Interworking with Cisco

- Presales questions
- Fax server talking to Cisco Gateway:
  - Configuring
  - Troubleshooting
- Fax server talking to Cisco Call Manager:
  - Configuring
  - Troubleshooting

Objectives for Presales Section

1. Pre-questions to ask customers (FoIP checklist)
2. Supported Cisco topologies (network deployments)
Intro: The Major Elements in a Cisco FoIP Network

- PSTN
- IP
- GW
- CM

Dialogic® Brooktrout®
SR140 Fax Software or
Dialogic® Brooktrout®
TR1034 Fax Board configured for IP; Cisco
Fax Server

Note: Cisco Call Manager CCM = Unified Call Manager =
Unified Communications Manager CUCM

Note: IOS of GW 12.4 (or 12.3T)

THE Question

- Interworking with Cisco gateway (GW)

OR

- Interworking with Cisco Call Manager (CM)?

Questions to Ask

- Call Manager?
- MGCP?

- Dialogic® Brooktrout®
- SR140 Fax Software talks directly to Gateway and
- routing via dial-peers

- Note 1: MGCP = media gateway control protocol
- Note 2: When MGCP, other less elegant solutions are: Dialogic® Brooktrout® TR1034
- Fax Board E1/T1 instead of IP, an extra GW, or an extra E1/T1 line and E1/T1 module
- in the GW
CCM/CUCM

- CCM = Cisco Call Manager
- CUCM = Cisco Unified Call Manager
- The Cisco IP-PBX
- Intelligence, call routing, billing, traffic analysis, voice prioritization centralized, managed and maintained via a GUI interface, e.g., adding later on an extra fax number outside of the already configured fax range can be done by a non-Cisco specialist as opposed to having to know the command to add this to the gateway’s Internetwork Operating System (IOS)
- Gateways are the endpoints for originating and terminating calls to and from the fax server while the Call Manager handles the call routing
  - Call Manager routes and sets up calls; media flows between User Agents (UA) and gateways
- MGCP is a media gateway control protocol; not a call control protocol
- Centralized architecture; fax server can easily talk to multiple gateways via the CCM

Remark on Cisco’s MGCP Installed Base

- MGCP – Media Gateway Control Protocol
  - Allows configuration of gateways set up for MGCP to reside on Call Manager/CUCM
  - Centralized control of call management
- Cisco tried to push MGCP, Cisco has since adopted SIP
- Cisco’s SIP was lacking certain ISDN features
- Until recently, Europe continued to depend on H.323 and MGCP

Questions to Ask

- Dialogic® Brooktrout® SR140 Fax Software talks to gateway or
  - H.323
  - H.323 gatekeeper?
  - SIP
  - SIP proxy?
**Dialogic® Brooktrout® SR140 Fax Software Talks to Gateway**

- Use dial-peers in GW config to route the calls

```plaintext
PSTN  SIP  GW  RTP/T.38
```

Dialogic® Brooktrout® SR140 Fax Software

**Dialogic® Brooktrout® SR140 Fax Software Talks to Gateway**

- Use dial-peers in GW config to route the calls

```plaintext
PSTN  H.323  GW  RTP/T.38
```

Dialogic® Brooktrout® SR140 Fax Software

**Dialogic® Brooktrout® SR140 Fax Software Talks to Call Manager**

- Set up route patterns in the Call Manager and dial-peers in the GW to route the calls
- Advantageous to use CM when no fax number range

```plaintext
PSTN  H.323  CM  H.323
```

Dialogic® Brooktrout® SR140 Fax Software

```plaintext
≥ 4.1.3
5.x
6.x
RTP/T.38
```
Set up route patterns in the Call Manager and dial-peers in the GW to route the calls.

Remark: You can also mix SIP and H.323.
Summary of Integration with Call Manager

- Different versions of Call Manager offer varying amounts of support for T.38 fax. Only the latest versions (6.x) provides T.38 support for H.323, SIP, and MGCP.

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

Call Manager...

- Not enough to say whether the Dialogic piece will talk to Call Manager
- Important to know what the version of Call Manager is, what it is talking to on the 'other side', what protocols it is using to talk to all the pieces

Configuration - CM

- Call Manager handles all call routing and call control signaling to the voice gateways when a fax server is directly connected
- For example, a fax server connected via H.323 to Call Manager can communicate with H.323, SIP, and MGCP voice gateways via T.38 FoIP
Remark on T.38 with MGCP Gateway

- No combination of modem-passthrough with T.38 possible in Cisco network when MGCP is used
  - i.e., no transition phase when moving to T.38 fax
  - Old ATA devices stop working
- No such problem with a H.323 gateway

Summary of Cisco Gateway Models and IOS Versions

- Several specific devices of the below models have been tested; other models like Catalysts and 1760 have been used too
- H.323 and SIP
- Min. IOS 12.3T or 12.4
- Also with H.323 IOS Cisco gatekeeper

Cisco 2600/2800 Series Routers
Cisco 3700/3800 Series Routers
Cisco AS5400 Series Universal Gateway

FoIP Interoperability

- Interoperability between the Cisco Gateway, Call Manager or Gatekeepers to Dialogic® Brooktrout® SR140 Fax Software is at the protocol level.
  - H.323
  - SIP
  - T.38
- Ask our tech support for the very detailed “Cisco Configuration Guide”
  - Has a chapter with a reference installation for each type of topology
  - Dialogic® Brooktrout® SR140 Fax Software configuration, gateway and Call Manager configuration included
  - Cannot be provided to a third party
Installation

- Find out first the Cisco network topology of your customer
  - What does the customer already have?
  - What does the customer want for fax?
- Make sure to have a Cisco specialist on site when you install the fax server
  - Easy to get the fax server installed, but not the Cisco network piece
  - Test back to back first!

Configuration of Cisco Gateway

- All recent Cisco voice gateways running IOS (Internet work Operating System) versions 12.3T and later should interoperate with Dialogic-based fax servers using the H.323 or SIP call control protocols.
- T.38 fax relay must be explicitly configured on Cisco voice gateways since the proprietary Cisco fax relay is the default.
- By default Cisco voice gateways utilize the G.729 voice codec. This must be changed to the G.711 codec in order to interoperate with Dialogic-based fax servers.

Configuration of Cisco Gateway

- Communicating with the CGW:
  - Telnet
  - HyperTerminal
- Commonly used commands:
  - Enable: enter privileged mode
  - Will see # sign at prompt – indicator you are in privileged mode
  - Config terminal: allows to change the configuration
  - Write memory: to save configuration changes permanently to NVRAM
Call Control Configuration – Cisco Gateway

- ISDN
- SIP
  - “voice service voip
  - Sip”
  - Dial-peer voice xxx voip
  - Session protocol sipv2
- H.323
  - “voice service voip
  - H323”
  - Dial-peer voice xxx voip
  - Voice-class h323 1

Dial-peer voice xxx voip

T.38 Configuration – Cisco Gateway

- "fax protocol 138"
  - Global: under "voice service voip"
  - Not global: in voip dial-peer
- Initial call setup is voice
  - The codec has to be G.711 (not G.729)
  - T.38 switchover
    - In SIP, by a re-INVITE message
    - In H.323, by the H.245 RequestMode message
- T.38 using UDP is only supported by Cisco and Dialogic®
  Brooktrout® Fax Products; no TCP
Configuration in Particular – Cisco Gateway

- H.323 ISDNTelco: does not expect “unknown”
  - GW: converting H.323 Q.931 into ISDN Q.931 without translation rules
  - Configurable from Dialogic® Brooktrout® SDK 5.2 on:
    - H323_OVERRIDE_NUMBERINGPLAN
    - H323_OVERRIDE_NUMBERINGTYPE

Troubleshooting – Cisco Gateway

- Where, in the network, does it go wrong?
- When, at what moment of the call, does it go wrong?

Troubleshooting – Cisco Gateway (CGW)

- Each call in CGW has 2 call legs, usually a
  - POTS leg
  - VoIP leg
Troubleshooting – Cisco Gateway (CGW)

- Informational; the Cisco support admin or Cisco TAC should handle configuration and debugging of the Cisco pieces
- Getting the context info:
  - Cisco model, type of PSTN interfaces, IP address of CGW, IP address of fax server host, IP address of Dialogic® Brooktrout® TR1034 Fax Board Ethernet, IP address of gatekeeper/proxy
  - "Show run/show config" to get the CGW configuration
  - "Show run | begin xxx" to get the IOS version
- Enable viewing of Cisco debugs:
  - "Terminal monitor"
- Log debug output to a log file
- Enable millisecond timestamps:
  - "service timestamps debug datetime msec" in configuration mode
- To disable debugging:
  - "no debug all"

Call Control Troubleshooting – Cisco Gateway

- Debug ISDN Q931: ISDN layer 3 protocol trace – establishing and terminating of an ISDN call
- Show Controller E1: ISDN layer 1 alarms?
  - Check when you are faxing but is "bad quality" and taking too long: FTT, PPR, fax messages (in API debug log)
  - Check when fax messages are not received in time and the fax device is already retransmitting the previous message
- Show ISDN Status: L1, L2, L3 states
- Show ISDN Service: status of each channel
- Debug isdn error
- Note: [http://www.linkbit.com/support-decoder.html](http://www.linkbit.com/support-decoder.html)
- Errors on ISDN side will have negative effect on IP side
Call Control Troubleshooting – Cisco Gateway

- Which dial-peer is my call hitting?
- Show call active voice brief
- Show call active fax brief
- Show call history voice brief
- Show call history fax brief
- Only for POTS leg

- Other commands that might be useful:
  - Show dial-peer voice xxx
  - Show dial-peer voice summary
  - Show dialplan number xxx
  - Debug voip dialpeer all
  - Show interface fast0/0: Ethernet interface

T.38 Troubleshooting – Cisco Gateway

- Debug Fax Relay T30 All:
  - The T.30 messages that got demodulated at Cisco DSP level
  - The T.30 messages in and out of the DSP on the POTS leg
  - Turn on before the call attempt

Call Control and T.38 Troubleshooting

- Network sniffer: Ethereal/Wireshark
- Best way to really see what is occurring on the VoIP leg
- Unbiased third party tool
  - Jitter?
  - Packet loss?
  - Protocol problems?
SIP - Problem Solving – Cisco Gateway

- SIP disconnect cause code: 404:
  - Number being passed into the Cisco is not being recognized
    => check dial-peers

T.38 Problem-Solving – Cisco Gateway (CGW)

- FTT, PPRs, ... in API debug log “debug fax relay 130 all”
  => Synchronization problem? -> “Show controller E1” -> clocking configuration
- Messages are not being received in time before the fax device retransmits the previous message
- Delay?
  - CGW jitter buffer: default 300 ms -> lower: “playout-delay fax 100”
  - Remove sources of delay
- Packet loss? -> sniffer

Configuration of Fax Server – with CallManager

- SIP
  - Defaults
    - Media_renegociate_delay_outbound: -1 (try also 2000 when troubleshooting the T.38 switchover)
  - H.323
    - Defaults
    - Except H.323 advanced settings:
      - H323_Faststart = 0
      - H323_H245Stage = 3
      - H323_H245Tunneling = 0
    - Media_renegociate_delay_outbound: -1 (try also 2000 when troubleshooting the T.38 switchover)
Configuration of Gateway – with CallManager

- G.711 codec required
- No T.38 inhibit
- Fax-package required on MGCP gateway for T.38
  - Implies that modem-passthrough stops working

Configuration on CallManager

- H323: fax server as H.323 Inter-Cluster trunk (as H.323 gateway should also be possible)
- SIP: fax server as SIP trunk
  - Non-secure SIP trunk on UDP
- Add the necessary Route Patterns

CallManager 6.1 Login Screen
Configuration in Particular - CallManager

- Fax devices taking several rings before picking up

- Cause at CUCM: wants the media to start before the connection happened => change timer configuration: e.g., Media Exchange Interface Capability Timer --> from 8 to 16 and Media Exchange Timer --> from 12 to 20
- From Dialogic® Brooktrout® SDK 5.2 on:
  - H323_MediaWaitForConnect

Call Control Troubleshooting on CallManager (CM)

- Unified CM versions prior to 5.x generate trace files to a specific directory much in the same sort of manner that trace files are generated for Dialogic® Brooktrout® SR140 Fax Software once certain trace parameters are enabled
- Use the Real Time Monitoring Tool (RTMT) tool for Unified CM versions 5.x and above after enabling the appropriate tracing options

Troubleshooting on CallManager (CM): Unified CM RTMT
Prerequisites - Connecting to a Cisco Gateway

- To connect to Cisco Gateway:
  - Prior knowledge of Cisco router installation and configurations
  - A basic understanding of Cisco networking and router/gateway configurations
  - Familiar with basic internet technology
  - Certified or experienced in Cisco IOS router/gateway technology
  - A thorough understanding of LAN/WANs, bridges, switches, protocols, and network management

Getting Cisco Technical Support

- To get technical support from Cisco, first obtain a username and password (for which you will provide your support contract)
- Then submit a TAC service request using the online web tool
- Open separate cases for each question if you have a wide variety of questions

Lab 2 – Dialogic® Brooktrout® SR140 Fax Software talking to Cisco

Lab 2 Exercise

1. Open up FVD tool
2. Select Dialogic® Brooktrout® SR 140 Fax Software
3. For SIP configuration:
   - A. Input GW IP address for Cisco 1751
   - B. Change “anonymous” field to your name
   - C. Verify PC IP address
4. “Save & Apply”
5. Start Packet Sniffer (Wireshark)
6. Test to fax machine, instructor laptop & phone
   - Stop!
7. Check with instructor before proceeding
8. Optional – repeat steps 3 - 6 for H.323
Example of Cisco Dial Peer – H.323

- Dial-peer voice 22 voip
- Description H.323 to BRKT/ CNTA host
- Destination-pattern 22
- Session target ipv4: 10.10.68.22
- Session transport udp
- Codec g711ulaw
- Fax rate 14400
- Fax nsf 000000
- Fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none

Example of Cisco Dial Peer – SIP

- Dial-peer voice 23 voip
- Description SIP to BRKT/ CNTA host
- Destination-pattern 23
- Voice-class codec 1
- Session protocol sipv2
- Session target ipv4: 10.10.68.22
- Session transport udp
- Dtmf-relay rtp-nre
- Fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback none

Interworking with Avaya
Interworking with Avaya

- Tested H.323 with Dialogic® Brooktrout® SR140 Fax Software (see Interworking Doc on web):
  - Avaya G650 Media Gateway with TN2302 Media Processor Circuit Pack (MedPRO – up to 64 channels) FW117
  - Avaya G350 Media Gateway (without MedPRO) with Avaya Communication Manager 4.0.1 (Load 731.2) on Avaya S8300 Server
- High density X-fire TN2602 Media Processor
  - Formerly an issue, now worked around in Dialogic® Brooktrout® SDK 5.2.x, also using fw 3.1.1 build 84 or later on SDK 5.1.x build
- Successful customer install H.323 with Brooktrout SR140 Fax Software:
  - Avaya Communication Manager 3.1.2 (with patch 12249) on Avaya 8400 server
  - Testing Completed 7/22/08

The tests were run using Dialogic® Brooktrout® SDK 5.2.1 build 14 on the following network configuration:
Dialogic Brooktrout TR1000 Media Board
(ISDN T1) <-> Avaya G650 <-> SR140 (H.323 T.38)

Configuration in Particular

- Called party number and Calling Party number (ANI) presented in "strange" way by Avaya in H.323:
- Example:
  - 12/06 16:12:20.23 [41,1] < (13) IP_CC (08) EVENT (01) SETUP_IND
  - (02) USER_REF [02: Fix Uns Long Unitless] 000B0002
  - (03) STACK_REF [02: Fix Uns Long Unitless] 0264D2C8
  - (01) CALL_STATE [00: Fix Uns Byte Unitless] 02
  - (09) CALLED_ADDRESS [10: Fix Uns Byte Char] "TEL:19441,19441"
  - (0A) CALLING_ADDRESS [10: Fix Uns Byte Char] "TA:10.126.192.105:1720,3170100"
  - (0E) CALL_TYPE [00: Fix Uns Byte Unitless] 00
- From SDK 5.2 on: H323_CalledPartyOption

Configuration in Particular

- Max_bitrate not implemented anymore with bstvb.dll from Dialogic® Brooktrout® SDK 5.1.3, version 50 on: 14400 baud used instead of 9600 baud
- Make sure that: T38_fax_version = 0
Interworking with Dialogic® DMG Media Gateway Series

Dialogic® DMG Media Gateway Series

- Tested with Dialogic® Brooktrout® SR140 Fax Virtual Board
  - SDK 5.2.1 and later
    - Dialogic® 3000 Media Gateway Series (DMG3000) and Dialogic® 4000 Media Gateway Series (DMG4000)
      - DMG3000 & DMG4000: Dialogic® Diva® media board with SIP control, Windows® OS server based
      - V.34 T.38 support currently
      - Running Diva® software, system release 8.5.1 or higher
    - SDK 5.2.4 and later
      - Dialogic® 1000 Media Gateway Series (DMG1000) and Dialogic® 2000 Media Gateway Series (DMG2000)
      - Gateway application version 5.1.118 or higher
      - V.34 T.38 support on roadmap for 2000 Series

Dialogic® 1000 Media Gateway Series (DMG1000)

- Product Overview
  - Connects VoIP and FoIP-based applications to a legacy PBX infrastructure with application feature support
  - Designed and tested against wide range of PBX infrastructures
  - Designed for ease of deployment and operation

- Product Specification
  - Small Appliance Form Factor (~10"x10")
  - Multiple SKUs for Specific PBX Models
  - 8 Port Digital PBX Station Emulation (4 SKUs)
  - 4 or 8 Port Analog FXO (2 SKUs)
  - VoIP Protocol Support
    - SIP (RFC 3261)
    - G.711, G.729AB, G.723.1
    - T.38 Fax over IP
  - Serial Protocols: SMDI, MCI, MD-110
  - G.168 Echo Cancellation
  - Security: TLS, sRTP, HTTPs
  - Web Configuration Interface
**Product Overview**
- Connects VoIP and FoIP-based applications and services to legacy PBX infrastructures with application feature support
- Designed and tested against wide range of PBX infrastructures
- Designed for ease of deployment and operation

**Product Specification**
- 1U Rack Mount Appliance (~19”x14”)
- Software Selectable T1/E1 Interfaces
  - Single, Dual, Quad Density SKUs
  - NI2, T1 CAS, T1 and E1 Q.SIG
  - 5ESS, DMIS100 and Euro ISDN
- VoIP Protocol Support
  - SIP (RFC 3261)
  - G.711, G.723.1
  - T.38 Fax over IP
- Serial Protocols: SDI, MD1, MD-110
- G.168 Echo Cancellation
- Security: TLS, sRTP, HTTPS
- Web Configuration Interface

**VoIP Protocol Support**
- SIP (RFC 3261)
- G.711, G.723.1
- T.38 Fax over IP
- Serial Protocols: SDI, MD1, MD-110
- G.168 Echo Cancellation
- Security: TLS, sRTP, HTTPS
- Web Configuration Interface

**DMG2000 Series**
- DMG2030DTIQ
- DMG2060DTIQ
- DMG2120DTIQ
- DMG2060DTIS
- DMG2120DTIS

**TDM Configuration**
- Verify or change line settings and framing
- Set PCM Coding as needed
VoIP Configuration

- Set Audio Compression as needed
- Set Voice Activity Detection to Off
- Set Fax Transport Mode to G.711 Passthrough

Routing Table for TDM

- Set interface and channel range
Routing Table for VoIP

Set VoIP Host and IP address

CONFIG.INI

Dialogic® 1000/2000 Media Gateway Series – Logging

- http://www.dialogic.com/support/helpweb/mg/tn888.aspx

- Logging ‘primer’ for 1000/2000 gateway series
- Include ‘config.ini’ file when submitting logs
  – Gives total config of system, including software release
Dialogic® 4000 Media Gateway Series (DMG4000)

- Product Overview
  - Connects Microsoft® Unified Communications (UC) to legacy PBX and PSTN infrastructures with application feature support
  - Also compatible with other SIP applications & services
  - Qualified for use with Microsoft® Office Communications Server 2007
  - Runs Microsoft® UC Mediation Server Software (included)
  - Designed and tested against wide range of PBX infrastructures
  - Designed for ease of deployment and operation

- Product Specification
  - 1U Rack Mount Integrated System (17w" x 20d"")
  - 2 & 4 T1/E1 Port SKUs
  - Provides 48/60 & 96/120 channel gateway
  - Q.SIG Protocol
  - ISDN Protocols
  - ETS, N0, 4ESS, 5ESS
  - CAS Protocols
  - TUCAS, E1/2TNI

- VoIP Protocol Support
  - SIP (RFC 3261), RTP G.711
  - T.38 Fax over IP (Q1/2008)
  - G.168 Echo Cancellation
  - T/L2, aTFTP, RTAudio via Microsoft® UC Mediation Server
  - Web Configuration Interface

Checking system release version of Dialogic® Diva® software suite

Checking version of Dialogic® Diva® SIPcontrol™ Software
Checking Dialogic® Diva® Media Board Licensing

Troubleshooting Dialogic® 3000 and 4000 Media Gateway Series

Dialogic® DMG Media Gateway Series

- DMG3000:
  - 4/8 port BRI
  - EOL’d from production
Dialogic® Media Gateway Series (DMG) – Technical Training

- Installation and Configuration Training Webinars
  - 1 hour per DMG
  - Enables attendees to effectively install DMG

- Advanced Level 1 Training (classroom)
  - 1 day
  - Enables attendees to effectively install DMG in both
    a Microsoft® Office Communications Server 2007 &
    Microsoft® Exchange Server 2007 environment as well as perform
    basic troubleshooting

- Advanced Level 2 Training (classroom)
  - Tuition-Based
  - 1 day per DMG
  - Provides experienced attendees with the tools and methodologies to effectively
    troubleshoot and other products via hands-on training exercises

Training Schedule: [http://www.dialogic.com/training/calendar.htm](http://www.dialogic.com/training/calendar.htm)

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Interworking with Dialogic® IMG 1010 Integrated Media Gateway

- IMG1010: "higher density" gateway
  - SIP, H.323
  - Release 10.3.2 or later

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Dialogic® IMG 1010 Integrated Media Gateway

- IMG1010: "higher density" gateway
  - SIP, H.323
  - Release 10.3.2 or later
Interworking with Dialogic® IMG 1010 Integrated Media Gateway

- An Integrated Media Gateway solution
  - Integrated TDM and IP with interworking
- Signaling and Bearer Capabilities
  - SS7 ISUP (ANSI and ITU base variants)
  - ISDN
  - SIP
  - H.323 v2
- Integrated Routing
  - ANI/DNIS routing
  - Incoming Channel Group
  - Nature of Address
  - Time of Day

Dialogic® IMG 1010 Integrated Media Gateway - Feature Set

- SIP UAC/UAS, H.323
- SS7/ C7 ISUP, ISDN PRI
- T.38 fax support
- Modem support
- T1, DS3 and E1 interfaces
- Works with GK, w/o GK, with SBC*
- GUI-based GC EMS
- RADIUS

*for a list of Dialogic® IMG 1010 Integrated Media Gateway interoperability, refer to:

ClientView Main Window
Dialogic® IMG 1010 Integrated Media Gateway

- Technical Support Contact Info
  - http://www.dialogic.com/support/helpweb/ex/
- Onsite Training & Support – available
- Additional Web Pages, White Papers

Interworking with Quintum

Quintum

- Quintum Tenor Series
  - SIP, H.323
  - Quintum System Software Release P104-12-10 or later with SIP module 2.1.0
  - Tenor DX 2024: H.323 with gatekeeper and SIP
  - Tenor DX 8192: H.323 direct
  - Besides digital (T1/E1) gateways, also analog gateways exist like the analog AF series (e.g., AFG200): should work too but not tested
Interworking with Nortel

- Interop-tested with CS1000 using SIP
  - Nortel release 5.5 or later
  - Interop using H.323 on roadmap

- Interop tests with Nortel BCM are planned

Interworking with Alcatel
Interworking with Alcatel

- OmniPCX Enterprise (OXE) – big system
  - Tested H.323/T.38 with R7.1 plus patches
  - Tested:
    - Without gatekeeper
    - With internal Alcatel gatekeeper
    - With external gatekeeper like Cisco gatekeeper
  - OXE does SIP/T.38 since R7.1 but this has not been tested with the Dialogic® Brooktrout® SR140 Fax Software
  - Same T.38 implementation as for H.323 on OXE
  - Brooktrout® SR140 Fax Software would talk to the CallServer (“the call manager”) directly, which manages SIP, proxy,…
  - Brooktrout talks to the GD or INTIP board (“the gateway”) in the OXE, that manages the H.323/T.38 protocol
- OmniPCX Office – small system – does T.38 but different software and has not been tested
- Dialogic® Brooktrout® SR140/ Dialogic® Brooktrout® TR1034 Fax Board IP H.323 T.38
  - Requires minimally Dialogic® Brooktrout® SDK 5.0 (boston.sys 5.2) plus patch
- See Alcatel Applications Partner Program Interworking Report

Interworking with Alcatel

- Features:
  - H.323/T.38
  - Max V. 29 9600 baud rate on OXE
  - No ECM on OXE
  - Requires G.711 codec configured in OXE
  - Requires default Faststart configured in OXE (when H.245 FastStart is turned on, also H.245 Tunneling is turned on the OXE)
Interworking with Alcatel

- Configuration Example without gatekeeper:
  - GD board with IP address: 172.25.33.41
  - Dialogic® Brooktrout® SR140 Fax Server’s IP address: 172.25.33.93
  - Net mask: 255.255.254.0
  - Default gateway IP address (for routing in IP network): 172.25.32.1
  - # analog fax device plugged into OXE: 102
  - Dialogic® Brooktrout® SR140 Fax # configured in “Direct Speed Dial/Abbreviation No Prefix” (~“voip dial-peer”) in OXE: 10107

Interworking with Siemens

- Interop-tested with HiPath 8000 using SIP
  - Release V3.0 R2 PS19.E05 or later

- Interop-tested with RG8702 Media gateway (SIP)
Interworking with AudioCodes

- Interop-tested with Mediant 1000 using SIP
  - Software Version: 5 or later

- Be sure to configure Mediant for SIP over UDP

Interworking with “SIP trunking”
SIP Trunk

- Service providers are replacing ISDN E1/T1 trunks with SIP trunks; voice-over-Internet
- For list of SIP trunk providers we have completed interoperability:
- Different providers utilize the SIP protocol in different ways
  - That part of the SIP protocol needs to mature and standardize more
- Ask if SIP trunk is used out to network in a given environment, instead of traditional T1/E1 lines going into media gateways