IP Telephony/Voice over IP (VoIP)

Fax-SIP Troubleshoot Guide

Contents

TAC

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Introduction

This document describes one of the most effective approaches to troubleshoot fax, which includes these steps:

1. Split the call into two legs.
2. Identify the protocol (SIP/H.323/SCCP/MGCP) on each leg.
3. Choose a leg and then check if the call is incoming or outgoing on that leg and if the gateway/endpoint associated is a terminating gateway (TGW) or originating gateway (OGW) correspondingly.

You can split a fax call into four parts:

1. Set up the voice call
   - Off-hook, Dial, Ring, Answer
   - Calling (CNG) and Called Equipment Identification (CED) Tones
2. Switchover
   - Codec upspeed/correction
   - Voice Activation Detection (VAD) disabled on DSP
   - Jitter buffer transitions from adaptive to a fixed optimum value
3. Pre-message procedures
   - Fax Terminal Identification
   - Capabilities exchange and setting
   - Training
4. In-message and post message procedures
   - Transmission of pages
   - Error detection and correction (ECM)
   - End of message and page confirmation
   - Call Disconnect, On-hook

This call flow includes the messages to look for when Session Initiation Protocol (SIP) is the protocol identified. There are corresponding sections based on whether your endpoint is a TGW or OGW.

Note: In the table in the next section, both T.38 Relay and Passthrough were tested simultaneously and differences between G3 and SG3 have been pointed out.

TGW - Fax Call Incoming on SIP Leg

Note that:

- T.38 - Delay<1000ms, Jitter<300ms, Packet loss should be NONE unless T.38 with redundancy.
- Passthrough - Delay<1000ms, Jitter<30ms, Packet loss should be NONE.
- Protocol Based switchover - This is standard based.
- NSE Based switchover - This is proprietary and works only between Cisco voice gateways.

Passthrough	T.38 Relay
GW------------------------CUCM/GW **<-------------INVITE-------------------** **------------100TRYING-------------->** **------------180RINGING------------>**	**GW------------------------CUCM/GW** **<-------------INVITE-------------------** **------------100TRYING-------------->** **------------180RINGING------------>**

<table>
<tr>
<td colspan="2">Check for VTSP shows:

Fax Relay=DISABLED - 'fax rate disabled'
set (dial-peer)
Primary Fax Protocol=IGNORE_FAX_RELAY,
Fallback Fax Protocol=IGNORE_FAX_RELAY
Fax Relay CM Suppression :=ENABLED
, Fax Relay ANS Suppression :=DISABLED</td>
<td colspan="2">Check for VTSP shows:

Fax Relay=ENABLED
Primary Fax Protocol=T38_FAX_RELAY,
Fallback Fax Protocol=NONE_FAX_RELAY
Fax Relay CM Suppression :=ENABLED,
Fax Relay ANS Suppression :=DISABLED</td>
</tr>
<tr>
<th>Protocol Based</th>
<th>NSE Based</th>
<th>Protocol Based</th>
<th>NSE Based</th>
</tr>
<tr>
<td>GW------CUCM/GW

---200OK+SDP----->

v=0
o=CiscoSystemsSIP-GW-UserAgent
0 6060 IN IP4
209.165.201.2
s=SIP Call
c=IN IP4
209.165.201.2
t=0 0
m=audio 17924
RTP/AVP 0
c=IN IP4
209.165.201.2
a=rtpmap:0
PCMU/8000
a=ptime:20

<----ACK+SDP-----

v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4
209.165.201.3
s=SIP Call
c=IN IP4
209.165.201.1
t=0 0
m=audio 16724
RTP/AVP 0
a=rtpmap:0
PCMU/8000
a=ptime:20

Note: In case of EO, a similar SDP would have been received with INVITE.</td>
<td>GW---------CUCM/GW

------200OK+SDP------>

v=0
o=CiscoSystemsSIP-GW-UserAgent
5944 7031 IN IP4 209.165.201.2
s=SIP Call
c=IN IP4 209.165.201.2
t=0 0
m=audio 18806 RTP/AVP 0 100
c=IN IP4 209.165.201.2
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=ptime:20
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100 192-194,200-202
a=X-cap: 2 image udptl t38

<------ACK+SDP--------

v=0
o=CiscoSystemsCCM-SIP
2000 1 IN IP4 209.165.201.4
s=SIP Call
c=IN IP4 209.165.201.1
t=0 0
m=audio 16724 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=rtpmap:101
telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100 192-194,200-202
a=X-cap: 2 image udptl t38

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m=audio 17924 RTP/AVP
0
c=IN IP4 209.165.201.2
a=rtpmap:0 PCMU/8000
a=ptime:20

<-------ACK+SDP---------

v=0
o=CiscoSystemsCCM-SIP
2000
1 IN IP4 209.165.201.3
s=SIP Call
c=IN IP4 209.165.201.1
t=0 0
m=audio 16724 RTP/AVP
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a=rtpmap:0 PCMU/8000
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s=SIP Call
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t=0 0
m=audio 18806
RTP/AVP 0 100
c=IN IP4
209.165.201.2
a=rtpmap:0
PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,
200-202
a=ptime:20
a=X-sqn:0
a=X-cap: 1 audio
RTP/AVP 100
a=X-cpar:
a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100
192-194,200-202
a=X-cap: 2 image
udptl t38

<-----ACK+SDP------

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o=CiscoSystemsCCM-SIP 2000 1 IN IP4
209.165.201.3
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RTP/AVP 0
a=rtpmap:0
PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,
200-202
a=rtpmap:101
telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=X-sqn:0
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RTP/AVP 100
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<tr><td colspan="2">**GW------------------------CUCM/GW**

<========AUDIO==========>

Audio call established at this stage, but as the fax machines talk they start to exchange tones in the audio call.

Initial T.30 tones (Cannot be seen in debugs as these are always sent in RTP.)

G3 FAX:

<<<<<<<<<<CNG<<<<<<<<<<<

1100 Hz, every 3 seconds for .5 seconds. Indicates a calling nonspeech terminal.

>>>>>>>>>>CED>>>>>>>>>>>

2100 Hz tone that lasts between 2.6 - 4.0 seconds. Disables echo suppressors in the transmission path.

SG3 FAX:
<<<<<<<<<<CNG<<<<<<<<<<<

1100 Hz, every 3 seconds for .5 seconds. Indicates a calling nonspeech terminal.

>>>>>>>>>>ANSAM>>>>>>>>>

2100 Hz tone as CED, but amplitude modulated by a sine wave at 15 Hz with phase reversal every 450 ms.

<<<<<<<<<<CM<<<<<<<<<<<<

>>>>>>>>>>JM>>>>>>>>>>>>

<<<<<<<<<<CJ<<<<<<<<<<<<

V.34 Initialization (Phases 2-4)

The TGW waits to detect V.21 Preamble in the tones. It finds it in CED tone (G3) or ANSAM (SG3). Once it detects the V.21 Flag, it initiates switchover.

Check for VTSP shows:

Event=E_DSMP_DSP_MODEM_TONE

One of the tasks in switchover is to make the Jitter buffer transitions from adaptive to a fixed optimum value.

Fax passthrough uses the last voice mode setting before the switchover for jitter or playout buffers. Enter the **show voice port X/X/X** command in order to check the current values of playout delay.</td><td colspan="2">**GW------------------------CUCM/GW**

<========AUDIO==========>

Audio call established at this stage, but as the fax machines talk they start to exchange tones in the audio call.

Initial T.30 tones (Cannot be seen in debugs as these are always sent in RTP.)

G3 FAX:

<<<<<<<<<<CNG<<<<<<<<<<<

1100 Hz, every 3 seconds for .5 seconds. Indicates a calling nonspeech terminal.

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2100 Hz tone that lasts between 2.6 - 4.0 seconds. Disables echo suppressors in the transmission path.

SG3 FAX:
<<<<<<<<<<CNG<<<<<<<<<<<

1100 Hz, every 3 seconds for .5 seconds. Indicates a calling nonspeech terminal.

>>>>>>>>>>ANSAM>>>>>>>>>

2100 Hz tone as CED, but amplitude modulated by a sine wave at 15 Hz with phase reversal every 450 ms.

Cisco gateways only support G3 fax calls with T.38. In order to properly handle the higher speeds of SG3 calls, modem passthrough must be used.

No V.34 Initialization (Phases 2-4) exists, the initial V.8 Phase I also does not complete. OGW squelchs the CM tone and as SG3 is backward compatible with G3 fax standard, the FAX machines failover to G3.

>>>>>>>>>CED>>>>>>>>>>>

2100 Hz tone that lasts between 2.6 - 4.0 seconds. Disables echo suppressors in the transmission path.

The TGW waits to detect V.21 Preamble in the tones. It finds it in CED tone (G3) or ANSAM (SG3). Once it detects the V.21 Flag, it initiates switchover.

Check for VTSP shows:

VTSP: Event=E_DSMP_DSP_FAX_TONE

Check for DSMP shows:
E_DSM_CC_MC_START

Check for CCAPI shows:
CCAPI:Caps(Codec=T38Fax(0x10000), Fax Rate=FAX_RATE_14400(0x80),Fax Version:=0, Vad=OFF(0x1),

One of the tasks in switchover is to make the Jitter buffer transitions from adaptive to a fixed optimum value.</td></tr>
</table>

		T.38 uses 300 ms fixed jitter or playout buffers. Enter the **playout-delay fax 100** command under voice port in order to reduce the buffer time if the delay is high. Enter the **show voice port X/X/X** command in order to check the current values of playout delay.	
Protocol Based	**NSE Based**	**Protocol Based**	**NSE Based**
GW------CUCM/GW **---INVITE+SDP---->** *v=0* *o=CiscoSystemsSIP-GW-UserAgent 0 6060 IN IP4 209.165.201.2* *s=SIP Call* *c=IN IP4 209.165.201.2* *t=0 0* *m=audio 17924 RTP/AVP 0* *c=IN IP4 209.165.201.2* *a=rtpmap:0 PCMU/8000* a=silenceSupp:off - - - - **<----100TRYING----** **<--200OK+SDP-----** *v=0* *o=CiscoSystemsCCM-SIP 2000 1 IN IP4 209.165.201.3* *s=SIP Call* *c=IN IP4 209.165.201.1* *t=0 0* *m=audio 16724 RTP/AVP 0* *a=rtpmap:0 PCMU/8000* *a=silenceSupp:off - - - -* **--------ACK---------->** **show call active voice brief** will not show change	**G3 FAX:** **GW----------CUCM/GW** **====NSE192======>** Upspeed Codec and Switch to Passthrough Mode. Check for VTSP shows: *E_DSM_CC_MODIFY _MEDIA_IND* **debug voip rtp session named event**: *Pt:100 Evt:192 Pkt:00 00 00 <Snd>>>* **<====NSE192=======** Check for VTSP shows: *E_DSMP_DSP_REPORT _PEER_TO_PEER* *_MSG* **debug voip rtp session named event**: *<<<Rcv> Pt:100 Evt:192 Pkt:00 00 00* **SG3 FAX:** **GW----------CUCM/GW** **====NSE192=====>** Upspeed Codec and Switch to Passthrough Mode. Check for VTSP shows: *E_DSM_CC_MODIFY _MEDIA_IND* **debug voip rtp session named event**: *Pt:100 Evt:192 Pkt:00 00 00 <Snd>>>* **<====NSE192======** Check for VTSP shows: *E_DSMP_DSP_REPORT _PEER_TO_PEER* *_MSG* **debug voip rtp session named event**: *<<<Rcv> Pt:100 Evt:192 Pkt:00 00 00* **=====NSE193=====>** Detect phase reversal of ANSam Disable ECAN. Check for VTSP shows:	**GW-------------CUCM/GW** **-------INVITE+SDP------>** *v=0* *o=CiscoSystemsSIP-GW-UserAgent* *0 6061 IN IP4 209.165.201.2* *s=SIP Call* *c=IN IP4 209.165.201.2* *t=0 0* *m=image 17924 udptl t38* *c=IN IP4 209.165.201.2* *a=T38FaxVersion:0* *a=T38MaxBitRate:14400* *a=T38FaxFillBitRemoval:0* *a=T38FaxTranscoding MMR:0* *a=T38FaxTranscodingJ BIG:0* *a=T38FaxRate Management: transferredTCF* *a=T38FaxMaxBuffer:200* *a=T38FaxMax Datagram:320* *a=T38FaxUdpEC: t38UDPRedundancy* **<--------100TRYING------** **<-----200OK+SDP---------** *v=0* *o=CiscoSystemsCCM -SIP 2000 2 IN IP4 209.165.201.3* *s=SIP Call* *c=IN IP4 209.165.201.1* *t=0 0* *m=image 16384 udptl t38* **-----------ACK------------->** **show call active voice brief** shows: *t38*	**G3 FAX:** **GW------CUCM/GW** **====NSE200====>** Transition from voice mode to T.38 Check for VTSP shows *E_DSM_CC_ MODIFY _MEDIA_IND* **debug voip rtp session named event**: *Pt:100 Evt:200 Pkt:00 00 00 <Snd>>>* **<===NSE201=====** T.38 ACK received, instructs TGW to start T.38 session. Check for VTSP shows: *E_DSMP_DSP_ REPORT_PEER_ TO_PEER* *_MSG* **debug voip rtp session named event**: *<<<Rcv> Pt:100 Evt:201 Pkt:00 00 00* **SG3 FAX:** **As you spoof SG3 to G3 by squelching the CM tone, there is no SG3 FAX scenario in T38 relay.** **Note**: NSE-202 is a NACK to an NSE-200 message that signifies that the peer gateway cannot process T.38 packets for the call. The call remains in voice mode and does not switch over to T.38. **show call active voice brief** shows:

	*E_DSM_CC_MODIFY_* *MEDIA_IND* **debug voip rtp session named event**: *Pt:100 Evt:193 Pkt:00 00 00* *<Snd>>>* **<====NSE193======** Check for VTSP shows: *E_DSMP_DSP_REPORT* *_PEER_TO_PEER* *_MSG* **debug voip rtp session named event**: *<<<Rcv> Pt:100 Evt:193 Pkt:00 00* *00* **Note**: NSE-194 is triggered by a local detection of 4 seconds of silence or carrier loss detection. This message instructs the remote gateway to return to voice mode. Basically, all the changes made by NSE-192 and NSE-193 are undone. **show call active voice brief** shows: *MODEMPASS nse*		*t38*
In Passthrough you cannot see any T.30 messages from debugs as all tones go in the RTP-like audio with G711ulaw/alaw. However, the fax tone negotiation remains the same irrespective of relay or passthrough. **GW-------------------------CUCM/GW** **>>>>>>>>>CSI>>>>>>>>>>> (optional)**(called subscriber identification) **>>>>>>>>>NSF>>>>>>>>>>> (optional)**(nonstandard facilities) **>>>>>>>>>DIS>>>>>>>>>>>** (digital identification signal) **<<<<<<<<<TSI<<<<<<<<<<< (optional)**(transmitting subscriber identification) **<<<<<<<<<DCS<<<<<<<<<<** (digital command signal) **<++++++++++TCF++++++++++ (high speed)**(training check) **>>>>>>>>>>>CFR>>>>>>>>>>** (confirmation to receive) If you see FTT here that means TCF training failed. Check the clocking and slips on T1/E1. In packet captures, check TCF should be all 0. **<++++Partial Page RX+++++++ (high speed)** **<<<<<<<<PPS/EOM<<<<<<<<** (partial page sent)/(end of message) **>>>>>>>>>MCF>>>>>>>>>>>** (message confirmation) **<++++Partial Page RX+++++++ (high speed)** **<<<<<<<<PPS/EOP<<<<<<<<** (partial page sent)/(end of procedure) **>>>>>>>>>MCF>>>>>>>>>>>** (message confirmation) **<<<<<<<<<<DCN<<<<<<<<<<** (disconnect) **Note**: **ECM** is Optional for G3, but Mandatory for SG3. As you can achieve SG3 speeds with passthrough, make sure **ECM** is enabled on the fax machines for the fax to succeed.		If T38 switchover is successful, these messages are seen in the corresponding debugs: Check for VTSP shows: *event:E_CC_T38_START* Check for DSMP shows: *E_DSM_CC_MC_LOCAL_DNLD_DONE* Check for CCAPI shows: *Caps(Codec=T38Fax(0x10000),* *Fax Rate=FAX_RATE_14400(0x80),* *Fax Version:=0,* *Vad=OFF(0x1),* **debug fax relay t30 all-level-1**: *timestamp=1321430729 fr-msg-det NSF* *timestamp=1321431129 fr-msg-det CSI* *timestamp=1321431879 fr-msg-det DIS* *timestamp=1321435719 fr-msg-tx TSI* *timestamp=1321436329* *FR_GOOD_CRC_LS_DATA 0x0 bytes* *timestamp=1321436329 fr-msg-tx good crc, 0 bytes* *timestamp=1321436439 fr-msg-tx DCS* *timestamp=1321436619* *FR_GOOD_CRC_LS_DATA 0x0 bytes* *timestamp=1321441499 fr-msg-det CFR* *timestamp=1321461449 fr-msg-tx PPS* *timestamp=1321461639* *FR_GOOD_CRC_LS_DATA 0x0 bytes* *timestamp=1321463099 fr-msg-det MCF* *timestamp=1321466789 fr-msg-tx DCN* *timestamp=1321466869* *FR_GOOD_CRC_LS_DATA 0x0 bytes* *timestamp=1321466869 fr-msg-tx good crc, 0 bytes* **GW-------------------------CUCM/GW** **>>>>>>>>>CSI>>>>>>>>>>>** **(optional)**(called subscriber identification) **>>>>>>>>>NSF>>>>>>>>>>>**	

Also, **TCF** training signal is Required for G3, but is not applicable for SG3. **Note**: For Passthrough, a common channel of 64kbps (g711) is allocated. So, the higher and the lower speeds of the messages becomes irrelevant.		**(optional)**(nonstandard facilities) **>>>>>>>>>DIS>>>>>>>>>>>** (digital identification signal) **<<<<<<<<<TSI<<<<<<<<<<<** **(optional)**(transmitting subscriber identification) **<<<<<<<<<DCS<<<<<<<<<<** (digital command signal) **<++++++++++TCF++++++++++ (high speed)**(training check) **>>>>>>>>>>>CFR>>>>>>>>>>** (confirmation to receive) If you see FTT here that means TCF, training failed, check clocking and slips on T1/E1. In packet captures check TCF should be all 0. **<++++Partial Page RX+++++++ (high speed)** **<<<<<<<<PPS/EOM<<<<<<<<** (partial page sent)/(end of message) **>>>>>>>>>MCF>>>>>>>>>>>** (message confirmation) **<++++Partial Page RX+++++++ (high speed)** **<<<<<<<<PPS/EOP<<<<<<<<** (partial page sent)/(end of procedure) **>>>>>>>>>MCF>>>>>>>>>>>** (message confirmation) **<<<<<<<<<<DCN<<<<<<<<<<** (disconnect)	
Protocol Based	**NSE Based**	**Protocol Based**	**NSE Based**
DP level config: ## **fax protocol pass-through g711ulaw/g711alaw** ## **fax rate disable** ## **fax nsf 000000**	DP level config: ## **modem passthrough nse codec g711ulaw/g711alaw** ## **fax rate disable** ## **fax nsf 000000**	DP level config: ## **fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none** ## **fax nsf 000000** ## **fax-relay ecm disable** ## **fax-relay sg3-to-g3 system** ## **fax rate 14400**	DP level config: ## **fax protocol t38 nse force version 0 ls-redundancy 0 hs-redundancy 0 fallback none** ## **fax nsf 000000** ## **fax-relay ecm disable** ## **fax-relay sg3-to-g3 system** ## **fax rate 14400**

OGW - FAX Call Outgoing on SIP Leg

Note that:

- T.38 - Delay<1000ms, Jitter<300ms, Packet loss should be NONE unless T.38 with redundancy.
- Passthrough - Delay<1000ms, Jitter<30ms, Packet loss should be NONE.
- Protocol Based switchover - This is standard based.
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Passthrough	T.38 Relay
GW------------------------CUCM/GW **-------------INVITE-------------------->** **<------------100TRYING--------------** **<------------180RINGING-------------** Check for VTSP shows: *Fax Relay=DISABLED - 'fax rate disabled' set (dial-peer)* *Primary Fax Protocol=IGNORE_FAX_RELAY,* *Fallback Fax Protocol=IGNORE_FAX_RELAY*	**GW------------------------CUCM/GW** **-------------INVITE-------------------->** **<------------100TRYING--------------** **<------------180RINGING-------------** Check for VTSP shows: *Fax Relay=ENABLED* *Primary Fax Protocol=T38_FAX_RELAY,* *Fallback Fax Protocol=NONE_FAX_RELAY*

<table>
<tr><td colspan="2">Fax Relay CM Suppression :=ENABLED,
Fax Relay ANS Suppression :=DISABLED</td><td colspan="2">Fax Relay CM Suppression :=ENABLED,
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<tr><th>Protocol Based</th><th>NSE Based</th><th>Protocol Based</th><th>NSE Based</th></tr>
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<td>GW------CUCM/GW

<----200OK+SDP----

v=0
o=CiscoSystemsSIP-GW-UserAgent 0 6060 IN IP4 209.165.201.2
s=SIP Call
c=IN IP4 209.165.201.2
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m=audio 17924 RTP/AVP 0
c=IN IP4 209.165.201.2
a=rtpmap:0 PCMU/8000
a=ptime:20

-----ACK+SDP----->

v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 209.165.201.3
s=SIP Call
c=IN IP4 209.165.201.1
t=0 0
m=audio 16724 RTP/AVP 0
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Note: In case of EO, a similar SDP would have been sent in INVITE.</td>
<td>GW----------CUCM/GW

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v=0
o=CiscoSystemsSIP-GW-UserAgent 5944 7031 IN IP4 209.165.201.2
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a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap: 100 X-NSE/8000a=X-cpar: a=fmtp:100 192-194, 200-202a=X-cap: 2 image udptl t38

-------ACK+SDP------->

v=0
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s=SIP Call
c=IN IP4 209.165.201.1
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m=audio 16724 RTP/AVP 0
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a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194, 200-202
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
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--------ACK+SDP------>

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Note: In case of EO, a similar SDP would have been sent in INVITE.</td>
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a=rtpmap:100 X-NSE/8000
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a=ptime:20
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
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-------ACK+SDP------>

v=0
o=CiscoSystemsCCM-SIP 2000 1 IN IP4 209.165.201.3
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c=IN IP4 209.165.201.1
t=0 0
m=audio 16724 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194, 200-202
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap: 100 X-NSE/8000
a=X-cpar: a=fmtp: 100 192-194,200-202
a=X-cap: 2 image udptl t38

Note: In case of EO, a similar SDP would have been sent in INVITE.</td>
</tr>
<tr><td colspan="2">GW------------------------CUCM/GW

<========AUDIO=========></td><td colspan="2">GW------------------------CUCM/GW

<========AUDIO=========></td></tr>
</table>

<table>
<tr>
<td colspan="2">
Audio call established at this stage, but as FAX machines talk, they exchange tones in the audio call.
Initial T.30 tones (Cannot be seen in debugs as these are always sent in RTP.)
G3 FAX:
>>>>>>>>>>CNG>>>>>>>>>>>
1100 Hz, every 3 seconds for .5 seconds. Indicates a calling nonspeech terminal.
<<<<<<<<<<CED<<<<<<<<<<<
2100 Hz tone that lasts between 2.6 - 4.0 seconds. Disables echo suppressors in the transmission path.
SG3 FAX:
>>>>>>>>>>CNG>>>>>>>>>>>
1100 Hz, every 3 seconds for .5 seconds. Indicates a calling nonspeech terminal.
<<<<<<<<<<ANSAM<<<<<<<<
2100 Hz tone as CED, but amplitude modulated by a sine wave at 15Hz with phase reversal every 450 ms.
>>>>>>>>>>CM>>>>>>>>>>>>
<<<<<<<<<<JM<<<<<<<<<<<<
>>>>>>>>>>CJ>>>>>>>>>>>>
V.34 Initialization (Phases 2-4)
The OGW waits for the terminating gateway to detect V.21 Preamble in the tones. Once TGW detects the V.21 Flag, it initiates switchover.
One of the tasks in switchover is to make the Jitter buffer transitions from adaptive to a fixed optimum value.
Fax passthrough uses the last voice mode setting before the switchover for jitter or playout buffers. Enter the show voice port X/X/X command in order to check the current values of playout delay.
</td>
<td colspan="2">
Audio call established at this stage, but as FAX machines talk, they exchange tones in the audio call.
Initial T.30 tones (Cannot be seen in debugs as these are always sent in RTP.)
G3 FAX:
>>>>>>>>>>CNG>>>>>>>>>>>
1100 Hz, every 3 seconds for .5 seconds. Indicates a calling nonspeech terminal.
<<<<<<<<<<CED<<<<<<<<<<<
2100 Hz tone that lasts between 2.6 - 4.0 seconds. Disables echo suppressors in the transmission path.
SG3 FAX:
>>>>>>>>>>CNG>>>>>>>>>>>
1100 Hz, every 3 seconds for .5 seconds. Indicates a calling nonspeech terminal.
<<<<<<<<<<ANSAM<<<<<<<<
2100 Hz tone as CED, but amplitude modulated by a sine wave at 15Hz with phase reversal every 450 ms.
>>>CM>>X
Cisco gateways only support G3 fax calls with T.38. In order to properly handle the higher speeds of SG3 calls, modem passthrough must be used.
No V.34 Initialization (Phases 2-4) exist, the initial V.8 Phase I also does not complete. OGW squelchs the CM tone and as SG3 is backward compatible with G3 fax standard, the FAX machines failover to G3.
<<<<<<<<<<CED<<<<<<<<<<<
2100 Hz tone that lasts between 2.6 - 4.0 seconds. Disables echo suppressors in the transmission path.
The OGW waits for the TGW to detect V.21 Preamble in the tones. Once TGW detects the V.21 Flag, it initiates switchover.
One of the tasks in switchover is to make the Jitter buffer transitions from adaptive to a fixed optimum value.
T.38 uses 300 ms fixed jitter or playout buffers. Enter the playout-delay fax 100 command under voice port to reduce the buffer time if the delay is high. Enter the show voice port X/X/X command in order to check the current values of playout delay.
</td>
</tr>
<tr>
<th>Protocol Based</th>
<th>NSE Based</th>
<th>Protocol Based</th>
<th>NSE Based</th>
</tr>
<tr>
<td>
GW------CUCM/GW
<---INVITE+SDP----
v=0
o=CiscoSystemsSIP-GW-UserAgent 0 6060 IN IP4 209.165.201.2
s=SIP Call
c=IN IP4 209.165.201.2
t=0 0
</td>
<td>
G3 FAX:
GW---------CUCM/GW
<====NSE192====
Upspeed Codec and Switch to Passthrough Mode.
Check for VTSP shows:
</td>
<td>
GW---------CUCM/GW
<-----INVITE+SDP------
v=0
o=CiscoSystemsSIP-GW
-UserAgent 0 6061 IN IP4 209.165.201.2
s=SIP Call
c=IN IP4 209.165.201.2
t=0 0
</td>
<td>
G3 FAX:
GW---------CUCM/GW
<====NSE200=====
Transition from voice mode to T.38
Check for VTSP shows:
E_DSMP_DSP_
</td>
</tr>
</table>

m=audio 17924 RTP/AVP 0 *c=IN IP4 209.165.201.2* *a=rtpmap:0 PCMU/8000* a=silenceSupp:off - - - - **----100TRYING---->** **----200OK+SDP---->** *v=0* *o=CiscoSystemsCCM-SIP 2000 1 IN IP4 209.165.201.3* *s=SIP Call* *c=IN IP4 209.165.201.1* *t=0 0* *m=audio 16724 RTP/AVP 0* *a=rtpmap:0 PCMU/8000* *a=silenceSupp:off - - - -* **<------ACK---------** **show call active voice brief** will not show change	*E_DSMP_DSP_REPORT _PEER_TO_PEER* *_MSG* **debug voip rtp session named event**: *<<<Rcv> Pt:100 Evt:192 Pkt:00 00 00* **====NSE192=====>** Check for VTSP shows: *E_DSM_CC_MODIFY _MEDIA_IND* **debug voip rtp session named event**: *Pt:100 Evt:192 Pkt:00 00 00 <Snd>>>* **SG3 FAX:** **GW----------CUCM/GW** **<====NSE192======** Upspeed Codec and Switch to Passthrough Mode. Check for VTSP shows: *E_DSMP_DSP_ REPORT _PEER_TO_PEER* *_MSG* **debug voip rtp session named event**: *<<<Rcv> Pt:100 Evt:192 Pkt:00 00 00* **====NSE192======>** Check for VTSP shows: *E_DSM_CC_MODIFY _MEDIA_IND* **debug voip rtp session named event**: *Pt:100 Evt:192 Pkt:00 00 00 <Snd>>>* **<====NSE193=====** Disable ECAN. Check for VTSP shows: *E_DSMP_DSP_REPORT _PEER_TO_PEER* *_MSG* **debug voip rtp session named event**: *<<<Rcv> Pt:100 Evt:193 Pkt:00 00 00* **====NSE193=====>** Check for VTSP shows: *E_DSM_CC_MODIFY _MEDIA_IND* **debug voip rtp session named event**: *Pt:100 Evt:193 Pkt:00 00 00*	*m=image 17924 udptl t38* *c=IN IP4 209.165.201.2* *a=T38FaxVersion:0* *a=T38MaxBitRate:14400* *a=T38FaxFillBit Removal:0* *a=T38FaxTranscoding MMR:0* *a=T38FaxTranscoding JBIG:0* *a=T38FaxRate Management: transferredTCF* *a=T38FaxMaxBuffer:200* *a=T38FaxMax Datagram:320* *a=T38FaxUdpEC: t38UDPRedundancy* **--------100TRYING----->** **-------200OK+SDP------>** *v=0* *o=CiscoSystemsCCM-SIP 2000 2 IN IP4 209.165.201.3* *s=SIP Call* *c=IN IP4 209.165.201.1* *t=0 0* *m=image 16384 udptl t38* **<--------ACK------------** **show call active voice brief** will show: *t38* **Note**: Whenever CUCM is involved, for the RE_INVITE in these topologies: Fax--GW---(h323)--CUCM---(sip)---GW---FAX Fax--GW---(mgcp)--CUCM---(sip)---GW---FAX Fax--GW---(sccp)---CUCM---(sip)---GW---FAX The SDP in the RE-INVITE will have: ... *m=image 17218 udptl t38* *c=IN IP4 0.0.0.0* ... It will always first send 0.0.0.0/t38, and then later send another t38 invite with a real IP. Such behavior is not seen in this topology since CUCM handles media differently for this scenario: Fax--GW---(sip)---CUCM---(sip)---GW---FAX	*REPORT _PEER_TO_PEER* *_MSG* **debug voip rtp session named event**: *<<<Rcv> Pt:100 Evt:200 Pkt:00 00 00* **====NSE201=====>** T.38 ACK received, instructs TGW to start T.38 session Check for VTSP shows: *E_DSM_CC_MODIFY_ MEDIA_IND* **debug voip rtp session named event**: *Pt:100 Evt:201 Pkt:00 00 00 <Snd>>>* **SG3 FAX:** **As you spoof SG3 to G3 by squelching the CM tone, there is no SG3 FAX scenario in T38 relay.** **Note**: NSE-202 is a NACK to an NSE-200 message that signifes that the peer gateway cannot process T.38 packets for the call. The call will remain in voice mode and not switch over to T.38. **show call active voice brief** shows: *t38*

<Snd>>>

Note: NSE-194 is triggered by a local detection of 4 seconds of silence or carrier loss detection. This message instructs the remote gateway to return to voice mode. Basically, all the changes made by NSE-192 and NSE-193 are undone.

show call active voice brie" shows *MODEMPASS nse*

Specially when CUBE is involved, keep in mind this:
CSCtj50993,
CSCtx83833

In Passthrough you cannot see any T.30 messages from debugs as all tones go in the RTP-like audio with G711ulaw/alaw. However, the FAX tone negotiation remains the same irrespective of relay or passthrough.

GW------------------------CUCM/GW

<<<<<<<<<CSI<<<<<<<<<<< **(optional)**(called subscriber identification)
<<<<<<<<<NSF<<<<<<<<<<< **(optional)**(nonstandard facilities)
<<<<<<<<<DIS<<<<<<<<<<< (digital identification signal)

>>>>>>>>>TSI>>>>>>>>>>>> **(optional)**(transmitting subscriber identification)
>>>>>>>>>DCS>>>>>>>>>>> (digital command signal)

++++++++++TCF+++++++++> **(high speed)**(training check)

<<<<<<<<<<CFR<<<<<<<<<< (confirmation to receive)

If you see FTT here that means TCF training failed, check clocking and slips on T1/E1. In packet captures check TCF should be all 0.

++++Partial Page RX++++++> **(high speed)**
>>>>>>>>>PPS/EOM>>>>>>> (partial page sent)/(end of message)

<<<<<<<<<MCF<<<<<<<<<<< (message confirmation)

++++Partial Page RX++++++> **(high speed)**
>>>>>>>>>PPS/EOM>>>>>>> (partial page sent)/(end of message)

<<<<<<<<<MCF<<<<<<<<<<< (message confirmation)

>>>>>>>>>DCN>>>>>>>>>>> (disconnect)

Note: **ECM** is Optional for G3, but Mandatory for SG3. As you can achieve SG3 speeds with passthrough, make sure **ECM** is enabled on the fax machines for the fax to succeed.
Also, **TCF** training signal is Required for G3, but is Not applicable for SG3.

Note: For Passthrough a common channel of 64kbps (g711) is allocated. So, the higher and the lower speeds of the messages becomes irrelevant.

If T38 switchover is successful, these messages are seen in the corresponding debugs:

Check for VTSP shows:

event:E_CC_T38_START

Check for DSMP shows:
E_DSM_CC_MC_LOCAL_DNLD_DONE

Check for CCAPI shows:
Caps(Codec=T38Fax(0x10000), Fax Rate=FAX_RATE_14400(0x80),Fax Version:=0, Vad=OFF(0x1),

debug fax relay t30 all-level-1:
timestamp=352583286 fr-msg-tx NSF
timestamp=352583686 fr-msg-tx CSI
timestamp=352583736
FR_GOOD_CRC_LS_DATA 0x0 bytes
timestamp=352583736 fr-msg-tx good crc, 0 bytes
timestamp=352584426 fr-msg-tx DIS
timestamp=352584456
FR_GOOD_CRC_LS_DATA 0x0 bytes
timestamp=352584456 fr-msg-tx good crc, 0 bytes
timestamp=352584906
FR_GOOD_CRC_LS_DATA 0x0 bytes
timestamp=352587656 fr-msg-det TSI
timestamp=352588376 fr-msg-det DCS
timestamp=352594056 fr-msg-tx CFR
timestamp=352594156
FR_GOOD_CRC_LS_DATA 0x0 bytes

timestamp=352613376 fr-msg-det PPS
timestamp=352615656 fr-msg-tx MCF
timestamp=352615776
FR_GOOD_CRC_LS_DATA 0x0 bytes
timestamp=352618716 fr-msg-det DCN

GW------------------------CUCM/GW

<<<<<<<<<CSI<<<<<<<<<<<
(optional)(called subscriber identification)
<<<<<<<<<NSF<<<<<<<<<<<
(optional)(nonstandard facilities)
<<<<<<<<<DIS<<<<<<<<<<< (digital identification signal)

>>>>>>>>>TSI>>>>>>>>>>>>
(optional)(transmitting subscriber identification)
>>>>>>>>>DCS>>>>>>>>>>> (digital command signal)

++++++++++TCF+++++++++> **(high speed)**(training check)

<<<<<<<<<<CFR<<<<<<<<<< (confirmation to receive)

If you see FTT here that means TCF training failed, check clocking and slips on T1/E1. In packet captures check TCF should be all 0.

		++++Partial Page RX++++++> (high speed) **>>>>>>>>>PPS/EOM>>>>>>>** (partial page sent)/(end of message) **<<<<<<<<<MCF<<<<<<<<<<<** (message confirmation) **++++Partial Page RX++++++> (high speed)** **>>>>>>>>>PPS/EOM>>>>>>>** (partial page sent)/(end of message) **<<<<<<<<<MCF<<<<<<<<<<<** (message confirmation) **>>>>>>>>>DCN>>>>>>>>>>>** (disconnect)	
Protocol Based	**NSE Based**	**Protocol Based**	**NSE Based**
DP level config: ## **fax protocol pass-through g711ulaw/g711alaw** ## **fax rate disable** ## **fax nsf 000000**	DP level config: ## **modem passthrough nse codec g711ulaw/g711alaw** ## **fax rate disable** ## **fax nsf 000000**	DP level config: ## **fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none** ## **fax nsf 000000** ## **fax-relay ecm disable** ## **fax-relay sg3-to-g3 system** ## **fax rate 14400**	DP level config: ## **fax protocol t38 nse force version 0 ls-redundancy 0 hs-redundancy 0 fallback none** ## **fax nsf 000000** ## **fax-relay ecm disable** ## **fax-relay sg3-to-g3 system** ## **fax rate 14400**

Debugs to Collect

- debug vpm all (in case of FXS)
- debug isdn q931 (in case of PRI)
- debug voice ccapi inout
- debug ccsip all/messages/verbos
- debug voip vtsp all
- debug voip dsmp all
- debug voip hpi all
- debug dsp-resource flex all
- debug voip dspapi
- debug fax relay t30 all-level-1
- debug voip rtp session named-event (in case of NSE based switchover)

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The Cisco Support Community is a forum for you to ask and answer questions, share suggestions, and collaborate with your peers.

Refer to Cisco Technical Tips Conventions for information on conventions used in this document.

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