Call-Handling Functions. Self-learning discovery functionality listens to INVITEs, ACKs and re-INVITES searching for "m=image t.38" in SDP offer. If such offer is present the DN of the subscriber is put into FAX number database and the calls from/to this number are routed via all IP route.

10.2 Smart routing based on calling UA fax capability.

This solution is based on two existing IETF standards:

1. IETF RFC 3840 [35] "*Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)*" defining mechanisms by which a Session Initiation Protocol (SIP) User Agent can convey its capabilities and characteristics to other user agents and to the registrar for its domain. This information is conveyed as parameters of the Contact header field.

2. IETF RFC 3841 [36] "*Caller Preferences for the Session Initiation Protocol (SIP)*" describing a set of extensions to the Session Initiation Protocol (SIP) which allow a caller to express preferences about request handling in servers. These preferences include the ability to select which Uniform Resource Identifiers (URI) a request gets routed to, and to specify certain request handling directives in proxies and redirect servers. It does

so by defining three new request header fields, Accept-Contact, Reject-Contact, and Request-Disposition, which specify the caller's preferences.

and the new standard:

3. IETF RFC 6913 “Indicating Fax over IP Capability in the Session Initiation Protocol (SIP)” [37] elaborated by the SIP Forum FoIP Task Group that extends RFC 3840 [35] by adding new media feature tag “+sip.fax” with two possible values: “t38” and “passthrough”.

  - t38: means that the device supports the image/t38 media type as defined in RFC3326 [38] and implements ITU-T T.38 [6] for transporting the ITU-T T.30 [7] and ITU-T T.4 **Errore. L'origine riferimento non è stata trovata.** fax data over IP.
  - passthrough: means that the device supports the audio/pcmu and audio/pcma media types RFC4856 [39] for transporting ITU-T T.30 [7] and ITU-T T.4 [28] fax data using the ITU-T G.711 [8] audio codec. Additional implementation recommendations are in ITU-T V.152 [15] Sections 6 and 6.1.

By means of this media feature tag SIP User Agent can convey its capability. Then using RFC 3841 [36] SIP user agent can express its preference to be routed to a server with fax capabilities. Following examples presented in RFC 6913 explain how the solution works:

I. Registration:

```
REGISTER sip:example.com SIP/2.0
       Via: SIP/2.0/TCP bob-TP@example.com;branch=z9hG4bK309475a2
       From: <sip:bob-tp@example.com>;tag=a6c85cf
       To: <sip:bob-tp@pexample.com>
       Call-ID: a84b4c76e66710
       Max-Forwards: 70
       CSeq: 116 REGISTER
       Contact: <sip:bob-tp@example.com;transport=tcp>;+sip.fax="t38"
       Expires: 3600
SIP/2.0 200 OK
       From: <sip:bob-tp@example.com>;tag=a6c85cf
       To: <sip:bob-tp@example.com>;tag=1263390604
       Contact: <sip:bob-tp@example.com;transport=tcp>;+sip.fax="t38"
       Expires: 120
       Call-ID: a84b4c76e66710
       Via: SIP/2.0/TCP bob-TP@example.com;branch=z9hG4bK309475a2
       CSeq: 116 REGISTER
       Expires: 3600
```

II. Call setup

```
INVITE sip:UserY@example.com SIP/2.0
       From: sip:UserX@operator.com
        To: sip:UserY@example.com
        Accept-Contact: *;+sip.fax="t38"
        Content-Type: application/sdp
```

This solution however if used for interconnection routing requires additional clarification. RFC 3841 [36] describes only the way how media feature tags are used for incoming call inside target domain. The standard states in p.7.2:

“*A proxy compliant to this specification MUST NOT apply the preferences matching operation described here to a request unless it is the owner of the domain in the request URI, and accessing a location service that has capabilities associated with request targets.”*

Additional clarifications must be delivered in the form of an Implementation Guide that should define how border Call Handling Functions will use +sip.fax media feature tag for interconnection routing.

Generally, Call Handling Functions (SIP proxies) that will perform smart fax routing using +sip.fax media feature tag should:

- Examine INVITE contact header field and check if +sip.fax media feature tag is present,
- If the proxy receiving such INVITE is the owner of target URI domain it should behave in the way described in RFC 3841,
- If the proxy receiving such INVITE is not the owner of the target URI domain it should query its routing database and choose the next hop in the way that guarantees IP-based route, and so on, until the border of the target Service Provider network is reached.

Implementation of such guidelines would make to some extent an edge proxy non-compliant with RFC 3841, but that should not cause any harm.

10.3 Smart routing using self-learning fax discovery

The idea of this solution is similar to the solutions used for HD voice. High Definition voice requires wider bandwidth and cannot use TDM interconnections. A solution called self-learning HD voice discovery was developed by a vendor and was presented at the 9th IMS World Forum in 2012 by a Service Provider.

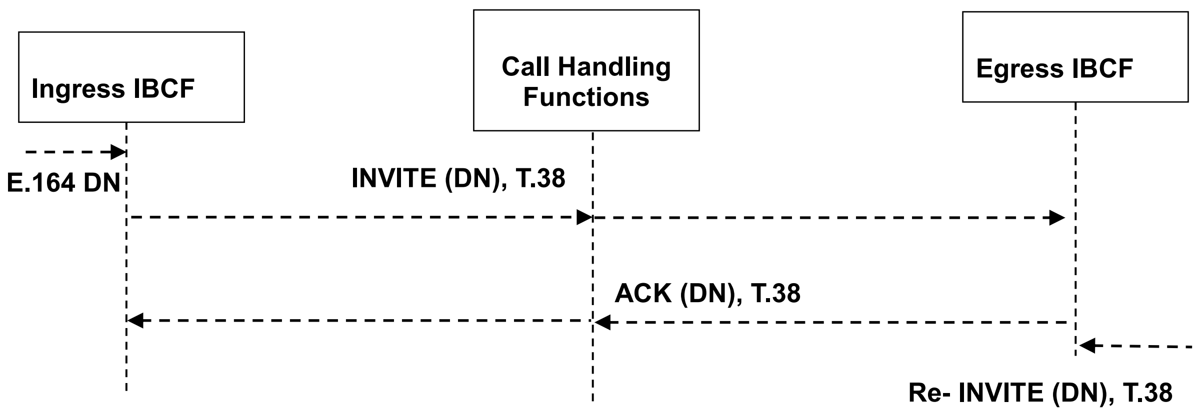

Fig. 10.1 Self-learning fax learning solution

Similar solution with a self-learning routing table could be used for Fax over IP. The way it should work is as follows:

- CHF listens to signaling messages coming from ingress and egress IBCF and search for "image/t38" in SDP media lines. if T.38 is included in INVITE and/or in re-INVITE and if yes add the DN to *Fax Number Table*
- If T.38 is not included in INVITE and/or in re-INVITE add DN to *Non Fax Number Table*
- During each call set up CHF checks if calling or called DN is in *Fax Number Table*. If yes, the call is routed via all IP route. If no, call can be routed via standard routing mechanism.

It should also use "aging" mechanism as follows:

- If DN from *Fax Number Table* N consecutive times does not support T.38 then DN is moved to *Non Fax Number Table*
- If DN appears again with t38 offer then it is moved back to *Fax Number Table*.

This solution has the capability to work also with TDM-originated fax calls. Even if the first fax call failed, the second call should have succeeded because, during the first re-INVITE, the Directory Number of at least one endpoint was added to the *Fax Number Table*.

10.4 Advantages and disadvantages of both solutions

1. Smart routing based on calling UA fax capability media feature tag that is defined in RFC 6913 [37], IETF RFC 3840 [35], and IETF RFC 3841 [36] standards.

Advantage:

- Simple algorithm and simple implementation

Disadvantage

- Implementation Guide necessary.
- Implementation requires changes in many terminals and in Call Handling Functions
- Implementation in user equipment that is out of carrier control

2. Smart routing using self-learning fax discovery

Advantages:

- No standard changes are necessary
- Implementation introduces changes only in Call Handling Functions (fewer entities)
- This solution could also work for TDM-originated Fax calls
- All devices requiring changes are under carrier control

Disadvantages:

- Complex algorithm causing additional signaling-processing delay
- Additional resources in Call Handling Functions are necessary (memory, processor capability).

11 Appendix A. Testing description

Fax over IP testing has been performed together by i3 forum carriers and SIP Forum members. The goal of the testing was to find and eliminate FoIP failures caused by international interconnection networks. Fax servers used for testing were provided by Copia International and Commetrex to testers. Originating and terminating endpoints were both identical CopiaFacts/Bladeware fax servers. In this way all possible terminal incompatibilities were eliminated. All failures identified during testing were introduced by networks. As most of the terminals were connected directly to the edge of carrier networks most of failure reasons were in interconnection network that are under carrier control. Interconnection testing configurations

Testing was performed mainly in private oriented interconnection configuration using real production carrier environment. For reference smaller amount of tests has been performed in public-oriented configuration via Public Internet.

11.1 Private-oriented Interconnection

In this configuration, testing endpoints were connected to domestic service-provider networks or to the edge of carrier interconnection networks. PSTN fax machines were also used for reference.
Testing results were elaborated on the basis of Wireshark call traces and Copia Logs analysis.
During the first phase of testing, common types of failures were identified and described. Reference PSTN fax calls showed that the success rate of PSTN originated fax calls were higher than IP originated calls.
During second testing, phase call traces from originating and terminating endpoints were compared to search for the reason of failures.

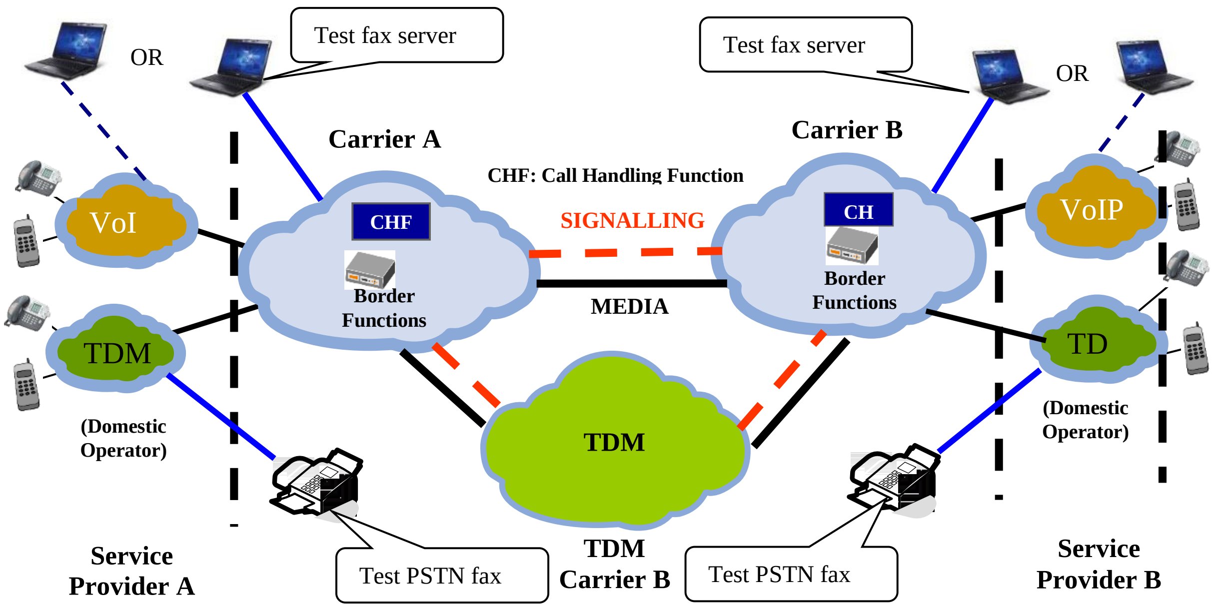

Fig. 11.1 Testing in private oriented configuration

Test calls in this configuration have been sent between endpoints located in Europe, North America, Hong Kong and Australia. In most cases, it was not possible to determine the exact route of each call. However, in most cases it was sure that mixed TDM/IP interconnection networks had been used.

11.2 Public-oriented Interconnection

In this configuration, two fax servers were connected to the Public Internet and they had a public IP address. Each of the peering fax servers routed the call to the known public address of a partner. There were neither call-handling functions nor border gateways along the route, only pure IP routing. It is worth noting that this is an ideal configuration which is different from the existing switched fax service.

Test calls were performed in T.38 mode, as well as in G.711 mode. Test calls have also been set up between Europe and North America, Hong Kong and Australia. RTD was measured ranging between 130 ms. up to 335 ms.

T.38 calls were setup with T.38 offer in the initial INVITE, as well as with initial G.711 offer and re-INVITE to T.38 mode.

11.3 Call Control Protocol

SIP/SDP protocol was used with very simple SDP session description containing only mandatory parameters. The only non-standard attribute was "silenceSupp:off- - - -" but it was supported by both endpoints.

Testing results with Wireshark traces were described in i3 forum and SIP Forum "Testing Report".

11.4 Testing results in private-oriented configuration

Most of the calls for entire testing campaign have been evaluated and following most frequent failure types have been identified:

Initial T.30 signalling errors
Call is set up correctly. SIP signalling on both sides OK.

- DIS sent by called endpoint but not heard by the calling one.
- DIS sent by called endpoint and heard by the calling one. DCS and sometimes training sent by calling endpoint but not heard by the called one.

In such cases media channel seems not to be set up or set up in one direction only.

Training errors
Call is set up correctly. SIP signalling on both sides OK.

- DIS/DCS exchange OK. FTT sent after several seconds delay and heard. BYE from calling terminal.
- DIS/DCS exchange OK. Training sent, CFR sent by called endpoint but not heard by the calling one.

Image transmission errors
Call is set up correctly. SIP signalling on both sides OK. T.30 signalling is successful and image sending starts.

- Image sent but not received. Post-image signalling sent but not received.
- Image sent and partially received. Call disconnected during image transmission.
- Image sent and received. Image with errors. EOP and RTN sent.

Post image T.30 signalling error
Call is set up correctly. SIP signalling on both sides OK. T.30 signalling is successful and complete image is sent.

- EOP sent by the calling endpoint but not received by the called one.
- EOP sent by the calling endpoint and received by the called one. MCF sent by the called endpoint and not received by the calling one.

11.5 Testing results in Public-oriented configuration

The number of calls performed in public-oriented configuration via Public Internet was significantly smaller. It is not possible to draw any statistical conclusions on the basis of such number of calls. However, the types of failures identified during the private oriented configuration testing phase never appeared.

All calls in T.38 mode were successful. About 20% of calls in G.711 mode failed because of burst packet loss. Usually 2 or 3 burst packet loss lasting about 30 ms. would terminate a FoIP session.

12 Appendix B. Setting up fax over IP calls recommendations

12.1 General recommendations

12.1.1 ECM and redundancy

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 when packet loss is significant. On high-quality networks with negligible packet loss, UDPTL high-speed redundancy is unnecessary. Where calls originate or terminate in the PSTN, ECM may still be desirable to correct errors introduced at the analog end.

ECM can be used if both terminals negotiate its use. *If ECM is enabled it is recommended to set redundancy for high-speed (image) not greater than 1*.

12.1.2 Support of “user=phone” in SDP offer.

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. User=phone should never be interpreted that sending a fax in the call is impossible.

12.1.3 Cisco fax protocol

Cisco FoIP protocol is sometimes supported by FoIP terminals but it is more and more rarely used and considered obsolete. It should be always disabled in fax terminals because it can become active during the call when T.30 signalling in audio or T.38 mode and interrupt it.

Cisco proprietary Named Signalling Events should be neither declared nor used. When one gateway or endpoint tries to switch to Cisco NSE pass-through or relay which the other side does not support it usually causes the call to fail.

12.2 Setting up fax call in audio mode

12.2.1 Use of ITU-T V.152

When fax call is set up in audio mode what means that a voice codec is to be used, it is recommended to use ITU –T V.152 [15] standard. V.152-compliant gateways enter this mode when [*a=gpmd: vbd=yes parameter]* has been mutually agreed and appropriate VBD stimuli are detected.

VBD mode assumes 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 is 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 [20] are disabled;
e. Echo cancellers as described in ITU-T G.168 [21] 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].

It is recommended to implement ITU-T V.152 if FoIP calls are to be set up in audio mode.

12.2.2 Fax tones transport via IP.

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 in IP network will not be sufficient for in-band tone transport. In this case the call may fail in the very beginning or during image transmission.

It is strongly recommended to negotiate and use DTMF/telephony events relay method as described in RFC 2833 [12] */RFC 4733* **Errore. L'origine riferimento non è stata trovata.***/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*
/CED	*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* [32] *is supported*

12.3 Setting up fax call in fax relay (T.38) mode

12.3.1 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 random burst packet loss 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. On high-quality routes, level 1 for low speed and level 0 for high speed is sufficient.

12.3.2 Transport

There are three following transport stacks that can be used in fax-relay mode as defined in ITU-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 to increase the chances of interoperability.

It is recommended to use UDPTL/UDP stack as FoIP transport protocol.

12.3.3 Fax training mode

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

It is recommended to use transferred-TCF mode of training. In this mode, the training signal is passed end-to-end, so if error sources are cumulative, such as occur when there are multiple TDM segments, the errors will accumulate in the TCF, causing the modem training to be at the lowest data rate that can be sustained over the entire call route.

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

12.3.4 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.

12.3.5 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-I10-120412 "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 re-INVITE 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.