AT&T IP Flexible Reach service on MIS, MPLS PNT and AT&T VPN: Connecting Cisco Business Edition 3000 Release 8.6.4 with Cisco Unified Border Element-Lite Release 8.8 using SIP
October 4, 2012 - Revision 2

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Introduction

Service Providers today, such as AT&T, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP Flexible Reach is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Business Edition 3000 (BE3000) for connectivity to AT&T’s IP Flexible Reach SIP trunk service. The deployment model covered in this application note is customer premise equipment (BE3000, CISCO UBE Lite) to PSTN (via AT&T IP Flexible Reach service). AT&T IP Flexible Reach provides inbound and outbound call service.

- Testing was performed in accordance to AT&T’s IP Flexible Reach test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, DNIS translations, CODEC negotiation, advanced 8YY call prompter, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail, auto-attendant, fax using T.38 and G.711 (G3 and SG3 speeds), and teleconferencing.

- For IP Flexible Reach service with MIS/MPLS PNT (and optionally with AT&T VPN) access, the Cisco Unified Border Element IP address (facing the Customer Edge Router) can be private IP address. This will be NATed by the AT&T managed Customer Edge Router (or customer managed/MRS managed Customer Edge Router for AT&T VPN). Consult with AT&T provisioning engineer to resolve any IP addressing issues.

- The Cisco BE3000 and CUBE Lite configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration parameters for successful interoperability. Care must be taken by the network administrator deploying Cisco BE3000 and CUBE Lite to ensure these configuration parameters are set to interoperate to AT&T SIP network.

- Please refer to the Emergency 911/E911 Limitations and Restrictions section of this document for more information on Emergency 911/E911 services.

- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communications Manager with Cisco Unified Border Element components.

- This Application Note uses the Cisco 881 Integrated Services Router to run the CISCO UBE feature set; however other Cisco voice gateways are also an option to use since CISCO UBE implementations do not depend on the platform. Below is a list of Cisco platforms capable of CISCO UBE functionality, although not all of them may support CISCO UBE version 8.8.

  - Cisco 3900 Series Integrated Services Routers
  - Cisco 2900 Series Integrated Services Routers
  - Cisco 2800 Series Integrated Services Routers
  - Cisco 3800 Series Integrated Services Routers
  - Cisco AS5350XM Universal Gateway
  - Cisco AS5400XM Universal Gateway
  - Cisco 1861 Integrated Services Router
  - Cisco 881 Integrated Services Router
  - Cisco 888 Integrated Services Router
  - Cisco IAD880 Series Integrated Access Devices
  - Cisco IAD2430 Integrated Access Device
Network Topology

Figure 1. Basic Call Setup

Note: The Cisco Unified Border Element depicted in Figure 1 is not an AT&T managed device. It is recommended that the group responsible for the administration, management and configuration of the Cisco Unified Communications Manager, also manage and configure the Cisco Unified Border Element.

For Cisco supported deployment strategies using CISCO UBE, refer to the Cisco Unified Communications SRND Based on Cisco Unified Communications Manager 8.X:

System Components

**Hardware Components**

- Cisco Business Edition 3000
- Cisco SPA8800 IP Telephony Gateway (needed whenever additional FXS ports are required)
- Cisco Catalyst 3560 switch with PoE
- Cisco IP Phones (CP-6900 and CP-8900 Series sets)
- Cisco 881 Series Integrated Services Access Device

**Software Requirements**

- Cisco Business Edition 3000 software version 8.6.4.10000-15 and future maintenance releases
- Cisco IOS version 15.2(1)T2. This configuration was validated using software version c880voice-universalk9-mz.152-1.T2.bin; however, this document is applicable to all software versions 15.2(1)T/CUBE 8.8 and 15.2(2)T/CUBE 8.9 software releases
- The documented CISCO UBE configuration can be supported with the following IOS feature sets: UNIVERSAL
- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communications components. For reference, please follow this link: [http://www.cisco.com/en/US/prod/collateral/voicesw/ps6790/gatecont/ps5640/order_guide_c07_462222.html](http://www.cisco.com/en/US/prod/collateral/voicesw/ps6790/gatecont/ps5640/order_guide_c07_462222.html)
- SPA8800 software version 6.1.9
Features

This section lists supported and unsupported features.

Features Supported

- Basic Call using G.729a / G.711mu
- Calling Party Number Presentation and Restriction
- Calling Name
- AT&T Advanced 8YY Call Prompter (8YY)
- Intra-Site Call Transfer
- Intra-Site Call Conference
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- AT&T IP Teleconferencing
- Fax over G.711
- Incoming DNIS Translation and Routing
- Outbound calls to AT&T’s IP and TDM networks
- CPE voicemail managed service, leave and retrieve voice messages via incoming AT&T SIP trunk (BE3000 Voicemail)
- Auto-attendant transfer-to service (BE3000 Automated Attendant)

Features Not Supported

- T.38 Fax Relay
Caveats

- Fax Pass-through: Upspeed to G.711 is not supported with Cisco Business Edition 3000. Fax calls must therefore be established using G.711. Since the Cisco BE3000 does not support device/region-based codec preference configuration, all calls from/to fax endpoints over SIP trunk facilities must be set to use G.711. This can be accomplished by dialing a unique outside dial access code for outbound calls, matching a dedicated dial-peer configured to use G.711 as the only codec. Similarly, inbound calls to fax endpoints must match dial-peer(s) based on the called number, configured to provide G.711 as the only codec to the Cisco BE3000.

- T.38 Fax relay: during testing, fax calls between fax endpoints configured for SG3 transmission were unreliable.

- SPA8800: During testing, some calls over from analog devices connected to SPA8800 gateway would not complete. The issue is caused by the default value of advanced SIP parameter “Max Forwards” being set too low (12). Refer to CSCtw50178 for further information.

- Call forwarding to 800-numbers: In order for the AT&T network to accept calls forwarding to 800-numbers, sending of Diversion header containing a valid customer telephone number is required.

Configuration considerations

- For outbound calls, AT&T SIP network will always select the first choice codec presented in the initial SIP INVITE (unless the end-device does not support the listed preferred codec), and processes calls accordingly. The codec preference order is defined in CISCO UBE (via Voice Class Codec configuration). For inbound calls, Cisco BE3000 when configured for “Most Calls” (Connections ➔ Sites ➔ CentralSite, Call Quality tab, parameter “Audio Quality/Call Quantity Tradeoff”), will use G.729 as the supported codec. When configured for “Best Quality”, Cisco BE3000 will use G.711 for inbound calls. In order to allow the Cisco BE3000 to process inbound calls using G.729 (e.g. for calls to subscribers stations), as well as G.711 (e.g. for calls to fax endpoints), configure parameter “Audio Quality/Call Quantity Tradeoff” for “Best Quality”; then configure dial-peer(s) used for inbound calls to subscribers stations to provide G.729 as the only codec. Additional dial-peer(s) for inbound calls to fax endpoints must also be configured on CISCO UBE, matching the fax endpoint(s) telephone number(s), and providing G.711 as the only available codec. Outbound calls requiring G.711 (e.g.calls from fax endpoints) must be prefixed by a dedicated outside dial access code (in this document, digit “8” was used), matching a separate outbound dial-peer configured to “strip” the outside dial access code and use G.711 as the only available codec.

- Some Cisco IP phones models, such as the 6900-series IP phones, use packetization time of 20 ms. AT&T recommends using 30 ms for optimal bandwidth utilization. Changing packetization time can be achieved by configuring a sip-profile in CISCO UBE.

Emergency 911/E911 Services limitations and restrictions

Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.

While AT&T IP Flexible Reach supports E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found in the SG Library at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.
Configuration

Configuring the Business Edition 3000 (Cisco BE3000)

Cisco BE3000 Software Version

![Cisco Business Edition 3000 Administrative Interface]

**Maintenance > Installed Software**

**Installed Software**

**System Software**

- Active Version: 8.6.4.10000-15
- Inactive Version: 8.6.3.10000-12

**Optional Software Packages**

Listed below are the optional software packages installed on the system. Use the Maintenance > Upgrade page to install or upgrade optional software package files.

<table>
<thead>
<tr>
<th>Name</th>
<th>Installation Date</th>
</tr>
</thead>
<tbody>
<tr>
<td>ciscocm_patch.cop</td>
<td>2013-Mar-05, 15:48:40 PST</td>
</tr>
<tr>
<td>ciscocm_post1.cop</td>
<td>2012-Mar-15, 09:51:36 PDT</td>
</tr>
<tr>
<td>cm-comp-CP-8.5.2-SIPTrunkConnectionATT-3.cop</td>
<td>2011-Sep-13, 10:12:15 PDT</td>
</tr>
<tr>
<td>cm-comp-CP-8.5.2-SIPTrunkConnectionATT-4.cop</td>
<td>2011-Oct-24, 13:50:49 PDT</td>
</tr>
</tbody>
</table>
Configuring Cisco BE3000 Dial Plan

Note: In this configuration page, you can define the customer’s main telephone number, which will be used as the external caller ID on outbound calls over the SIP trunk, whenever parameter “External Caller ID” within the User settings page is left blank. This configuration page is also used to define local extensions’ length and Voicemail/Auto Attendant pilot number. Also, note that two Outside Dial Codes are configured: regular calls (using G.729 codec) will be placed dialing “9” as the outside dial access code. Fax calls (using G.711 codec) will require dialing “8” as the outside dial access code. The leading “8” will not be stripped by the Cisco BE3000, and will be matched against a specific dial-peer on CISCO UBE that will provide G.711 as the only available codec towards the AT&T network.
Configuring Cisco BE3000 Hub/Central Site

Note: Using the General settings tab, define the Local Area Code(s), as well as the internal network subnet(s) used by the system. Please note that the Cisco BE3000 will “strip” the Area Code whenever dialing local Area Code numbers (both 1+10 digit and 10-digit dialing), outpulsing the 7-digit local number. 7-digit dialing over SIP trunk is not supported. If 10-digit local dialing is required, more than one Area Code must be defined in the Local Area Codes field. When multiple Area Codes are configured, the Cisco BE3000 will outpulse 10 digits over PSTN facilities whenever 10-digit local Area Code numbers are dialed. 10-digit dialing to numbers outside defined Local Area Codes is not supported by the Cisco BE3000.
Note: Using the Call Settings tab, define the trunk(s) used for PSTN access. This configuration page is also used to define the highest call system-wide privileges allowed by the Cisco BE3000. Highest level of calls allowed can be further restricted using Usage Profile settings.
Note: Using the Call Quality tab, configure the Cisco BE3000 to support both G.729 and G.711 calls. This is done by selecting “Best Quality” using the slider for parameter “Audio Quality/Call Quantity Tradeoff”.
Configuring Cisco BE3000 PSTN Connections

Note: the T1 PRI port configuration on the Cisco BE3000 internal-gateway is required, as the system stores the DSP configuration on the internal-gateway configuration file. Without this configuration, no DSP resources will be available for conferencing, etc.
Configuring SIP Trunk to AT&T (via CUBE Lite)
Note: In order to allow calls to properly forward over SIP trunk, enable “Send Redirecting Diversion Header” and “Accept Incoming Redirecting Diversion Header”. Also, configure DTMF signaling for in-band (RFC 2833).
Outbound Call Routing: In order to allow the Cisco BE3000 to pass the “8” outside dial access code to CUBE, configure this parameter to “Set Manually”, and then add/edit the Call Number Pattern starting with “8” so as not to remove the first digit. Example is provided below:
Configuring Cisco BE3000 Devices (SPA8800 IP telephony gateway)

Note: When configuring the SPA8800 to support fax equipment, ensure that “Fax Mode” is set to “G.711 Passthrough”, as it is the most reliable fax relay method offered by the SPA8800 analog gateway.
Configuring SPA8800 Advanced Voice Settings

Note: The SPA8800 Advanced Voice settings page is accessible via Web Browser, using the following URL: http://IP_Address_Of_SPA/admin/voice/advanced During first-time installation of the SPA8800, configure parameter “Profile Rule” as “ftp://BE3000_ipaddress/spa$MA.cnf.xml”. In the example above, 172.20.8.67 is the IP Address assigned to the Cisco BE3000. This parameter is found in the “Provision” tab.
Note: During testing, it was noticed that the default value of SPA advanced SIP parameter “Max Forwards” may be too low when placing outbound calls. Increase the value accordingly. Parameter “Max Forwards” is found in the “SIP” tab.
Configuring Cisco BE3000 Usage Profile

Note: Usage Profile allows to set the highest level of calls allowed to be assigned to users, as well as system forwarding for busy/no answer calls.
Configuring Cisco BE3000 Users
## Configuring Cisco BE3000 Phones

### Phones

<table>
<thead>
<tr>
<th>Name</th>
<th>Owner</th>
<th>Extension</th>
<th>Description</th>
<th>Model</th>
<th>Actions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SPA28CBA2FDE001</td>
<td>fauline</td>
<td>2712</td>
<td>Fan Machine A</td>
<td>Analog Phone (SPA8960)</td>
<td>Edit</td>
</tr>
<tr>
<td>SPA28CBA2FDE003</td>
<td>farbwo</td>
<td>2713</td>
<td>Fan Machine B</td>
<td>Analog Phone (SPA8860)</td>
<td>Edit</td>
</tr>
<tr>
<td>SEPc9c1da3d421</td>
<td>joes</td>
<td>2714</td>
<td>Joe Doe</td>
<td>Cisco 8941</td>
<td>Edit</td>
</tr>
<tr>
<td>SEPc9c1da3651f</td>
<td>daddams</td>
<td>2715</td>
<td>Dilbert Adams</td>
<td>Cisco 8961</td>
<td>Edit</td>
</tr>
<tr>
<td>SEP5982f9725cas</td>
<td>dmenace</td>
<td>2716</td>
<td>Dennis The Menace</td>
<td>Cisco 8921</td>
<td>Edit</td>
</tr>
<tr>
<td>SEP6400f13aca84</td>
<td>jerry</td>
<td>2717</td>
<td>Tom N Jerry</td>
<td>Cisco 8921</td>
<td>Edit</td>
</tr>
<tr>
<td>SEPc9c1da3aceb6</td>
<td>deploarer</td>
<td>2718</td>
<td>Dora The Explorer</td>
<td>Cisco 8941</td>
<td>Edit</td>
</tr>
</tbody>
</table>
Configuring CISCO UBE Lite

CUBE-Lite-1#sho version
Cisco IOS Software, C880 Software (C880VOICE-UNIVERSALK9-M), Version 15.2(1)T2,
RELEASE SOFTWARE (fc2)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2012 by Cisco Systems, Inc.
Compiled Sun 19-Feb-12 09:20 by prod_rel_team

ROM: System Bootstrap, Version 12.4(22r)YB5, RELEASE SOFTWARE (fc1)

CUBE-Lite-1 uptime is 22 hours, 13 minutes
System returned to ROM by reload at 19:15:17 UTC Wed Mar 7 2012
System restarted at 19:16:01 UTC Wed Mar 7 2012
System image file is "flash:c880voice-universalk9-mz.152-1.T2.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:

If you require further assistance please contact us by sending email to export@cisco.com.

Cisco 881 (MPC8300) processor (revision 1.0) with 747520K/38912K bytes of memory

Processor board ID FTX152300BD

5 FastEthernet interfaces
1 Virtual Private Network (VPN) Module
256K bytes of non-volatile configuration memory.
125440K bytes of ATA CompactFlash (Read/Write)

License Info:

License UDI:

Device#   PID SN
--------- -------
*0  CISCO881-K9 FTX152300BD

License Information for 'c880-iad'
License Level: advipservices Type: Permanent
Next reboot license Level: advipservices

Configuration register is 0x2102

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UBE-Lite-1#show running-config
Building configuration...

Current configuration : 9485 bytes
!
! Last configuration change at 17:07:51 UTC Tue Nov 15 2011 by cisco
! NVRAM config last updated at 17:07:52 UTC Tue Nov 15 2011 by cisco
! NVRAM config last updated at 17:07:52 UTC Tue Nov 15 2011 by cisco
version 15.2
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CUBE-Lite-1
!
boot-start-marker
boot system flash:c880voice-universalk9-mz.152-1.T2.bin
boot-end-marker
!
logging buffered 9999999
enable secret 5 $1$JpiT$oYedxRlWfNYecapKBz9YN/
!
no aaa new-model
!
!
ip dhcp excluded-address 10.10.10.1
!
!
no ip domain lookup
ip domain name yourdomain.com
ip cef
no ipv6 cef
!
!
voice service voip
ip address trusted list
ipv4 207.xxx.xxx.xxx
ipv4 135.xxx.xxx.xxx
no ip address trusted authenticate
address-hiding
allow-connections sip to sip
redirect ip2ip
fax protocol pass-through g711ulaw
!
sip
header-passing
error-passthrhu
early-offer forced
asserted-id pai
!
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all

1 This command enables router to perform G.711 fx pass-through.
2 This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai)
no call service stop
!
!
**voice class codec** 1
codec preference 1 g729r8
codec preference 2 g711ulaw
!
**voice class sip-profiles** 1
request INVITE sdp-header Audio-Attribute add "a=ptime:30"
response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
!
**voice class sip-profiles** 2
request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
!
**voice translation-rule** 1
rule 1 /81/ /11/
! voice translation-profile outbound_fax_g711 translate called 1
! interface FastEthernet0
  switchport access vlan 110
  no ip address
! interface FastEthernet1
  no ip address
! interface FastEthernet2
  no ip address
! interface FastEthernet3
  no ip address
! interface FastEthernet4
  ip address 99.xxx.xxx.xxx 255.255.255.0
duplex auto
speed auto
!
! interface Vlan110
  ip address 172.20.xxx.xxx 255.255.255.0
  !
ip default-gateway 99.xxx.xxx.xxx
ip forward-protocol nd
ip http server
ip http access-class 23
ip http authentication local

---

3 This command configures the codec preference to be assigned to dial-peers. Alternatively, single codec's can be configured into individual dial-peers.
4 By default, Cisco 6900-series IP phones use ptime value of 20 ms. AT&T networks prefer ptime value of 30 ms. This SIP profile modifies SDP ptime value from 20 to 30.
5 As previously stated, AT&T networks prefer ptime value of 30 ms. This SIP profile modifies SDP ptime value from 20 to 30 for inbound/outbound G.711 calls.
6 This command allows CUBE to strip the leading “8” from the dialed string used for outbound fax calls using G.711 codec. This voice translation rule is assigned to voice translation profile “outbound_fax_g711”, and is then configured in the dial-peer used for outbound G.711 fax calls.
ip http secure-server
!
ip route 99.xxx.xxx.xxx 255.255.255.255 FastEthernet4
ip route 135.xxx.xxx.xxx 255.255.255.0 FastEthernet4
ip route 172.20.0.0 255.255.0.0 172.20.110.1
ip route 207.xxx.xxx.xxx 255.255.255.255 99.xxx.xxx.xxx
!
control-plane
!
dial-peer voice 2051 voip
description "Int'l calls to AT&T - AT&T facing side"
destination-pattern 011T
session protocol sipv2
session target ipv4:207.xxx.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2050 voip
description "Int'l calls to AT&T - IP-PBX facing side"
session protocol sipv2
incoming called-number 011T
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2041 voip
description "N11 Calls to AT&T - AT&T facing side"
destination-pattern .11
session protocol sipv2
session target ipv4:207.xxx.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2040 voip
description "N11 Calls to AT&T - IP-PBX facing side"
session protocol sipv2
incoming called-number .11
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2000 voip
description "Outgoing calls to AT&T - IP-PBX facing side"
session protocol sipv2
incoming called-number .T

7 This command allows CISCO UBE to process DTMF Payload Types different than the default value (101)
8 This command enables DTMF digit passing using RTP NTE (RFC2833) to calls matching this dial-peer
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2001 voip
description "Outgoing to AT&T - AT&T facing side"
destination-pattern 1T
session protocol sipv2
session target ipv4:207.xxx.xxx.xxx
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2011 voip
description "Incoming calls to IP-PBX - IP-PBX facing side"
destination-pattern [37][13][24]........
session protocol sipv2
session target ipv4:172.20.xxx.xxx
voice-class sip profiles 1
voice-class sip asymmetric payload full
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2010 voip
description "Incoming calls to IP-PBX - AT&T facing side"
session protocol sipv2
incoming called-number [37][13][24]........
voice-class sip profiles 1
voice-class sip asymmetric payload full
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2100 voip
description "Outgoing G.711 fax calls to AT&T - AT&T facing side"
translation-profile outgoing outbound_fax_g711
destination-pattern 8T
session protocol sipv2
session target ipv4:207.xxx.xxx.xxx
voice-class sip early-offer forced
voice-class sip profiles 2
voice-class sip asymmetric payload full
dtmf-relay rtp-nte
codec g711ulaw
fax-relay sg3-to-g3
fax rate 14400
fax protocol pass-through g711ulaw
!
dial-peer voice 2101 voip
description "Incoming G.711 fax calls - IP-PBX facing side"
destination-pattern 7322162714
session protocol sipv2
session target ipv4:172.20.xxx.xxx

When configuring dial-peers used for inbound G.711 calls to fax endpoint(s) residing on the Cisco BE3000, configure the fax endpoint telephone number as the destination-pattern.
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 2
voice-class sip asymmetric payload full
dtmf-relay rtp-nte
codec g711ulaw
fax-relay sg3-to-g3
fax rate 14400
fax protocol pass-through g711ulaw
!
dial-peer voice 2102 voip
description "Incoming G.711 fax calls - AT&T facing side"
session protocol sipv2
incoming called-number 7322162714\(^{10}\)
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 2
voice-class sip asymmetric payload full
dtmf-relay rtp-nte
codec g711ulaw
fax-relay sg3-to-g3
fax rate 14400
fax protocol pass-through g711ulaw
!
!
num-exp 2162714 7322162714\(^{11}\)
num-exp 2162715 7322162715
num-exp 2162716 7322162716
!
sip-ua
no remote-party-id
retry invite 2
!
!
line con 0
line aux 0
line vty 0 4
exec-timeout 3000 0
privilege level 15
password cisco
login local
transport input telnet ssh
!
end

\(^{10}\) When configuring dial-peers used for inbound G.711 calls to fax endpoint(s) residing on the Cisco BE3000, configure the fax endpoint telephone number as the incoming called-number.

\(^{11}\) This command expands 7-digit telephone number passed on inbound local calls to full 10-digit DID number. This is necessary in order to allow inbound 7-digit local calls to be relayed to the Cisco BE3000.
### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
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<tbody>
<tr>
<td>CODEC</td>
<td>Coder-Decoder (in this document a device used to digitize and undigitize voice signals)</td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>DSL</td>
<td>Digital Subscriber Line</td>
</tr>
<tr>
<td>DNIS</td>
<td>Dialed Number Identification Service</td>
</tr>
<tr>
<td>DSP</td>
<td>Digital Signal Processor</td>
</tr>
<tr>
<td>IP</td>
<td>Internet Protocol</td>
</tr>
<tr>
<td>ISP</td>
<td>Internet Service Provider</td>
</tr>
<tr>
<td>PoE</td>
<td>Power over Ethernet</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SP</td>
<td>Service Provider</td>
</tr>
<tr>
<td>TDM</td>
<td>Time-Division Multiplexing</td>
</tr>
</tbody>
</table>
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