Understanding, Integrating, and Troubleshooting Fax Communications in Today's Enterprise Networks

BRKUCC-2021
Agenda

- Fundamental Concepts
- Fax Transport Methods
- QoS Considerations
- Design Best Practices
- Troubleshooting
- Fax Servers
Fundamental Concepts
Fax Communications

- Fax invented by Alexander Bain in 1843 (patented 30 years before the telephone)
- Fax is a ubiquitous form of communication today and Fax over IP (FoIP) is often overlooked in IP Telephony
- Group 3 (G3) is today’s fax standard (speeds up to 14.4 kbps)
- Super G3 is an optional extension of G3 that allows for speeds up to 33.6 kbps
G3 Fax Protocols

- Modern fax communication is based on the Group 3 (G3) fax standard, which is composed of the ITU-T specifications of T.30, T.4, and T.6
- T.30 details the signaling, T.4 specifies the Modified Huffman (MH) and Modified Read (MR) encoding methods, and T.6 covers Modified Modified Read (MMR)
Basic G3 Fax Transaction

- The optional Called Subscriber Identification (CSI) and Non-Standard Facilities (NSF) messages may be transmitted along with the DIS message.
- The optional Transmitting Subscriber Identification (TSI) message may be transmitted along with the DCS message.
- Blue messages indicate low speed T.30 signaling and gold messages represent higher speed modulations used in training and page transmission.
Super G3 Fax

- Super G3 uses the V.34 modulation to achieve page transmission speeds up to 33.6 Kbps compared to the maximum G3 speed of 14.4 Kbps.

- The signals used by SG3 are different than G3 and are currently incompatible with T.38 and Cisco fax relay.
Transporting Fax Over IP

- The default behavior for Cisco voice gateways is to initially handle all calls as voice calls so all fax calls start as voice calls
- Compressed voice codecs are not designed to handle modulated data so alternative, real-time transport methods must be utilized
- There are two principal methods of transporting fax communications
  1. Passthrough
  2. Relay
Transporting G3 Fax Protocols Over IP

T.4/T.6—Specifies the Fax Page Encoding Algorithm

PSTN

T.30—Signaling and Negotiation Between Fax Machines

How Do Standard Fax Communications Integrate With IP Networks?

T.4/T.6—Specifies the Fax Page Encoding Algorithm

Passthrough or Relay Transport

T.30—Signaling and Negotiation Between Fax Machines
Passthrough

- Modulated fax data information is sampled and encoded as standard Pulse Code Modulation (PCM) (i.e., G.711) and encapsulated in Real Time Protocol (RTP) for transport over IP just like a voice codec does for human speech.
- From the gateway perspective, this is more or less a G.711 voice call.
- Commonly referred to as Voice Band Data (VBD).
Relay

- The analog modulated fax data is demodulated by a Digital Signal Processor (DSP) on the gateway and the binary information is extracted.
- Binary information is passed over IP using one of several available relay protocols.
- A DSP on the destination gateway takes the binary information from the relay packets and re-modulates it into a fax data signal on the telephony side.
The Switchover

- All modulated calls begin as voice calls
- The process that transitions a gateway from a VoIP call to the final fax media stream is known as the switchover
- There are a variety of different switchover mechanisms that we will discuss throughout this presentation
Fax Transport Methods
Fax Transport Methods

Fax over IP

Passthrough

Fax Passthrough (NSE)

Fax Pass-through (Protocol)

T.38 Fax Relay (Protocol)

T.38 Fax Relay (NSE)

Cisco Fax Relay
Fax Passthrough Switchover Methods

Voice Mode

NSE-Based Switchover

Passthrough Switchover:
1. Codec ups speed
2. VAD disabled on DSP
3. Jitter buffer transitions from adaptive to a fixed optimum value

Protocol-Based Switchover

Fax Passthrough

Fax Passthrough
Passthrough Terminology

- “Fax Passthrough” generically refers to passing fax calls over the G.711 codec.
- “Fax Passthrough” can also refer to the more specific passthrough transport method using an NSE switchover to pass faxes over G.711.
- “Modem Passthrough” is a general term used to refer to an NSE-based switchover to passthrough for both fax and modem calls because it is the syntax used by the configuration command for this feature.
- The underlying operation of “modem passthrough” is different depending on whether it is a fax or a high speed modem call, so to avoid confusion we will refer to “modem passthrough” for a fax call as “fax passthrough” and “modem passthrough” for a modem call as “modem passthrough”.
- “Fax pass-through” details a switchover to passthrough using the call control protocol.
Passthrough Switchover Mechanisms

1. Terminating Gateway (TGW) initiates the switchover to passthrough upon detection of fax signals played by the answering fax device.

2. If NSE-based passthrough is configured, the TGW sends NSE switchover packets in the RTP media stream upon detection of **CED (G3 fax)** or **ANSam (SG3 fax)** tones.

3. If fax pass-through is configured, the TGW initiates the switchover in the H.323 or SIP call control protocols upon detection of **fax message flags**.
### Payload Types Used for Modulated Data

- The Payload Type field identifies the type of data being carried in the RTP packet.

<table>
<thead>
<tr>
<th>Payload Type</th>
<th>Payload Encoding</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>G.711 µ-law</td>
</tr>
<tr>
<td>8</td>
<td>G.711 a-law</td>
</tr>
<tr>
<td>18</td>
<td>G.729</td>
</tr>
<tr>
<td>90</td>
<td>RFC 2198 Passthrough Redundancy</td>
</tr>
<tr>
<td>96</td>
<td>Cisco Fax Relay Switchover</td>
</tr>
<tr>
<td>97</td>
<td>Cisco Fax Relay Switchover ACK</td>
</tr>
<tr>
<td>100</td>
<td>Named Signaling Event (NSE)</td>
</tr>
<tr>
<td>119</td>
<td>Cisco Text Relay</td>
</tr>
<tr>
<td>122</td>
<td>Cisco Fax Relay</td>
</tr>
</tbody>
</table>

**Static**

**Dynamic And Unassigned**
Named Signaling Events (NSE)

- Cisco proprietary message sent as part of the RTP stream to signal an event, such as a switchover
- This form of signaling is call control independent
- The Event ID field uses Cisco-defined event numbers to signal a specific task in-band

![Diagram of Named Signaling Events (NSE)]
**NSE Events Used for Modulated Data**

- The NSE Event ID field uses Cisco-defined values to signal in-band a variety of tasks

<table>
<thead>
<tr>
<th>NSE</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>192</td>
<td>Triggered by 2100Hz tone. Signals switchover to passthrough</td>
</tr>
<tr>
<td>193</td>
<td>Triggered by ANSam tone detection. Signal to disable ECANs</td>
</tr>
<tr>
<td>194</td>
<td>Triggered by 4s silence or carrier loss. Signals return to voice mode</td>
</tr>
<tr>
<td>199</td>
<td>Triggered by ANSam tone. Signals support of Cisco Modem Relay</td>
</tr>
<tr>
<td>200</td>
<td>Triggered by fax V.21 Preamble. Signals switchover to T.38</td>
</tr>
<tr>
<td>201</td>
<td>ACK to NSE-200 confirming peer gateway switchover to T.38</td>
</tr>
<tr>
<td>202</td>
<td>NACK to NSE-200 implying peer gateway cannot switchover to T.38</td>
</tr>
<tr>
<td>203</td>
<td>Triggered by V.8 CM detection. Signals transition to Modem Relay</td>
</tr>
</tbody>
</table>
Fax Passthrough Switchover (G3)

**OGW:** Upspeed codec and switch to Passthrough mode

**TGW:** Upspeed codec and switch to Passthrough mode

The initial VoIP call can be setup by any of the common call control protocols (H.323, SIP, MGCP, or SCCP)

The underlying call control protocol is unaware of the transition to fax passthrough due to the protocol independent nature of the NSE switchover.
Fax Passthrough Switchover (SG3)

The initial VoIP call can be setup by any of the common call control protocols (H.323, SIP, MGCP, or SCCP).

OGW: Upspeed codec and switch to Passthrough mode

OGW: Disable echo cancellers

TGW: Detect 2100 Hz tone. Upspeed codec and switch to Passthrough mode

TGW: Detect phase reversal of ANSam. Disable ECAN
Confirming Fax Passthrough Switchover

**Before Switchover:** show call active voice brief
11F1 : 10 2924510ms.1 +10530 pid:1 Originate 200 active
dur 00:00:07 tx:99/1903 rx:253/4993
IP 1.1.1.2:17932 SRTP: off rtt:0ms pl:4810/0ms lost:0/0/0 delay:60/60/70ms \textcolor{red}{g729r8}

**VoIP Call**

**After Switchover:** show call active voice brief
11F1 : 10 2924510ms.1 +10530 pid:1 Originate 200 active
dur 00:00:22 tx:877/125447 rx:1040/128809
IP 1.1.1.2:17932 SRTP: off rtt:1ms pl:40/0ms lost:0/0/0 delay:60/60/60ms \textcolor{red}{g711ulaw}
TextRelay: off
media inactive detected:n media contrl rcvd:n/a timestamp:n/a
long duration call detected:n long duration call duration:n/a timestamp:n/a \textcolor{red}{MODEMPASS}
nse buf:0/0 loss 0% 0/0 last 1359s dur:0/0s

Fax Passthrough Call
Fax Passthrough Configuration

SIP/H.323:

dial-peer voice 1 voip
  modem passthrough nse codec g711ulaw

(Note: Can be configured globally under voice service voip or the dialpeer for SIP and H.323. The configuration under the dialpeer takes precedence.)

MGCP:

mgcp modem passthrough voip mode nse
mgcp modem passthrough voip codec g711ulaw

SCCP:

voice service voip
  modem passthrough nse codec g711ulaw

(Note: Can only be configured globally under voice service voip for SCCP.)
Fax Pass-through Switchover (G3)

Pass-through parameters for the upcoming session are established in the SIP INVITE and 200 OK messages.

An H.245 Request Mode for G.711 is used to signal the switchover in H.323 to fax pass-through mode.

The initial VoIP call is setup using the SIP call control protocol.

The V.21 Preamble of the first T.30 message following the CED Tone (i.e. DIS) is the trigger for the transition to pass-through.
Confirming Fax Pass-through Switchover

**Before Switchover:** show call active voice brief

126C : 58 11411290ms.1 +10520 pid:1 Originate 200 active
dur 00:00:08 tx:112/2154 rx:307/6092
IP 1.1.1.2:17620 SRTP: off rtt:1ms pl:5930/0ms lost:0/0/0 delay:60/60/70ms

**VoIP Call**

**After Switchover:** show call active voice brief

126C : 58 11411290ms.1 +10520 pid:1 Originate 200 active
dur 00:00:41 tx:1677/252554 rx:1953/258112
IP 1.1.1.2:17620 SRTP: off rtt:4ms pl:40/0ms lost:1/0/0 delay:60/60/60ms
TextRelay: off
media inactive detected:n media contrl rcvd:n/a timestamp:n/a
long duration call detected:n long duration call duration:n/a timestamp:n/a

**Fax Pass-through Call**
Fax Pass-through Configuration

SIP/H.323:
dial-peer voice 1 voip
  fax protocol pass-through g711ulaw

Note: Can be configured globally under voice service voip or the dialpeer for SIP and H.323. The configuration under the dialpeer takes precedence.

MGCP:
Fax pass-through is NOT supported for MGCP

SCCP:
Fax pass-through is NOT supported for SCCP
Fax Transport Methods

Fax over IP

Passthrough

Relay

Fax Passthrough (NSE)
Fax Pass-through (Protocol)
T.38 Fax Relay (Protocol)
T.38 Fax Relay (NSE)
Cisco Fax Relay
Fax Relay Switchover Methods

- Protocol-Based Switchover (Standards-Based)
- NSE-Based Switchover
- RTP PT Switchover

Fax Relay

- T.38 Fax Relay
- T.38 Fax Relay
- Cisco Fax Relay
Cisco Fax Relay (CFR)

- Pre-standard and proprietary fax relay implementation for G3 fax transmissions (no SG3 support)
- Default fax transport method on most Cisco voice gateways
- Uses a unique Payload Type switchover that is not used by any other modulated communication transport method

Cisco Fax Relay Packet

<table>
<thead>
<tr>
<th>IP Header</th>
<th>UDP Header</th>
<th>RTP Header</th>
</tr>
</thead>
</table>

Demodulated fax HDLC data carried in the Cisco Fax Relay packet

RTP Payload Type is set to 122 for Cisco Fax Relay data

© 2010 Cisco and/or its affiliates. All rights reserved. Cisco Confidential
The TGW notifies the OGW of the impending switchover with a special RTP message using a PT value of 96.

The CFR switchover packet is ACK’d with PT-97 message.

The detection of fax flags (V.21 Preamble) at the TGW triggers the transition to CFR.
Confirming Cisco Fax Relay Switchover

Before Switchover: `show call active voice brief`

121C : 26 1027355900ms.1 +10510 pid:1 Originate 200 active
dur 00:00:11 tx:117/2244 rx:424/8432
IP 1.1.1.2:18274 SRTP: off rtt:2ms pl:7810/0ms lost:0/0/0 delay:60/60/70ms g729r8

VoIP Call

After Switchover: `show call active voice brief`

121C : 26 1027355900ms.1 +10510 pid:1 Originate 200 active
dur 00:00:45 tx:1113/22164 rx:1001/19972
IP 1.1.1.2:18274 SRTP: off rtt:2ms pl:0/0ms lost:0/0/0 delay:0/0/0ms cisco TextRelay: off
media inactive detected:n media contrl rcvd:n/a timestamp:n/a
long duration call detected:n long duration call duration:n/a timestamp:n/a

Cisco Fax Relay Call
Cisco Fax Relay Configuration

**SIP/H.323:**
```
dial-peer voice 1 voip
  fax protocol cisco
```
Note: CFR is the default fax configuration on most IOS gateways.

**MGCP:**
```
ccm-manager fax protocol cisco
mgcp fax t38 inhibit
```
Note: Like CFR, T.38 is also enabled by default for MGCP. However T.38 takes precedence over CFR and must be explicitly disabled with `mgcp fax t38 inhibit`.

**SCCP:**
```
voice service voip
  fax protocol cisco
```
Note: CFR can only be configured under `voice service voip` for SCCP starting in IOS Release 12.4(11)T.
T.38 Fax Relay

- T.38 is the de facto standard for handling fax transmissions today
- Cisco has always supported the 1998 version (known as version 0) of the ITU-T T.38 specification with UDP/UDPTL encapsulation
- In IOS version 15.1(1)T Cisco started supporting SG3 fax over T.38 (version 3)
- Cisco products can use either NSEs (proprietary) or the call control protocol (standards-based) to switch a call to T.38 mode

UDP Transport Layer (UDPTL) header is simply a 2 byte sequence number

T.38 Internet Fax Packets (IFP) and optional redundancy packets are carried in the UDPTL payload

<table>
<thead>
<tr>
<th>IP Header</th>
<th>UDP Header</th>
<th>UDPTL Header</th>
<th>UDPTL Payload (T.38 IFPs)</th>
</tr>
</thead>
</table>

UDP Encapsulated T.38
The initial VoIP call can be setup by any of the common call control protocols (H.323, SIP, MGCP, or SCCP).

The detection of fax flags (V.21 Preamble) at the TGW triggers the transition to T.38.

OGW: Transition from voice mode to T.38

TGW: T.38 ACK received, start T.38 session
T.38 Switchover—H.323 (G3)

T.38 parameters for the upcoming fax relay session are negotiated in the H.245 Request Mode messages.

Each gateway closes its voice media channel and opens a T.38 fax relay media channel while acknowledging the same events by the other side.

The initial VoIP call is setup using the H.323 call control protocol.

The detection of fax flags (V.21 Preamble) at the TGW triggers the transition to T.38.
T.38 Switchover—SIP (G3)

The initial VoIP call is setup using the SIP call control protocol.

The detection of fax flags (V.21 Preamble) at the TGW triggers the transition to T.38.

T.38 fax relay switchover process using the SIP protocol stack is complete.

T.38 parameters for the upcoming fax relay session are negotiated in the SIP INVITE and 200 OK messages.

The detection of fax flags (V.21 Preamble) at the TGW triggers the transition to T.38.

The initial VoIP call is setup using the SIP call control protocol.

T.38 fax relay switchover process using the SIP protocol stack is complete.
T.38 Switchover—MGCP (G3)

The initial VoIP call is setup using the MGCP call control protocol.

CA instructs OGW to switchover to T.38 with an MDCX message.

TGW notifies the CA that fax V.21 flags are detected.

CA instructs TGW to switchover to T.38 with an MDCX message.
The initial VoIP call is setup using the SIP call control protocol.

The initial VoIP call is setup using the SIP call control protocol.

The detection of Calling Menu (CM) at the OGW triggers the transition to T.38.

Note: CM is squelched by the OGW DSP and sent over T.38. The CM/JM exchange is done entirely over T.38 prior to T.30 negotiation.
Confirming T.38 Fax Relay Switchover

**Before Switchover:** show call active voice brief

11E2 : 4 2956390ms.1 +10500 pid:1 Originate 200 active
dur 00:00:11 tx:117/2244 rx:436/8643
IP 1.1.1.2:18496 SRTP: off rtt:3ms pl:4800/0ms lost:0/0/0 delay:60/60/70ms g729r8

**T.38 Fax Relay Call**

**After Switchover:** show call active voice brief

11E2 : 4 2956390ms.1 +10500 pid:1 Originate 200 active
dur 00:00:46 tx:707/13770 rx:914/12339
IP 1.1.1.2:18496 SRTP: off rtt:3ms pl:4800/0ms lost:0/0/0 delay:60/60/70ms t38
TextRelay: off
media inactive detected:n media contrl rcvd:n/a timestamp:n/a
long duration call detected:n long duration call duration:n/a timestamp:n/a

© 2010 Cisco and/or its affiliates. All rights reserved. Cisco Confidential
NSE-Based T.38 Configuration

SIP/H.323:

dial-peer voice 1 voip
  fax protocol t38 nse ls-redundancy 0 hs-redundancy 0 fallback cisco

Note: Configured globally under voice service voip or the dialpeer for SIP and H.323. The dialpeer configuration takes precedence.

MGCP:
  Enabled by default

Note: no mgcp fax t38 inhibit is required but is default.

SCCP:

voice service voip
  fax protocol t38 nse ls-redundancy 0 hs-redundancy 0 fallback cisco

Note: Can only be configured under voice service voip for SCCP. Cisco IOS gateway support for NSE-based T.38 fax relay with the SCCP voice signaling protocol is introduced in Cisco IOS Release 12.4(11)T.
Protocol-Based T.38 Configuration

**SIP/H.323:**

dial-peer voice 1 voip

fax protocol t38 version [0|3] ls-redundancy 0 hs-redundancy 0 fallback cisco

Note: version 0 configures legacy T.38 G3 fax support, whereas version 3 enables the newly supported SG3 over T.38 feature [as of 15.1(1)T]

**MGCP:**

mgcp package-capability fxr-package

mgcp default-package fxr-package

Note: These two commands are needed to go from the default configuration of NSE-based T.38 to protocol-based T.38. The command `no mgcp fax t38 inhibit` is required for both NSE and protocol-based T.38, but it is the default.

MGCP does not offer any version 3 support.

**SCCP:**

SCCP does not support protocol-based (version 0 or version 3) T.38

Note: These two commands are needed to go from the default configuration of NSE-based T.38 to protocol-based T.38. The command `no mgcp fax t38 inhibit` is required for both NSE and protocol-based T.38, but it is the default.

MGCP does not offer any version 3 support.
QoS Considerations
Are VoIP QoS Policies Sufficient for Fax?

- In many cases, fax devices are located where VoIP is already installed.
- You can “piggyback” fax traffic onto the same QoS policy implemented for VoIP.
- As a rule of thumb, if good voice quality exists between locations, then fax communications should also work.
QoS Network Factors

- **Delay or latency**: the amount of time it takes a packet to travel from source to destination
- **Packet loss**: the amount of packets that are unsuccessful in arriving at the destination
- **Jitter**: the measure of the variability over time of the latency across a network
Fax over IP (FoIP) is generally more affected by packet loss than VoIP.

Ideally no packet loss should occur for a fax call.

If packet loss is present then use T.38 with redundancy.

Enabling T.38 redundancy requires more bandwidth.
FoIP and Delay

- Delay is not as impacting to FoIP compared to VoIP
- FoIP calls have been known to handle network delays of 1 second or more
- However, as a best practice it is still recommended to minimize network delays as much as reasonably possible because too much delay will cause FoIP calls to fail
- Watch out for multiple IP and PSTN hops and satellite links
FoIP and Jitter

- Variably spaced T.38 packets arrive at the playout buffer and some may even be out of sequence.
- Packets are re-sequence if necessary and placed in the required order for playout.
- Evenly spaced packets are played out to the DSP for transmission on the PSTN.

- T.38 and Cisco fax relay use 300 ms fixed jitter or playout buffers.
- Fax passthrough uses the last voice mode setting before the switchover.
- With large playout buffers, FoIP can handle larger amounts of jitter than VoIP but as a best practice it is still recommended to keep jitter to a minimum.
## QoS Design Parameters for Fax

<table>
<thead>
<tr>
<th></th>
<th>Delay</th>
<th>Jitter</th>
<th>Packet Loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice</td>
<td>&lt; 150 ms</td>
<td>&lt; 30 ms</td>
<td>&lt; 1%</td>
</tr>
<tr>
<td></td>
<td>(one-way,</td>
<td>(average,</td>
<td></td>
</tr>
<tr>
<td></td>
<td>mouth to ear)</td>
<td>one-way)</td>
<td></td>
</tr>
<tr>
<td>Fax</td>
<td>&lt; 1000 ms</td>
<td>&lt; 300 ms</td>
<td><strong>None</strong>, unless using T.38 with redundancy</td>
</tr>
<tr>
<td></td>
<td></td>
<td>for fax relay,</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>&lt; 30 ms for passthrough</td>
<td></td>
</tr>
</tbody>
</table>

*Fax passthrough is very sensitive to packet loss and may be able to handle 0.1%–0.2% loss depending on when in the fax transaction the loss occurs and if it is consecutive packets. Cisco fax relay can handle more loss than passthrough but T.38 with redundancy is still the best choice for fax calls when packet loss is occurring.*
QoS Marking Configuration on Gateways

- Fax, modem, and text QoS marking via DSCP can be simply added at the dial-peer level for H.323 and SIP or globally for MGCP with the command “mgcp ip qos dscp”

- Dial-peer QoS markings can be viewed with the command `show dial-peer voice`

- Default dial-peer QoS markings are `af31` for signaling and `ef` for media

- Cisco recommended best practice is to use `cs3` for signaling
QoS Marking and Shaping Example

CLASSIFICATION & MARKING  QUEUEING & DROPPING  POST-QUEUING OPERATIONS

dial-peer voice 919 voip
destination-pattern 919........
session target ipv4:10.0.0.19
fax protocol t38 ls-redundancy 0 hs-redundancy 0
fallback cisco
ip qos dscp cs3 signaling
no vad

Voice and T.38 fax traffic can share the same QoS marking and shaping.

class-map match-all fax_voice
match ip dscp ef
class-map match-any call_signaling
match ip dscp cs3
match ip dscp af31
!
policy-map WAN
class fax_voice
priority percent 33
class call_signaling
bandwidth percent 5
class class-default
fair-queue
Design Best Practices
Design Best Practices

- Unified Communications Integration Methods
  - Cisco Unified Communications Manager, Gateway-Controlled
  - Cisco Unified Communications Manager, CA-Controlled

- Design considerations
  - Bandwidth utilization
  - Super G3
  - Protocol and product support
  - SIP Trunking and FoIP Support
  - T.38: de facto standard
Gateway to Gateway

- All signaling and media packets are exchanged between the voice gateways
- NSE-based switchovers (fax passthrough and NSE-based T.38) and payload type switchovers (Cisco fax relay) happen within the voice media stream while protocol-based switchovers (fax pass-through and protocol-based T.38) happen within the call control protocol
Cisco Unified Communications Manager, Gateway-Controlled

- Cisco Unified Communications Manager (Unified CM) controls the setup of the initial voice call between the gateways using standard call control signaling such as H.323, SIP, MGCP, and SCCP.

- The switchover to a fax media stream is handled by the gateways themselves without the knowledge of Unified CM.

Voice media is established by Unified CM using the H.323, SIP, MGCP, or SCCP protocols.

Switchover for fax passthrough, NSE-based T.38, and Cisco fax relay occurs within the voice media stream.
Cisco Unified Communications Manager, Gateway Controlled

- Use Cisco Unified Communications Manager, Gateway-Controlled, when a Cisco Unified Communications Manager version does not support a certain fax media negotiation in the call control protocol (like T.38)
- NSE-based T.38 allows for T.38 to occur even if a Cisco Unified Communications Manager version does not offer T.38 support within the call control protocol
- This integration method must also be used with non-standard switchover mechanisms such as fax passthrough and Cisco fax relay
- Remember that NSE and PT switchovers are Cisco proprietary and this integration method only works with Cisco devices and not third-party equipment
Cisco Unified Communications Manager controls the setup of the initial voice call between the voice gateways and controls the switchover to T.38 fax relay.

Instead of the gateways independently handling the switchover, Cisco Unified Communications Manager coordinates the switchover using the call control protocol.
Cisco Unified Communications Manager, CA-Controlled

- Recommended as a best practice!
- Only protocol-based T.38 (using H.323, SIP, or MGCP) is capable of this integration method
- This integration method is standards-based and allows for interoperability with third-party devices such as gateways and fax servers
- If Cisco Unified Communications Manager and voice gateways support this integration method then this is recommended compared to the Cisco Unified Communications Manager, Gateway-Controlled method which depends on proprietary switchover mechanisms
- T.38 support within Cisco Unified Communications Manager varies depending on software version
Cisco Unified Communications Manager and T.38 Support

<table>
<thead>
<tr>
<th>Call Control Protocol Support for T.38</th>
<th>Cisco Unified Communications Manager Software Release</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.323</td>
<td>4.1(1), 4.2(3), 5.0(1), and 6.0(1)</td>
</tr>
<tr>
<td>H.323 and MGCP</td>
<td>4.2(3) and 6.0(1)</td>
</tr>
<tr>
<td>H.323 and SIP</td>
<td>5.0(1) and 6.0(1)</td>
</tr>
<tr>
<td>H.323, SIP, and MGCP</td>
<td>6.0(1) and later</td>
</tr>
</tbody>
</table>

- Only release 6.0(1) and later offers support for T.38 within the call control protocols of H.323, SIP, and MGCP
- The SCCP protocol only handles T.38 using an NSE-based switchover
Design Best Practices

- Unified Communications Integration Methods
  - Cisco Unified Communications Manager, Gateway-Controlled
  - Cisco Unified Communications Manager, CA-Controlled

- Design considerations
  - Bandwidth utilization
  - Super G3
  - Protocol and product support
  - SIP Trunking and FoIP Support
  - T.38: de facto standard
FoIP Bandwidth Utilization

- Different FoIP transports use varying amounts of bandwidth
- On links where saving bandwidth is a priority then relay is a better choice
- T.38 redundancy handles packet loss much better than fax passthrough/pass-through with redundancy

<table>
<thead>
<tr>
<th>Codec</th>
<th>Bandwidth¹</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711 (64 Kbps)</td>
<td>83 Kbps</td>
</tr>
<tr>
<td>G.729 (8 Kbps)</td>
<td>27 Kbps</td>
</tr>
<tr>
<td>G.723 (6.3 Kbps)</td>
<td>19 Kbps</td>
</tr>
<tr>
<td>Fax passthrough/pass-through (G.711)</td>
<td>83 Kbps</td>
</tr>
<tr>
<td>Fax passthrough (G.711) with redundancy</td>
<td>170 Kbps</td>
</tr>
<tr>
<td>T.38 (no redundancy)</td>
<td>25 Kbps²</td>
</tr>
<tr>
<td>T.38 (redundancy level 1)</td>
<td>41 Kbps²</td>
</tr>
<tr>
<td>T.38 (redundancy level 2)</td>
<td>57 Kbps²</td>
</tr>
<tr>
<td>Cisco fax relay</td>
<td>48 Kbps²</td>
</tr>
</tbody>
</table>

¹Values are approximate with Ethernet or Frame Relay headers
²Values are peak and only occur during the sending of a page at 14.4 Kbps; gateways can force lower fax speeds for additional bandwidth savings
Methods for Handling SG3 Fax Calls

- The **sg3-to-g3 spoofing feature** forces Super G3 fax calls to negotiate down to G3 speeds for compatibility with T.38 or Cisco fax relay.

- **Modem passthrough** will handle SG3 calls at their native speeds.

- **Fax pass-through** is not compatible with SG3 because of its dependency on the V.21 flags found in G3 negotiations for initiating the switchover.

- Starting in IOS 15.1(1)T SG3 over T.38 is supported and these other methods for handling SG3 faxes are not necessary.
Error Correction Mode (ECM)

- ECM ensures error-free transmission of fax pages
- When ECM is disabled, fax pages can proceed with minor errors that may or may not be noticeable
- Cisco gateways can force ECM to be disabled when configured for T.38 or Cisco fax relay
- Disabling ECM may be useful for scenarios where network impairments make it difficult for ECM to be efficient
T.38 Fallback

dial-peer voice 100 voip
fax protocol t38 ls-redundancy 0
hs-redundancy 0 fallback cisco

Sending Fax
OGW
T.38
IP
6608 TGW
Receiving Fax
Cisco fax relay
T.38 call fails to 6608 but fallback to Cisco fax relay is successful

- For increased interoperability, T.38 fax relay can fall back to Cisco fax relay or fax pass-through if the initial T.38 negotiation fails

- T.38 also has an automatic fallback mechanism that tries a protocol-based T.38 fax relay switchover if the configured NSE-based switchover fails
## FoIP Call Control Protocol Support

<table>
<thead>
<tr>
<th>Transport Method</th>
<th>H.323</th>
<th>SIP</th>
<th>MGCP</th>
<th>SCCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco fax relay</td>
<td>Yes*</td>
<td>Yes*</td>
<td>Yes*</td>
<td>Yes*</td>
</tr>
<tr>
<td>NSE-based T.38</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Protocol-based T.38</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Fax pass-through</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>Modem passthrough</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
</tbody>
</table>

*IOS platforms such as the 5350, 5400, and 5850 utilizing the Nextport DSP cards do not support Cisco fax relay
FoIP Product Support Caveats

- Cisco ATA only supports NSE-based passthrough (modem passthrough)
- SCCP voice gateways (IOS or VG248) do not support any protocol-based switchovers (like fax pass-through and protocol-based T.38), so NSE-based T.38 and modem passthrough must be used
- MGCP gateways do not support fax pass-through, only modem passthrough can be utilized
- Legacy Nextport DSP platforms (AS5350, AS5400, and AS5850) do not support Cisco fax relay
- Legacy 6608 and 6624 cards only support Cisco fax relay and modem passthrough
SIP Trunks and FoIP

- SIP Trunks typically use three different methods to transport standard G3 fax calls
  - **G.711 Voice** - All calls (voice and fax) are always sent using the G.711 voice codec
  - **Pass-through** - A SIP RE-INVITE message is used to up-speed to G.711 when a fax tone is detected
  - **T.38 Fax Relay** – Call is switched to T.38 fax relay using a SIP RE-INVITE when a fax tone is detected

<table>
<thead>
<tr>
<th>Fax Method</th>
<th>T.38 Fax Relay</th>
<th>G.711 Voice</th>
<th>Pass-through</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Pros</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Highest fax success rates can be achieved</td>
<td>Most widely deployed</td>
<td>Provides bandwidth savings as G.729 voice call only upspeeds to G.711 if call is fax</td>
</tr>
<tr>
<td></td>
<td>Cleanest solution from signaling and media point of view</td>
<td>Simplest solution</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Use less bandwidth than G.711</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Cons</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Interoperability between different vendors</td>
<td>Consumes a large amount of bandwidth for all calls</td>
<td>Interoperability between different vendors</td>
</tr>
<tr>
<td></td>
<td>Not offered by many Service Providers</td>
<td>Sensitive to impairments, no redundancy</td>
<td>Not supported by Cisco Unified Communications Manager</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Sensitive to impairments, no redundancy</td>
</tr>
</tbody>
</table>
### T.38: De Facto Standard for FoIP

<table>
<thead>
<tr>
<th>Feature</th>
<th>T.38</th>
<th>CFR</th>
<th>Fax Passthrough</th>
<th>Fax Passthrough</th>
</tr>
</thead>
<tbody>
<tr>
<td>Standards-Based, Third-Party Integration</td>
<td>✔</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Low Bandwidth</td>
<td>✔</td>
<td>✔</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Redundancy</td>
<td>✔</td>
<td></td>
<td></td>
<td>✔</td>
</tr>
<tr>
<td>Fallback Mechanisms</td>
<td>✔</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SRTP and CRTP Support</td>
<td>✔</td>
<td></td>
<td></td>
<td>✔</td>
</tr>
<tr>
<td>Legacy Platform Support</td>
<td>✔</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**T.38 Should Always Be Your First Choice When Handling FoIP Traffic!**
Troubleshooting
Call Troubleshooting Stages

- Any fax call can be broken down into three troubleshooting stages
  1. Initial Voice Call
  2. Switchover
  3. Fax Media

- Once the problem has been narrowed down to a specific stage, then the appropriate troubleshooting strategy can be implemented
Packet Captures

- Packet captures are one of the best tools for troubleshooting fax problems
- Ideally captures should be made at both endpoints to ensure that both signaling and media are seen
- Cisco Unified Communications Manager only participates in the call control signaling
Packet capture software such as Wireshark is one of the best tools for troubleshooting either NSE or protocol-based switchovers as well as the fax media.
Packet Captures: VoIP Call Analysis

- For protocol-based switchovers like fax pass-through and T.38, the “VoIP Call Analysis” feature in Wireshark offers a graphical means for analyzing the switchover.

- “VoIP Call Analysis” also breaks down the T.38 media stream, showing the T.30 signaling and page transmission.

- Select “Statistics” from top of Wireshark window, “VoIP Calls”, click on a call, and then click “Graph”
Troubleshooting NSE-Based Switchovers

<table>
<thead>
<tr>
<th>debug voip rtp session named-event</th>
</tr>
</thead>
<tbody>
<tr>
<td>Jan 10 22:01:58.471:</td>
</tr>
<tr>
<td>timestamp 0x20631C26</td>
</tr>
</tbody>
</table>
| Jan 10 22:01:58.471:               | Pt:100 Evt:192 Pkt:00 00 00 <Snd>>>
| Jan 10 22:01:58.483:               | s=VoIP d=DSP payload 0x64 ssrc 0x9A20101 sequence 0x18C2 |
| timestamp 0xADA80F0A               |                                                       |
| Jan 10 22:01:58.483:               | <<<Rcv> Pt:100 Evt:192 Pkt:00 00 00 |

- Use the command `debug voip rtp session named-event` to view NSE-related switchovers and events on IOS gateways, such as fax passthrough and NSE-based T.38

- Most common cause of problems with NSE-based switchovers are an incorrect configuration or a transcoder or proxy in the call path
Packet Captures: NSE Switchover

The default RTP payload type for NSE messages is 100.

The NSE Event ID is shown in the payload as a hex value. In this case 0xC8 is 200 in decimal (NSE-200), which is a T.38 fax relay switchover request.
Troubleshooting PT-Based Switchovers

Like NSE-based switchovers, confirm RTP payload type (PT) switchovers as utilized by Cisco fax relay using the command `debug voip rtp session named-event`
Troubleshooting Protocol-Based Switchovers—H.323 and T.38

H.245 Request Mode message (from the terminating side) initiates the T.38 switchover

*Feb 3 04:34:53.070: H245 MSC OUTGOING PDU ::= value MultimediaSystemControlMessage ::= request : requestMode :

<SNIP>

  type dataMode :
  {
    application t38fax :
    {
      t38FaxProtocol udp : NULL
      t38FaxProfile
      {
        fillBitRemoval FALSE
        transcodingJBIG FALSE
        transcodingMMR FALSE
        version 0
        t38FaxRateManagement transferredTCF : NULL
        t38FaxUdpOptions
        {
          t38FaxMaxBuffer 200
          t38FaxMaxDatagram 72
          t38FaxUdpEC t38UDPRedundancy : NULL
        }
      }  
    }
  }

<SNIP>
Troubleshooting Protocol-Based Switchover—SIP and T.38

SIP re-INVITE message (from the terminating side) initiates the switchover to T.38

*Feb 3 22:32:18.514: //1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg: Sent:
INVITE sip:172.18.110.86:5060 SIP/2.0
Via: SIP/2.0/UDP 14.1.97.15:5060;branch=z9hG4bK618FF
From: <sip:100@14.1.97.15>;tag=4320068C-1F6B
<SNIP>
v=0
o=CiscoSystemsSIP-GW-UserAgent 8190 1299 IN IP4 14.1.97.15
s=SIP Call
c=IN IP4 14.1.97.15
t=0 0
m=image 16582 udptl t38
c=IN IP4 14.1.97.15
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:72
a=T38FaxUdpEC:t38UDPRedundancy
Troubleshooting Protocol-Based Switchover—MGCP and T.38

A NTFY containing FXR/t38(start) initiates the MGCP T.38

*Feb 4 17:01:36.597: MGCP Packet sent to 172.18.110.85:2427--->
NTFY 935850485 S2/SU0/DS1-0/23@CAE
- DH
- 3845
- 2

MGCP 0.1
N: ca@172.18.110.85:2427
X: 17
O: FXR/t38(start)
<---

CAE-DH-3845-2#show ccm-manager
MGCP Domain Name: CAE-DH-3845-2
Priority       Status          Host
==============================================
Primary Registered CAE-DH-7845 (172.18.110.85)
First Backup  None
Second Backup None

Current active Call Manager: 172.18.110.85
<SNIP>

TIP: Omission of the configuration command mgcp package-capability fxr-package is the most common cause of MGCP T.38 failing
Troubleshooting Protocol-Based Switchovers With Cisco RTMT

- Cisco Real Time Monitoring Tool (RTMT) log viewer can show protocol-based switchover information
Troubleshooting the Fax/Modem Media: Gateway DSP Statistics

- DSP packet statistics on Cisco voice gateways provide IP network information about the active, incoming media stream for fax/modem passthrough, Cisco fax relay, and T.38
- Errors in these stats typically indicate packet loss and/or jitter problems
- The command “show call active voice brief” also provides some DSP statistics and information about the media stream
Fax Relay Message Debugging

- The command `debug fax relay t30 all-level-1` displays T.30 messages for T.38 and Cisco fax relay from the DSP’s perspective on the POTS/PSTN side of the gateway.
- Inbound messages are flagged `fr-msg-det` and messages sent by the DSP out the telephony interface are indicated by `fr-msg-tx`
Sample Output From “debug fax relay t30 all-level-1”

Nov 2 17:19:22.928: 0/0/0 (42) 626612205 fr-entered=10(ms)
  timestamp=626613325 fr-msg-det NSF
  timestamp=626614305 fr-msg-det CSI
  timestamp=626615005 fr-msg-det DIS
  timestamp=626617385 fr-msg-tx TSI
  timestamp=626618185 fr-msg-tx DCS
  timestamp=626623295 fr-msg-det CFR
  timestamp=626648365 fr-msg-tx EOP
  timestamp=626649985 fr-msg-det MCF
  timestamp=626652165 fr-msg-tx DCN

Nov 2 17:20:03.208: 0/0/0 (42) 626652485 fr-end-dcn

- This debug is only valid for T.38 and Cisco fax relay as the gateway plays an active role in modulating and demodulating the T.30 fax messages
Tips for Using “debug fax relay t30 all-level-1”

- Primary debug used for troubleshooting T.38 and Cisco fax relay problems on a Cisco voice gateway
- Only low-speed T.30 messages are displayed so high speed communications such as trainings and page transmissions are not shown
- If there are multiple fax calls occurring at once you can specify a called or calling number to only view the debug messages for a specific call
- The “level-2” option for this debug is only readable by the DSP vendor and it should not be enabled
- This debug (as well as packet captures) is useful for viewing unique T.30 problem signatures
Unique T.30 Debug Signature: Delay

T.30 Negotiation Failure With a Likely Cause of Too Much Delay Between the Fax Endpoints

- T.30 messages are not being received before previous messages are retransmitted
- Common causes of delay are satellite links or multiple VoIP hops
- Adjusting the playout buffer on Cisco voice gateways may help
- For example, configure `playout-delay fax 100` under the appropriate dial-peer to lower the playout buffer to 100 ms
Unique T.30 Debug Signature: Corruption

- Fax machine is receiving a corrupted training signal and sending an FTT message.
- Most common cause of a corrupted training is PSTN impairments, such as clock slips.
- First step is to always confirm that any digital circuits are free of errors.
Viewing Digital Circuit Errors

T1 0/1/0 is up.
Applique type is Channelized T1
Cablelength is long gain36 0db
No alarms detected.
alarm-trigger is not set
Soaking time: 3, Clearance time: 10
AIS State:Clear LOS State:Clear LOF State:Clear
Version info Firmware: 20060711, FPGA: 13, spm_count = 0
Framing is ESF, Line Code is B8ZS, **Clock Source is Line.**
CRC Threshold is 320. Reported from firmware is 320.
Data in current interval (729 seconds elapsed):
  0 Line Code Violations, 0 Path Code Violations
  **24 Slip Secs**, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
  **24 Errored Secs**, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
Total Data (last 24 hours)
  0 Line Code Violations, 0 Path Code Violations,
  2989 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins,
  2989 Errored Secs, 0 Bursty Err Secs, 0 Severely Err
The TCF or training signal is a pattern of all 0s

Other hex values shown in the TCF decode can cause the training to fail

This training pattern can be checked within Wireshark to confirm its integrity
Scan Line Corruption

In addition to training problems, PSTN errors and impairments can also cause scan lines to become corrupted.

Problem will not be viewable with Error Correction Mode (ECM) faxes but instead faxes will take a long time to complete or fail completely.

Gateway upgrades:

As stated in the introduction, you will need to upgrade any e-mail gateways individually. Note that this upgrade should be done after the server upgrade.

1. To upgrade a gateway, run the installation from the disk and select Custom Install. As before, on the installed modules should be selected. Make sure that only one of the gateways is selected and possibly the PS & PCL converters in the advanced features (this is used for the PDF notifications).
Audio Trace of Scan Line and Training Corruption
Preventing Corruption of the Training and Scan Lines

- Ensure that all digital circuits in the fax path are error free.
- Configure the clocking correctly on Cisco voice gateways to avoid slips on digital PSTN interfaces, which are notorious for causing training and scan line corruption.
- Make sure that the IP path is free from jitter and packet loss.
- Major corruption problems prevent the fax endpoints from training while less severe problems may allow training to occur but fax page data is affected.
Fax Servers
What Is a Fax Server?

- A software application running on a server that processes fax calls over a PSTN or IP connection
- A “Cisco Unified Communications Manager” for handling fax traffic
- Provides secure, automated, and efficient handling of an organization’s fax traffic
Fax Server Components

- Server hardware requirements are defined by fax server vendors
- Application software is produced by a variety of vendors and it handles administration, configuration, and integration functions
- Fax engine is either hardware (typically a PCI fax board installed into the server) or software-based
# Why Migrate to a Fax Server?

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Security</td>
<td>Faxes can be directed to a specific user or storage system with limited access</td>
</tr>
<tr>
<td>Workflow Integration</td>
<td>Fax servers can help automate process management procedures and specific workflow applications</td>
</tr>
<tr>
<td>Archiving</td>
<td>Faxes can efficiently stored and indexed to comply with industry mandates such as Sarbox or SOx and HIPAA</td>
</tr>
<tr>
<td>Accountability</td>
<td>All faxes sent or received along with confirmation of their success or failure can be handled with a fax server solution</td>
</tr>
<tr>
<td>MFP Integration</td>
<td>MFPs integrate directly with fax servers for increased cost savings and flexibility</td>
</tr>
<tr>
<td>Email Integration</td>
<td>Users have the ability to send/receive faxes directly via email</td>
</tr>
</tbody>
</table>
Fax Server Access Methods

- Users can access a fax server through interfaces such as a web browser, email, an MFP, and client software.
- Other devices and automated processes can communicate directly with a fax server through different APIs.
Fax servers communicate with Cisco Unified Communications Manager using H.323 or SIP.

Regardless of the call control protocol between the fax server and Cisco Unified Communications Manager, H.323, SIP, or MGCP can be used between Cisco Unified Communications Manager and the gateway.

T.38 media flows between the fax server and the voice gateways.
Integrating a Fax Server with a SIP Trunk

- T.38 fax relay is the best choice for sending faxes over SIP trunks
- No special modifications or configurations are typically necessary on a fax server to utilize SIP trunks, especially when a border element is present
Cisco Fax Server: Centralized Solution

- Cisco fax server is the Open Text Fax Server (RightFax) application on a Cisco MCS server
- A full featured, scalable enterprise solution that is centralized and supports both hardware and software-based fax engines to handle various integration solutions
- Deployed internally by Cisco in conjunction with Multifunction Peripherals (MFPs)
AXP-XMediusFAX: Branch Solution

- Branches typically have little IT infrastructure and staff to maintain server and application hardware
- Integrate fax server directly into the local ISR
Cisco AXP: XmediusFAX Solution

- XmediusFAX server on an ISR blade
- AXP runs a hardened Linux OS and provides a virtualized hosting environment
- XmediusFAX integrates directly with ISR architecture and is able to be configured and managed through the IOS CLI
- Functionality of a standalone fax server with no footprint
- Addresses the need for efficiently handling fax traffic at the local branch level
Complete Your Online Session Evaluation

- Give us your feedback and you could win fabulous prizes. Winners announced daily.
- Receive 20 Cisco Preferred Access points for each session evaluation you complete.
- Complete your session evaluation online now (open a browser through our wireless network to access our portal) or visit one of the Internet stations throughout the Convention Center.

Don’t forget to activate your Cisco Live and Networkers Virtual account for access to all session materials, communities, and on-demand and live activities throughout the year. Activate your account at any internet station or visit www.ciscolivevirtual.com.