SIP Real-time Fax Call Flow Examples
And
Best Current Practices

1. Abstract
The Session Initiation Protocol (SIP) allows the establishment of real-time Internet fax communications. Real-time facsimile communications over IP may follow 2 modes of operation: T.38 fax relay as defined by the ITU-T T.38 recommendation or fax pass-through.

This document clarifies the options available to Internet telephony gateway vendors to handle real-time fax calls using SIP. While our primary focus is to address the more reliable real-time T.38 Group 3 fax mode, we briefly cover the fax pass-through mode to enable fallback operations and super G3 fax communications using SIP. We also give examples of SIP call flows for real-time Internet fax gateways or SIP proxy redirect servers. Elements in these call flows include SIP User Agents, SIP Proxy Servers, and Gateways to the PSTN (Public Switch Telephone Network).

This document introduces best current practices for SIP T.38 fax and SIP fax pass-through sessions. A session starts with audio capabilities, and, upon fax tone detection, T.38 fax capabilities are negotiated; upon successful negotiation, the session continues with fax capabilities and the media termination hosts exchange T.38 Internet fax packets. The T.38 fax call scenarios include various aspects of the call sequence: the detection of fax transmission, the usage of the T.38 session description attributes, the optional
fallback into fax pass-through mode and the session termination. The fax pass-through call scenarios involve some specific SDP media attributes to enable proper fax transmission. Fax transmission can be detected by the receiving side, the emitting side or both (in the latter case, a ‘glare’ effect may appear). This document only covers the case when the fax transmission is detected by the receiving side: it is the most common practice and the other cases do not represent any particular challenges and are therefore left for future discussions. Call flow diagrams and message details are shown. A list of IANA defined SDP attribute names for T.38 is summarized in section 7.

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2. Conventions used in this document

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC-2119 [2].

3. Overview

The Session Initiation Protocol (SIP) is defined by RFC2543 [3] and T.38 is the ITU-T Recommendation defining the procedures for real-time Group 3 facsimile communications over IP networks [4]. This document deals with the handling of real-time fax communications using the SIP protocol for T.38 fax and fax pass-through modes. For T.38, this document deals primarily with one transport protocol for the media: T.38 over UDP/UDPTL; T.38 fax packet transport over TCP using SIP session establishment can easily be extrapolated. These T.38 call flows were developed in the design of carrier-class SIP Telephony products supporting voice and real-time fax traffic. It is the hope of the authors that this document will be useful for the SIP community, SIP implementors of T.38 fax products, designers. Call flows shown in this document complement the main SIP call flow Internet-Draft [5]. These call flows are based on the current version 2.0 of SIP in RFC2543[3], the ITU-T T.38 Amendment 2 [6] and an ITU-T SG8 liaison to IANA. Note: IANA registration of the T.38 SDP attributes is currently a work in progress.

3.1. General Assumptions

Internet telephony gateways with T.38 support MAY pass various fax tones (T.30 CNG, CED, etc.) or flag sequence (like V.21 Preamble) in the RTP audio streams before those tones or flag sequence are detected. Once detected by the DSP resources, T.38 fax-capable devices switch from audio to fax mode and initiate a T.38 fax packet transmission.

These best current practices MAY apply to "Network Gateway", "Enterprise Gateway" and Internet-aware fax devices or terminals.

3.2 Legend for Message Flows

The legend defined [5] also applies in this document. Dashed lines (---) represent control messages that are mandatory to the call scenario. These control messages can be SIP or PSTN
signaling. Double dashed lines (===) represent media paths between network elements. Messages with parenthesis around name represent optional control messages. Messages are identified in the Figures as F1, F2, etc. This references the message details in the table that follows the Figure. Comments in the message details are shown in the following form: /* Comments. */

3.3 Changes to previous drafts

- Changes to draft-01
  The changes between the current draft and draft-01 are:
  - Major updates:
    + additions based on SIPit fax testing event 8/2001: added text regarding the negotiation of T38MaxBitRate in 4.2,
    + added more general text for the handling of real-time fax pass-through,
    + added of echo cancellation & silence suppression SDP attributes and usage recommendations in 6.2,
    + correction of the session version number in SDP when issuing re-INVITE with updated SDP information as per draft-ietf-sip-rfc2543bis-05,
    + started a section on the use of the OPTIONS method to determine the support of T.38 capabilities (work-in-progress),
  - Minor editorial updates (added tag fields in To and From header lines as well as Contact header lines as per draft-ietf-sip-rfc2543bis-05, added proper reference to T.38 Amendment 2 pre-published version, exact case matching for T38MaxBitRate, etc.)

- Changes to draft-00
  The major changes between draft-01 and draft-00 are listed below:
  - Major updates:
    + new text in fax handling section 4.2 based on sip list: clarified text for fax detection, capability negotiation and termination
    + additions of new successful call flows in section 5
    + additions of unsuccessful call flows in section 6
  - Added Record-Route header, modified Cseq numbers,
  - Added T38FaxMaxDatagram attribute in SDP and corrected the sample value for T38FaxMaxBuffer,
  - Minor editorial changes (e.g. removed references in Abstract).

4 Handling of real-time fax communication using SIP

The handling of real-time fax sessions using SIP involves the initial session establishment, the modification or negotiation of session capabilities upon fax detection, the optional fallback into fax pass-through mode and the session termination.

Some variants of these best current practices MAY apply depending on the nature or the configuration of Internet telephony gateways. Two distinct cases are considered in this document:
  - The Internet telephony gateway only supports T.38 real-time fax communications (by design or by configuration). In this case, the Internet fax gateway SHOULD initiate the SIP session with T.38
SDP capabilities (this is typically the case of Internet fax terminals, also called Internet-aware fax devices or the case of gateways statically configured to support T.38 fax calls only);

- The Internet telephony gateway supports voice and real-time fax communications. In this case, the Internet telephony gateway initiates the SIP session with audio capabilities, and, upon fax detection, the switchover to T.38 fax capabilities is triggered. The fax media connection may replace or be added to the audio connection depending on the target applications.

4.1 Internet telephony gateways and fax detection

Two possible ways exist for an Internet telephony gateway to detect a facsimile communication:

1. On the emitting gateway (the one sending the facsimile document), the T.30 calling tone (CNG) is detected. The CNG tone is however an optional signal [7];
2. On the receiving gateway, the V.21 Preamble flag sequence is detected. The Preamble is always present and follows the called station identification tone (CED) when CED is present. Note that even though the CED tone may be detected on the receiving gateway, since the fax CED tone is similar to V.25 ANS modem tones, it SHOULD NOT be used to trigger the fax switchover.

In conclusion, several conditions can trigger fax detection. Internet telephony gateways supporting T.38 real-time facsimile MUST support the detection of fax transmission on the V.21 Preamble flag sequence; Internet telephony gateways MAY support the detection of fax on CNG tone. The authors recommend that a facsimile transmission SHOULD be detected on the receiving Internet telephony gateway.

4.2 Internet telephony gateways and T.38 fax media connections

Upon successful fax detection, the description of the media connection(s) is modified using the SDP protocol [8] as described in SIP (see sections in chapter 5).

Several cases should be considered:
- In the case of Internet fax devices (supporting T.38 fax only communications), the SIP session is established with SDP description containing the T.38 media attributes; refer to section 5.1 for a sample call flow for Internet fax devices.
- In the case of Internet telephony gateways, the SIP/SDP session is modified and the T.38 fax media connection SHOULD replace the existing audio media connection. The SIP UAC triggering the modification of the session description MUST indicate all of its T.38 media attributes in SDP. This is the recommended behavior for gateways; refer to section 5.2 for a sample call flow illustrating this scenario.

Some implementations MAY choose to add the fax media connection to the existing audio connection and in that case, they SHOULD mute the
audio connection while fax is transmitted. Refer to section 5.3 for a sample call flow illustrating this optional scenario.

Note:
Among the SDP T.38 media attributes being exchanged after fax detection, implementors should particularly pay attention to T38MaxBitRate, the maximum T.38 bit rate for the media connection. It is recommended to implement the following rule when responding to a maximum bit rate proposal. If the remote peer’s SDP description contains a maximum fax bit rate of X and the local host is set to a maximum bit rate of Y (either by fax protocol detection or configuration), the local host MUST propose a maximum fax rate of min (X,Y) in the SIP response to the re-INVITE in order to facilitate the negotiation. This note is exemplified in section 5.2.1. in messages F11-F14. Failures to implement this rule MAY imply call teardown in some cases.

4.3 SIP session and fax termination

Upon completion of the fax transmission, any audio connection successfully established prior to fax detection SHOULD be restored or the session MUST be terminated.

4.4 Internet telephony gateways and fax pass-through mode

This section deals real-time facsimile communications over IP using fax pass-through instead of T.38 fax relay. In this mode, the facsimile communication is handled as a PCM audio call (PCMA/PCMU as specified in ITU-T recommendation G.711).

The fax pass-through mode is important to prevent call failures, for example in cases when one of the SIP communication peers does not support T.38. For Internet telephony gateways with support for PCM/G.711 audio but no support for T.38 real-time fax, it is recommended to switch the session to fax pass-through mode. Internet telephony gateways SHOULD handle the fall back mode to fax pass-through by recognizing the SDP T.38 connection and proposing to switch to a new audio connection. The new audio connection SHOULD have the following characteristics: at a minimum, specify PCM G.711 codec, silence suppression OFF, echo cancellation ON. The telephony gateway that originated the failed T.38 re-INVITE SHOULD initiate the subsequent re-INVITE to fax pass-through mode. Refer to section 6.2 for a sample call flow illustrating this scenario.

4.5 Responses to SIP OPTIONS request

Suggestion currently logged in section 8.4, just a placeholder for now; <<work in progress >>

4.6 Beyond Internet telephony gateways
This document may also serve to vendors of SIP redirect servers or other SIP UA types like SIP phones. Such SIP entities may recognize the fax nature of the sessions and redirect the flows to the proper real-time fax-capable devices or to other fax service providers.

5 Successful SIP T.38 fax Call Scenarios

5.1 Internet fax device → fax only support

This section represents a SIP call flow for a T.38 fax session between 2 Internet fax terminals. It is also applicable to Internet telephony gateways when telephony ports are statically configured to support fax only (case of analog IP telephony gateways where a specific port is connected to a fax terminal). A session starts with fax capabilities (t.38/udptl). The mechanism for supporting T.38 in SIP & SDP is detailed in T.38 Annex D [9], a temporary document that proposes an amendment to T.38 describing SIP call establishment procedures.

5.1.1 Sequence Diagram

<table>
<thead>
<tr>
<th>Internet Fax UA</th>
<th>Proxy</th>
<th>IFTGW UA</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fax</td>
<td></td>
<td></td>
</tr>
<tr>
<td>emitted</td>
<td>F1 INVITE</td>
<td></td>
</tr>
<tr>
<td></td>
<td>-------------------&gt; F2 INVITE</td>
<td></td>
</tr>
<tr>
<td></td>
<td>F3 100 Trying</td>
<td>F4 100 Trying</td>
</tr>
<tr>
<td></td>
<td>&lt;-------------------</td>
<td>&lt;-------------------</td>
</tr>
<tr>
<td></td>
<td>Li/Mule</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>F5 180 Ringing</td>
<td></td>
</tr>
<tr>
<td></td>
<td>-------------------&gt;</td>
<td>F7 200 OK</td>
</tr>
<tr>
<td></td>
<td>F6 180 Ringing</td>
<td>F8 200 OK</td>
</tr>
<tr>
<td></td>
<td>&lt;-------------------</td>
<td>&lt;-------------------</td>
</tr>
<tr>
<td></td>
<td>F9 ACK</td>
<td>F10 ACK</td>
</tr>
<tr>
<td></td>
<td>-------------------&gt;</td>
<td></td>
</tr>
<tr>
<td></td>
<td>T.38/UDPTL Fax Flow Established</td>
<td></td>
</tr>
<tr>
<td></td>
<td>&lt;=======================================&gt;</td>
<td></td>
</tr>
<tr>
<td></td>
<td>End of fax detected</td>
<td></td>
</tr>
<tr>
<td></td>
<td>-------------------&gt;</td>
<td></td>
</tr>
<tr>
<td></td>
<td>F11 BYE</td>
<td>F12 BYE</td>
</tr>
</tbody>
</table>
5.1.2 Message Details

F1 INVITE I.FAX UA -> PROXY

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ifax.here.com:5060
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab111
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ifax.here.com
CSeq: 17 INVITE
Content-Type: application/sdp
Content-Length: 320

v=0
o=ifaxgw1 2890846527 2890846527 IN IP4 ifax.here.com
s=Session SDP
c=IN IP4 ifaxmg.here.com
t=0 0
m=image 15002 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF

F2 INVITE PROXY -> IFTGW UA

INVITE sip:+1-650-555-2222@iftgw.there.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ifax.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab111
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ifax.here.com
CSeq: 17 INVITE
Content-Type: application/sdp
Content-Length: 320

v=0
o=ifaxgw1 2890846527 2890846527 IN IP4 ifax.here.com
s=Session SDP
c=IN IP4 ifaxmg.here.com
t=0 0
m=image 15002 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPFEC
a=T38FaxUdpEC:t38UDPRedundancy

F3 (100 Trying) PROXY -> I.FAX UA

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ifax.here.com:5060
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ifax.here.com
CSeq: 17 INVITE
Content-Length: 0

F4 (100 Trying) IFTGW UA -> PROXY

SIP/2.0 100 Trying
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ifax.here.com:5060
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ifax.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Length: 0

F5 180 Ringing IFTGW UA -> PROXY

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ifax.here.com:5060
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ifax.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Length: 0
F6 180 Ringing PROXY -> I.FAX UA

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ifax.here.com:5060
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ifax.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Length: 0

F7 200 OK IFTGW UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ifax.here.com:5060
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ifax.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890844527 2890844527 IN IP4 iftgw.there.com

Li/Mule

SIP T.38 Call Flows October 2001

s=Session SDP
c=IN IP4 iftgw.there.com
t=0 0
m=image 49172 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPRedundancy

F8 200 OK PROXY -> I.FAX UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP ifax.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ifax.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
v=0
o=faxgw1 2890844527 2890844527 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftgw.there.com
t=0 0
m=image 49172 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPRedundancy

F9 ACK I.FAX UA -> PROXY

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ifax.here.com:5060
Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab11

SIP T.38 Call Flows October 2001

To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ifax.here.com
CSeq: 17 ACK
Content-Length: 0

F10 ACK PROXY -> IFTGW UA

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ifax.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ifax.here.com
CSeq: 17 ACK
Content-Length: 0

/* T.38 fax streams are established. */

F11 BYE I.FAX UA -> PROXY

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP ifax.here.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ifax.here.com
CSeq: 18 BYE
Content-Length: 0

F12 BYE PROXY -> IFTGW UA

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP ifax.here.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ifax.here.com
CSeq: 18 BYE
Content-Length: 0

SIP T.38 Call Flows October 2001

F13 200 OK IFTGW UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ifax.here.com:5060
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde34
Call-ID: 1717@ifax.here.com
CSeq: 18 BYE
Content-Type: application/sdp
Content-Length: 0

F14 200 OK PROXY -> I.FAX UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP ifax.here.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ifax.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde34
Call-ID: 1717@ifax.here.com
CSeq: 18 BYE
Content-Type: application/sdp
Content-Length: 0

5.2 SIP T.38 fax call Â» fax stream replaces voice stream

This section represents the SIP call flow for a T.38 fax session between 2 Internet telephony gateways (fax detection on terminating
A session starts with audio capabilities, and then the session is modified to switch to t38 fax mode (t.38/udptl).

Scenario:
1. A SIP INVITE is sent to the called party requesting a voice connection per RFC2543; our scenario involves 1 SIP proxy. An audio connection is established,
2. Upon detection of Preamble by the terminating gateway, a SIP re-INVITE request is sent to the emitting gateway to modify the parameters of the session to allow a T.38 fax media connection. This INVITE request contains new SDP information detailing T.38 capabilities,
3. Upon successful acknowledgments, T.38 IFP fax packets are sent/received on UDP ports, which may be the same UDP ports used for audio RTP,
4. Once the fax transmission is terminated, audio capabilities are restored or the call is terminated.

5.2.1 Sequence Diagram
In our example, we illustrate the fact that Internet telephony gateways may use multiple network interfaces for signaling and/or media streaming or one network interface with multiple IP addresses. Typically, our example shows:
- one interface for SIP signaling
  (ingress gateway = ift.here.com, egress gateway = iftgw.there.com)
- one or multiple interface(s) for media transport:
  (ingress gateway = iftmg.here.com, egress gateway = iftmg.there.com)
A proxy is acting as a pure SIP signaling proxy (obelix.wcom.com).

<table>
<thead>
<tr>
<th>IFT UA</th>
<th>Proxy</th>
<th>IFTGW UA</th>
</tr>
</thead>
<tbody>
<tr>
<td>F1 INVITE</td>
<td>F2 INVITE</td>
<td></td>
</tr>
<tr>
<td>F3 100 Trying</td>
<td>F4 100 Trying</td>
<td></td>
</tr>
<tr>
<td>F6 180 Ringing</td>
<td>F5 180 Ringing</td>
<td></td>
</tr>
<tr>
<td>F8 200 OK</td>
<td>F7 200 OK</td>
<td></td>
</tr>
<tr>
<td>F9 ACK</td>
<td>F10 ACK</td>
<td></td>
</tr>
<tr>
<td>Both Way RTP Media Established</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Fax emitted

---

F11 INVITE detected

---

F12 INVITE

---

F13 200 OK

---

F14 200 OK

---

F15 ACK

---

F16 ACK

---

T.38/UDPT Fax Flow Established

---

End of fax detected

---

F17 INVITE detected

---

F18 INVITE

---

F19 200 OK

---

F20 200 OK

---

F21 ACK

---

F22 ACK

---

Both Way RTP Media Re-Established

---

F23 BYE

---

F24 BYE

---

F25 200 OK

---

End of fax detected
5.2.2 Message Details

F1 INVITE  IFT UA -> PROXY

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 146

V=0
O=IFAXTERMINAL01 2890844527 2890844527 IN IP4 ift.here.com
S=Session SDP
C=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 INVITE  PROXY -> IFTGW UA

INVITE sip:+1-650-555-2222@iftgw.there.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 146

V=0
O=IFAXTERMINAL01 2890844527 2890844527 IN IP4 ift.here.com
S=Session SDP
C=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F3 (100 Trying)  PROXY -> IFT UA

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Content-Length: 0

F4 100 Trying IFTGW UA -> PROXY
SIP/2.0 100 Trying
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com

F5 180 Ringing IFTGW UA -> PROXY
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Length: 0

F6 180 Ringing PROXY -> IFT UA
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Length: 0

F7 200 OK IFTGW UA -> PROXY
SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 150

v=0
o=IFAXTERMINAL01 2890844527 171089 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F8 200 OK PROXY -> IFT UA

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060
Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 ACK
Content-Type: application/sdp
Content-Length: 150

v=0
o=IFAXTERMINAL01 2890844527 171089 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F9 ACK IFT UA -> PROXY

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 ACK
Content-Length: 0

F10 ACK PROXY -> IFTGW UA

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com;
/* RTP streams are established. The CNG fax tone is sent in-band if it is present. The receiving side IFT gateway DSP detects the Preamble sequence. A new UDP port is open on IFTGW for T.38 IFP packets and the IFTGW signals the switch over to fax mode by send a re-INVITE with the new UDP port in the SDP.
Note: the IFTGW has a T38MaxBitRate configured to 9600 */

F11 INVITE IFTGW UA -> PROXY

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890844527 171090 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=image 49172 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:9600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPFEC
a=T38FaxUdpEC:t38UDPRedundancy

F12 INVITE PROXY -> IFT UA

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>

From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp

Content-Length: 320
v=0
o=faxgw1 2890844527 171090 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=image 49172 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:9600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPFEC
a=T38FaxUdpEC:t38UDPRedundancy

/* Note: the IFT has a T38MaxBitRate configured to 14400 but since it received a T38MaxBitRate proposal of 9600, it replies with T38MaxBitRate = min (9600,14400) */

F13 200 OK IFT UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 320
v=0
o=faxgw1 2890846527 2890846528 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=image 15002 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:9600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
Li/Mule

SIP T.38 Call Flows October 2001

F14 200 OK PROXY -> IFT UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com; user=phone; tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com; user=phone; tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890846527 2890846528 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=image 15002 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:9600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPRedundancy

F15 ACK IFTGW UA -> PROXY

ACK sip:+1-650-555-2222@obelix.wcom.com; user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com; user=phone; tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com; user=phone; tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 ACK
Content-Length: 0

F16 ACK PROXY -> IFT UA

ACK sip:+1-650-555-2222@obelix.wcom.com; user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
/* T.38 fax transmission established both ways */

/* Then, the end of the fax transmission is detected on ingress side and sent to the egress side (IFTGW). IFTGW initiates the switch back to voice communication */

F17 INVITE IFTGW UA -> PROXY

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com;
      maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 181

v=0
o=faxgw1 2890844527 171091 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F18 INVITE PROXY -> IFT UA

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
      maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 181
v=0
o=faxgw1 2890844527 171091 IN IP4 iftgw.there.com
s=Session SDP

SIP T.38 Call Flows October 2001

Li/Mule

F19 200 OK IFT UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
o=faxgw1 2890844527 2890844529 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F20 200 OK PROXY -> IFTGW UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
o=faxgw1 2890844527 2890844529 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
F21 ACK IFTGW UA -> PROXY

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 ACK
Content-Length: 0

F22 ACK PROXY -> IFT UA

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 ACK
Content-Length: 0

/* Voice streams re-established both ways */

F23 BYE IFT UA -> PROXY

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Length: 0

F24 BYE PROXY -> IFTGW UA

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ift.here.com
This section represents the SIP call flow for a T.38 fax session between 2 Internet telephony gateways. It shows the detection of the fax communication on terminating or receiving side. A session starts with one audio stream in SDP, and, upon detection of fax, an additional fax stream is added to the session while the voice stream is muted. After the fax communication is terminated, the fax stream is terminated and the voice stream is restored.

Scenario:
1. A SIP INVITE is sent to the called party requesting a voice connection per RFC2543; our scenario involves 1 SIP proxy. An audio connection is established,
2. Upon detection of Preamble by the terminating gateway, a SIP reINVITE request is sent to the emitting gateway to modify the parameters of the session to add a T.38 fax connection. The connection data is specified per media description and not at the session-level. This allows the audio stream and t.38 fax stream to be controlled independently (i.e., placing the audio on hold while the t.38 fax transmission is active). Also note that the t.38 fax

5.3 SIP T.38 fax call û fax stream added to voice stream

CSeq: 20 BYE
Content-Length: 0

F25 200 OK IFTGW UA -> PROXY
SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde24
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Type: application/sdp
Content-Length: 0

F26 200 OK PROXY -> IFT UA
SIP/2.0 200 OK
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde24
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Type: application/sdp
Content-Length: 0
media stream has been added to the session. This re-INVITE request contains SDP information detailing the additional T.38 capabilities,
3. It is suggested that the voice pipe be muted while the fax session is in progress. Upon successful acknowledgments, T.38 IFP fax packets are sent/received on different UDP ports than the one used for audio RTP,
4. Once the fax transmission is terminated, audio capabilities are restored or the call is terminated.

5.3.1 Sequence Diagram

<table>
<thead>
<tr>
<th>IFT UA</th>
<th>Proxy</th>
<th>IFTGW UA</th>
</tr>
</thead>
<tbody>
<tr>
<td>F1 INVITE</td>
<td>F2 INVITE</td>
<td></td>
</tr>
<tr>
<td>F3 100 Trying</td>
<td>F4 100 Trying</td>
<td></td>
</tr>
<tr>
<td>F6 180 Ringing</td>
<td>F5 180 Ringing</td>
<td></td>
</tr>
<tr>
<td>F8 200 OK</td>
<td>F7 200 OK</td>
<td></td>
</tr>
<tr>
<td>F9 ACK</td>
<td>F10 ACK</td>
<td></td>
</tr>
</tbody>
</table>

Fax emitted
Both Way RTP Media Established

| | F11 INVITE |
| | detected |
| F12 INVITE |
| F13 200 OK |
| F14 200 OK |
| F15 ACK |
| F16 ACK |

SIP T.38 Call Flows October 2001
T.38/UDPT Fax Flow Established
Voice connection muted

<======================================>

End of fax detected
-------->

End of fax detected

F17 INVITE
<-------------------

F18 INVITE
<-------------------

F19 200 OK
<-------------------

F20 200 OK
<-------------------

F21 ACK
<-------------------

F22 ACK
<-------------------

Both way RTP Flow restored
<======================================>

F23 BYE
<-------------------

F24 BYE
<-------------------

F25 200 OK
<-------------------

F26 200 OK

5.3.2 Message Details

F1 INVITE IFT UA -> PROXY

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>

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Content-Type: application/sdp
Content-Length: 146

v=0
F2 INVITE PROXY -> IFTGW UA

INVITE sip:+1-650-555-2222@iftgw.there.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com; user=phone; tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com; user=phone
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 146

v=0
o=IFAXTERMINAL01 2890844527 2890844527 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F3 (100 Trying) PROXY -> IFT UA

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com; user=phone; tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com; user=phone
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Content-Length: 0

F4 100 Trying IFTGW UA -> PROXY

SIP/2.0 100 Trying
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060

Li/Mule

SIP T.38 Call Flows October 2001
F5 180 Ringing IFTGW UA -> PROXY

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Length: 0

F6 180 Ringing PROXY -> IFT UA

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Length: 0

F7 200 OK IFTGW UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 150

v=0
o=IFAXTERMINAL01 2890844527 171089 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000

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SIP T.38 Call Flows October 2001

F8 200 OK PROXY -> IFT UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 150

v=0
o=IFAXTERMINAL01 2890844527 171089 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F9 ACK IFT UA -> PROXY

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 ACK
Content-Length: 0

F10 ACK PROXY -> IFTGW UA

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 17 ACK
Content-Length: 0

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SIP T.38 Call Flows October 2001

/* RTP streams are established. The CNG fax tone is sent in-band if it is present. The receiving side IFT gateway DSP detects the preamble. A new UDP port is open on IFTGW for T.38 IFP packets and the IFTGW signals the switch over to fax mode by send a re-INVITE with the new UDP port in the SDP */

F11 INVITE IFTGW UA -> PROXY

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com; user=phone; tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com; user=phone; tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890844527 171090 IN IP4 iftgw.there.com
s=Session SDP
t=0 0
m=audio 12322 RTP/AVP 0
c=IN IP4 0.0.0.0
m=image 49172 udptl t38
c=IN IP4 iftmg.there.com
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPFEC
a=T38FaxUdpEC:t38UDPRedundancy

F12 INVITE PROXY -> IFT UA

INVITE sip:+1-650-555-2222@obelix.wcom.com; user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com; user=phone; tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com; user=phone; tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>

Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890844527 171090 IN IP4 iftgw.there.com
s=Session SDP
t=0 0
m=audio 12322 RTP/AVP 0
c=IN IP4 0.0.0.0
m=image 49172 udptl t38
c=IN IP4 iftmg.there.com
a=T38FaxVersion:0
a=T38MaxBitRate:14400
F13 200 OK IFT UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890846527 2890846528 IN IP4 ift.here.com
s=Session SDP
t=0 0
m=audio 3456 RTP/AVP 0
c=IN IP4 0.0.0.0
m=image 15002 udptl t38
c=IN IP4 iftmg.here.com
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPRedundancy

F14 200 OK PROXY -> IFT UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 320
v=0
o=faxgw1 2890846527 2890846528 IN IP4 ift.here.com
s=Session SDP
t=0 0
m=audio 3456 RTP/AVP 0
c=IN IP4 0.0.0.0
m=image 15002 udptl t38
c=IN IP4 iftmg.here.com
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPRedundancy

F15 ACK IFTGW UA -> PROXY

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 ACK
Content-Length: 0

F16 ACK PROXY -> IFT UA

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060

/* T.38 fax transmission established both ways and the voice stream is muted */
/* Then, the end of the fax transmission is detected on ingress side and sent to the egress side (IFTGW). IFTGW initiates the switch back to voice communication */
INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 181

v=0
o=faxgw1 2890844527 171091 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F18 INVITE PROXY -> IFT UA

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Route-Record: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 181

v=0
o=faxgw1 2890844527 171091 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F19 200 OK IFT UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 150

v=0
o=faxgw1 2890844527 2890844529 IN IP4 ift.here.com
s=Session SDP
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F20 200 OK PROXY -> IFTGW UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 150

v=0
o=faxgw1 2890844527 2890844529 IN IP4 ift.here.com
s=Session SDP
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F21 ACK IFTGW UA -> PROXY

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 ACK
Content-Length: 0

F22 ACK PROXY -> IFT UA

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 ACK
Content-Length: 0

F23 BYE IFT UA -> PROXY

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Length: 0

F24 BYE PROXY -> IFTGW UA

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Length: 0

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F25 200 OK IFTGW UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde24
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Type: application/sdp
Content-Length: 0

F26 200 OK PROXY -> IFT UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde24
Call-ID: 1717@ift.here.com
5.4 SIP T.38 fax call » fax stream forced to replace voice stream

This section illustrates the following scenario:
1. A SIP INVITE is sent to the called party requesting a voice connection per RFC2543; our scenario involves 1 SIP proxy. An audio connection is established,
2. Upon detection of Preamble by the terminating gateway, a SIP reINVITE request is sent to the emitting gateway to modify the parameters of the session to add a T.38 fax connection. This re-INVITE request contains SDP information detailing the additional T.38 capabilities. It is also suggested that the voice pipe be muted while the fax session is in progress.
3. The emitting gateway does not have the resources to maintain both the audio & fax streams and replies by allowing the fax media stream. Upon successful acknowledgments, the audio stream is terminated and T.38 IFP fax packets are exchanged,
4. Once the fax transmission is terminated, audio capabilities are re-negotiated and the call is terminated.

5.4.1 Sequence Diagram
Refer to section 5.3.1; sequence diagram is unchanged.

5.4.2 Message Details
See section 5.3.2 for the complete set of messages. We only indicate the changed messages based on the emitting gateway's decline to support both voice and fax streams.

... F11 INVITE IFTGW UA -> PROXY

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890844527 171090 IN IP4 iftgw.there.com
s=Session SDP
t=0 0
m=audio 12322 RTP/AVP 0
c=IN IP4 0.0.0.0
m=image 49172 udptl t38
c=IN IP4 iftmg.there.com
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPFEC
a=T38FaxUdpEC:t38UDPRedundancy

F12 INVITE PROXY -> IFT UA

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp

Content-Length: 320

v=0
c=faxgw1 2890844527 171090 IN IP4 iftgw.there.com
s=Session SDP
t=0 0
m=audio 12322 RTP/AVP 0
c=IN IP4 0.0.0.0
m=image 49172 udptl t38
c=IN IP4 iftmg.there.com
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPFEC
a=T38FaxUdpEC:t38UDPRedundancy

F13 200 OK IFT UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 320

v=0
c=sess 2890846527 2890846528 IN IP4 ift.here.com
m=audio 0 RTP/AVP 0
m=image 15002 udptl t38
c=IN IP4 iftmg.here.com
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPRedundancy

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F14 200 OK PROXY -> IFT UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 320

v=0
c=sess 2890846527 2890846528 IN IP4 ift.here.com
m=audio 0 RTP/AVP 0
m=image 15002 udptl t38
c=IN IP4 iftmg.here.com
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
6. Unsuccessful Scenarios

6.1. Unsuccessful T.38 fax scenario – no support for T.38

This section represents a SIP T.38 call failure because the emitting gateway does not support T.38.

Scenario:
1. A SIP INVITE is sent to the called party requesting a voice connection per RFC2543; our scenario involves 1 SIP proxy. An audio connection is established.
2. Upon detection of Preamble by the terminating gateway, a SIP re-INVITE request is sent to the emitting gateway to modify the parameters of the session to allow a T.38 fax connection. This re-INVITE request contains SDP information detailing T.38 capabilities.
3. The emitting gateway rejects the re-INVITE with a 415 Unsupported Media Type response.
4. Upon reception of the reject, the call is terminated.

6.1.1. Sequence Diagram

The messages F1 to F12 are identical to section 5.2.
### 6.1.2. Message Details

Messages F1 to F12 are identical to section 5.2; refer to 5.2.

F13 415 Unsupported Media Type  IFT UA -> PROXY
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Content-Length: 0

F14 415 Unsupported Media Type  PROXY -> IFTGW UA

SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d010.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE

F15 BYE IFTGW UA -> PROXY

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde34
Call-ID: 1717@ift.here.com
CSeq: 19 BYE
Content-Length: 0

F16 BYE PROXY -> IFT UA

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d010.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde34
Call-ID: 1717@ift.here.com
CSeq: 19 BYE
Content-Length: 0

F17 200 OK IFT UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
6.2. Unsuccessful T.38 fax scenario — 606 response and G.711 fallback

This section represents an unsuccessful SIP T.38 fax call: the emitting gateway does not support T.38 and responds with a 606 Not Acceptable to indicate that some aspects of the session description are not acceptable. The terminating gateway may react by proposing a fallback to G.711 fax pass-through with special codec characteristics — echo cancellation ON, silence suppression OFF. The message details in this section make use of the generic SDP attributes ecan and silenceSupp, defined in RFC3108 [10].

Scenario:
1. A SIP INVITE is sent to the called party requesting a voice connection per RFC2543; our scenario involves 1 SIP proxy. An audio connection is established,
2. Upon detection of Preamble by the terminating gateway, a SIP reINVITE request is sent to the emitting gateway to modify the parameters of the session to allow a T.38 fax connection. This re-INvITE request contains SDP information detailing T.38 capabilities,
3. The emitting gateway rejects the re-INvITE with a 606 Not Acceptable response. Upon reception of the reject, the terminating gateway proposes to use plain G.711 to carry the fax stream.
4. The call is terminated.

6.2.1. Sequence Diagram

The messages F1 to F12 are identical to section 5.2.

```
<table>
<thead>
<tr>
<th>IFT UA</th>
<th>Proxy</th>
<th>IFTGW UA</th>
</tr>
</thead>
<tbody>
<tr>
<td>F1 INVITE</td>
<td></td>
<td>F2 INVITE</td>
</tr>
<tr>
<td>----------------</td>
<td>---------------------</td>
<td>----------------</td>
</tr>
</tbody>
</table>
```

Li/Mule
F3 100 Trying
<-------------------
F4 100 Trying

F6 180 Ringing
<-------------------
F5 180 Ringing

F8 200 OK
<-------------------
F7 200 OK

F9 ACK
------------------->
F10 ACK

Both Way RTP Media Established
<=======================================>
Fax emitted

F12 INVITE
<-------------------
F11 INVITE detected

F13 606
<-------------------
F14 606

F15 INVITE
<-------------------
F16 INVITE

F17 200 OK
<-------------------
F18 200 OK

F19 ACK
<-------------------
F20 ACK

G.711 RTP Flow Established
<=======================================>
for fax pass-through

F21 BYE
<-------------------
F22 BYE

F23 200 OK
<-------------------
6.2.2. Message Details

Messages F1 to F12 are identical to section 5.2; refer to 5.2.

F13 606 Not Acceptable IFT UA -> PROXY

SIP/2.0 606 Not Acceptable
Via: SIP/2.0/UDP ift.here.com:5060

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Warning: 304 Media type not available: One or more media types contained in the session description are not available.
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Content-Length: 0

F14 606 Not Acceptable PROXY -> IFTGW UA

SIP/2.0 606 Not Acceptable
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d010.1
Via: SIP/2.0/UDP ift.here.com:5060
Warning: 304 Media type not available: One or more media types contained in the session description are not available.
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 18 INVITE
Content-Length: 0

F15 INVITE IFTGW UA -> PROXY

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 181

v=0
o=faxgw1 2890844527 171091 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000
da=ecan:fb on -
da=silenceSupp:off -- -

F16 INVITE PROXY -> IFT UA

INVITE sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com>
Content-Type: application/sdp
Content-Length: 181

v=0
o=faxgw1 2890844527 171091 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12322 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ecan:fb on -
a=silenceSupp:off -- -

F17 200 OK IFT UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 150

v=0
o=faxgw1 2890844527 2890844528 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ecan:fb on -
a=silenceSupp:off -- -

F18 200 OK PROXY -> IFTGW UA
SIP/2.0 200 OK
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; 
                   maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com

CSeq: 19 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com>
Content-Type: application/sdp
Content-Length: 150

v=0
o=faxgw1 2890844527 2890844528 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ecan:fb on -
a=silenceSupp:off - - -

F19 ACK IFTGW UA -> PROXY

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com; 
                   maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 ACK
Content-Length: 0

F20 ACK PROXY -> IFT UA

ACK sip:+1-650-555-2222@obelix.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com; 
                   maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab11
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde17
Call-ID: 1717@ift.here.com
CSeq: 19 ACK
Content-Length: 0

/* G.711 audio streams established both ways for fax pass through */

F21 BYE IFT UA -> PROXY
BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22

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To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde34
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Length: 0

F22 BYE PROXY -> IFTGW UA

BYE sip:+1-650-555-2222@obelix.wcom.com SIP/2.0
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde34
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Length: 0

F23 200 OK IFTGW UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP obelix.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde34
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Type: application/sdp
Content-Length: 0

F24 200 OK PROXY -> IFT UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@obelix.wcom.com;
maddr=obelix.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone;tag=ab22
To: sip:+1-650-555-2222@obelix.wcom.com;user=phone;tag=cde34
Call-ID: 1717@ift.here.com
CSeq: 20 BYE
Content-Type: application/sdp
Content-Length: 0
7. SDP Attribute Table for T.38 sessions
For a detailed description of these attributes, refer to IANA. The tables below are replicated here for reference only.

<table>
<thead>
<tr>
<th>SDP Attribute Name</th>
<th>Appropriate values</th>
<th>Example</th>
</tr>
</thead>
<tbody>
<tr>
<td>T38FaxVersion</td>
<td>1*DIGIT</td>
<td>0</td>
</tr>
<tr>
<td>T38MaxBitRate</td>
<td>1*(DIGIT)</td>
<td>14400</td>
</tr>
<tr>
<td>T38FaxFillBitRemoval</td>
<td>boolean</td>
<td>0</td>
</tr>
<tr>
<td>T38FaxTranscodingMMR</td>
<td>boolean</td>
<td>0</td>
</tr>
<tr>
<td>T38FaxTranscodingJBIG</td>
<td>boolean</td>
<td>0</td>
</tr>
<tr>
<td>T38FaxRateManagement</td>
<td>localTCF; transferredTCF</td>
<td>transferredTCF</td>
</tr>
<tr>
<td>T38FaxMaxBuffer</td>
<td>1*(DIGIT); optional</td>
<td>72 (bytes)</td>
</tr>
<tr>
<td>T38FaxMaxDatagram</td>
<td>1*(DIGIT); optional</td>
<td>Depends on redundancy; 316 (bytes)</td>
</tr>
<tr>
<td>T38FaxUdpEc</td>
<td>t38UDP FEC; t38UDPRedundancy</td>
<td>T38UDPRedundancy</td>
</tr>
</tbody>
</table>

Registered SDP Protocol `proto` for T.38:

```
+-----------------+
| Name            |
| UDPTL           |
| TCP             |
```

Registered SDP Protocol `fmt`, MIME media type image/t38:
MIME media type name: image
MIME subtype name: t38
8. Considerations

In this temporary section, we log the open items for discussion as well as issues that have been solved on the mailing list:

8.1. Level of requirements for Internet gateways
--- Status: closed, solved by draft01 text in section 4.2
Based on the comments received on draft-00, the authors recommend that: T.38 Internet fax gateways MUST trigger a SIP re-INVITE upon detection of the V.21 Preamble flag sequence on the receiving side. Emitting gateways MAY trigger session modification upon CNG.

8.2. Negotiation of UDP ports for T.38 transmission
--- Status: not an issue, just logged here
When switching to T.38 mode, Internet fax gateways specify a new set of parameters for the media connection in SDP. Some gateways may choose the UDP/RTP port used in audio mode for UDP/UDPTL/T.38 traffic. Some gateways may choose to open a new UDP port for UDPTL/T.38 fax (and keep the previous UDP/RTP port open for the switch back to audio).
SDP provides the mechanism to precisely define in SIP all cases:
- if an Internet Fax gateway desires to re-use the UDP/RTP port for UDP/UDPTL/T.38, put a line in the re-INVITE.
  m=image <original_RTP_port> udptl t38
- if an Internet Fax gateway desires to release the RTP port and open a new UDP port for T.38:
  m=image <new_UDP_port> udptl t38
- if an Internet Fax gateway desires to keep the RTP port for future use in the session and open a new UDP port for T.38, we need to repeat the RTP/AVP port:
  m=audio <original_RTP_port> RTP/AVP 0
  m=image <new_UDP_port> udptl t38

8.3. Adding a fax stream to SIP session vs. Replacing audio stream
--- Status: closed, see flows 5.2 and 5.3
Based on the mailing list discussion, it is recommended in section 5.2 to replace an existing audio media connection with a fax one. We however kept section 5.3 to illustrate what adding a fax stream would entail.

8.4. Adding text to make use of SIP OPTIONS: in progress in draft-02
--- Status: Open for discussion
As discussed with Bill Sulzen, the SIP OPTIONS response could be used to assert that real-time fax communications using either T.38 or fax-passthrough are supported. This could certainly improve the negotiation phases and reduce the number of SIP messages. It is also key to get the capabilities of the Internet-aware fax device or gateway other than on a per call basis.
What needs to be added:
- add text in sub-section in section 4.5
- specify how the Accept header line should be populated (do we want to specify t.38 and/or fax pass-through (meaning ability to...
support ecann and silencesuppression), and within t.38, do we want to specify whether it is t.38 over tcp, udptl, both, etc.

8.5. Use of IP addresses in SDP connection information
--- Status: Open for discussion
Comment from Christer Holmberg: SDP connection information should use IP addresses instead of domain names (as per draft-ietf-mmusic-sdp-new-03)

8.6. Unsuccessful scenario: 606 or 488
--- Status: Open for discussion
Comment from Bill Sulzen: it is acceptable or even preferable to reject with a 488 Not Acceptable Media?

9. Security Considerations

The security mechanisms provided in RFC2543 apply: message authentication can be performed on SIP INVITEs and also BYE.

10. References

1  S. Bradner, "The Internet Standards Process -- Revision 3", BCP 9, RFC 2026, October 1996.

2  S. Bradner, "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.


11. Acknowledgments

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