Dual-Stack Cisco Unified Border Element
Customer Configuration Guide for use with
AT&T IP Flexible Reach Service on MIS, MPLS PNT
or AT&T VPN as the Underlying Transport
and
Microsoft® Lync™ Server 2010
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Note: Testing was conducted at AT&T labs.
Introduction

This application note describes how to configure a dual-stack Cisco Unified Border Element (UBE) to provide voice over IP (VoIP) connectivity between a IPv4 enabled Microsoft Lync Server 2010 environment and the IPv6 enabled AT&T IP Flexible Reach Service.

AT&T IP Flexible Reach, on MIS, MPLS PNT or AT&T VPN as the Underlying Transport, is an AT&T offering that allows calls from a user managed VoIP network to the PSTN and offers an end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines.

Laboratory testing was performed for the preparation of this guide. Key features verified are described in Features Supported section below.

The CISCO UBE configuration detailed in this document is based on a dual-stack lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Microsoft Lync Server 2010. The configuration described in this document details the important commands to enable for interoperability to be successful. Care must be taken, by the network administrator deploying CISCO UBE to ensure these commands are set per each dial-peer required to interoperate with the AT&T IP Flexible Reach Service.

This guide does not cover configuration of the Microsoft Lync Server 2010 environment.

This Application Note uses the Cisco 2921 Integrated Services Router (ISR) G2 to run CISCO UBE feature set however other Cisco voice gateways are also an option to use since CISCO UBE implementation does not depend on the platform. Here is a list of Cisco platforms capable of CISCO UBE functionality: (Note: Not all the routers listed below support CISCO UBE 8.8)

Cisco 3900 Series Integrated Services Routers
Cisco 2900 Series Integrated Services Routers

Network Topology

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Note: Testing was conducted at AT&T labs.
Limitations

These are the known limitations, caveats, or integration issues.

- G.711 is the only supported codec. AT&T IP Flexible Reach Service supports the G.711 codec for bandwidths of 1.544 Mbps (T1) or greater.
- FAX is not supported.
- Compressed RTP (cRTP) is not supported.
- Network Address Translation (NAT) is not supported.
- In order for call transfers and conference calls to work when a Lync Client transfers or conferences an off-net call, the "Enable refer support" option under the Trunk Configuration for the AT&T facing Trunk must be disabled. This option is in the Microsoft Lync Server 2010 Control Panel under: Voice Routing -> Trunk Configuration. Please note that the default configuration when configuring a trunk has this option enabled. This option forces Lync to send CISCO UBE a SIP Refer instead of a Re-Invite, which results in a failed call.
- Calling number privacy is not supported by Lync 2010 on Lync 2010 to AT&T IP Flexible Reach Service calls.
- Call hold, transfer and conference initiated by a Lync 2010 client are not supported on native IP calls to the AT&T IP Teleconferencing Service. The Lync client informs the user that the "call cannot be placed on hold" and closes the voice channel. The user must disconnect the call and reconnect.
- The dial peers in this configuration guide are currently configured to accommodate Lync 2010 / CISCO UBE installations in the US. Additional customization of the dial peers will be required to accommodate non-US sites.
- The configuration and management of the Microsoft servers required for the Lync 2010 IP-PBX environment are the responsibility of the customer.
- For management of the CISCO UBE and CE Router, AT&T supports the following options:
  - Customer Managed CISCO UBE and CE Router.
  - AT&T Managed Router Service (MRS) offers management of the CISCO UBE and/or CE Router (offered for AT&T VPN only).
  - AT&T Managed CE Router with Customer Managed CISCO UBE (offered for MIS/PNT only).
- For outbound calls from Lync 2010 to AT&T IP Flexible Reach Service, Lync 2010 must send “+” followed by country code and number. CISCO UBE will then send all numbers with a leading “+” to the AT&T IP Flexible Reach Service.
  - There is one exception to this rule: For N11 calls, CISCO UBE will remove the “+” otherwise AT&T IP Flexible Reach Service will not process the N11 call. Note that custom configuration is required on Lync 2010 to support N11 dialing.
- For inbound calls, a customer may receive one of 2 types of DID’s from AT&T IP Flexible Reach Service: Virtual TN’s and Non-Virtual TN’s.
  - A Virtual TN is one that has an NPA that is different from the NPA at the customer site to which it is being routed. For a Virtual TN, AT&T will pass 10 digits to the PBX. For example, if a PBX telephone were associated with a Virtual TN, the number received from AT&T would be 10 digits (i.e. 732-216-2700). Dial peers are provided in this guide for adding a “+1” to these types of TN’s.
  - A Non-Virtual TN has an NPA that is the NPA at the customer site. For a Non-Virtual TN, AT&T will pass the length of the phone extension plus some prefix if needed (typically a 4 digit extension without a prefix). If a PBX telephone is associated with a Non-Virtual TN, the number received from AT&T would be 7 digits (i.e. 368-4997 for a 732-368-4997 TN). Dial peers are provided in this guide for adding a “+1” and an NPA to these types of TN’s.
- The Lync 2010 identity should always be +1 followed by the 10 digit TN.

Note: Testing was conducted at AT&T labs.
• Private dialing plans will require dial-peer customization beyond the scope of this document.

• Display name is not supported. Lync 2010 did not pass display name in the tested configuration.

• The following redundancy is supported:
  o Outbound calls from a single CISCO UBE to multiple AT&T IP Flexible Reach Service Border Elements.
  o Lync 2010 to the Mediation server is part of the standard Microsoft configuration.

• Redundancy from the Mediation Server to the CISCO UBE is not currently supported.

• Lync 2010 does not currently support the Diversion header for forwarded calls. A fixed CPN (calling party number) will be added to all Lync 2010 originated calls with the following clarifications. The CPN must be one of the AT&T assigned TN’s for the Lync 2010 site.
  o The Diversion header with the fixed CPN will not be added to the N11 calls. Thus the true CPN from Lync 2010 will be sent to the AT&T IP Flexible Reach network for these calls.
  o Calls forwarded to N11 numbers (except 911) will be blocked by AT&T IP Flexible Reach Service.
  o 911 calls will always be routed by AT&T IP Flexible Reach Service regardless of the presence of CPN.

• Dial peer configuration can be modified at the customer’s request to accommodate special needs.

• 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 Business Voice over IP Services 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 Business Voice over IP (BVoIP) Services Service Guide in detail to understand the limitations and restrictions.

**System Components**

**Hardware Requirements**

The following hardware is required:

• Cisco Integrated Service Router G2. This solution was tested with C2921 but this application note applies to any ISR G2 platforms. Refer to the following links provided in the Introduction section of this document for more information on ISR platforms.

**Software Requirements**

The following software is required:

• CISCO UBE version 8.8
  o Cisco Unified Border Element Release 8.8 with IOS version 15.2.1T2 release. This configuration was tested with C2900-universalk9-mz.SPA.152-1.T2.bin.
  • Microsoft Lync Server 2010 version 4.0.7577.0 (Standard Edition or Enterprise Edition)
  • Microsoft Exchange Server 2010 Service Pack 1
  o RTM version 14.00.0639.021
Features
This section lists supported and unsupported features.

Features Supported

• Basic Call using G.711 including outbound basic calls (Lync to AT&T IP Flexible Reach Service) to N11 endpoints.
• Attended and unattended intra-site transfers
• Intra-site Conference
• Call Hold and Resume
• Call Forward
• AT&T IP Teleconferencing (see caveats)
• Interoperability of IPv4 IP-PBX environment with AT&T IP Flexible Reach Service IPv6 environment using a dual-stack CISCO UBE.

Configuration
This section contains configuration menus and commands and describes configuration sequences and tasks.

Configuring the Cisco Unified Border Element

Cisco IOS version
CUBE#show ver
Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9_NPE-M), Version 15.2(1)T, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Thu 21-Jul-11 18:24 by prod_rel_team

ROM: System Bootstrap, Version 15.0(1r)M1, RELEASE SOFTWARE (fc1)

CUBE uptime is 2 weeks, 5 days, 1 hour, 8 minutes
System returned to ROM by power-on
System restarted at 14:12:50 UTC Fri Feb 10 2012
System image file is "flash0:/c2900-universalk9_npe-mz.SPA.152-1.1.T.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command

Note: Testing was conducted at AT&T labs.
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: http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

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

Cisco CISCO2911/K9 (revision 1.0) with 487424K/36864K bytes of memory.
Processor board ID FHK1334F17T
3 Gigabit Ethernet interfaces
1 terminal line
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
248472K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

<table>
<thead>
<tr>
<th>Device#</th>
<th>PID</th>
<th>SN</th>
</tr>
</thead>
<tbody>
<tr>
<td>*0</td>
<td>CISCO2911/K9</td>
<td>###</td>
</tr>
</tbody>
</table>

Technology Package License Information for Module: 'c2900'

<table>
<thead>
<tr>
<th>Technology Package</th>
<th>Technology-package</th>
<th>Technology-package</th>
<th>Current</th>
<th>Type</th>
<th>Next reboot</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipbase</td>
<td>ipbasek9</td>
<td>Permanent</td>
<td>ipbasek9</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>security</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>uc</td>
<td>uck9</td>
<td>Permanent</td>
<td>uck9</td>
<td></td>
<td></td>
</tr>
<tr>
<td>data</td>
<td>datak9</td>
<td>Permanent</td>
<td>datak9</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Configuration register is 0x2102

Configuring Cisco Unified Border Element (UBE)
The following is an example CISCO UBE configuration with all critical commands marked in bold and with footnotes.

CUBE#show runn
Building configuration...

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Note: Testing was conducted at AT&T labs.
Current configuration: 6639 bytes

! Last configuration change at 14:41:02 UTC Thu Mar 1 2012
! NVRAM config last updated at 20:17:24 UTC Tue Feb 14 2012
! NVRAM config last updated at 20:17:24 UTC Tue Feb 14 2012
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CUBE
!
boot-start-marker
boot system flash0:/c2900-universalk9-npe-mz.SPA.152-1.T.bin
boot-end-marker
!
no aaa new-model
!
ipv6 unicast-routing
ipv6 cef
!
!
ip cef
multilink bundle-name authenticated
!
!
!
!
!
!
!
!
!
!
crypto pki token default removal timeout 0
!
!
voice-card 0
!
!
!
!
voice service voip
no ip address trusted authenticate
address-hiding
allow-connections sip to sip ¹
redirect ip2ip
sip
re1xx disable ²
header-passing
error-passthru ³
asserted-id pai
privacy pstn
no update-callerid
midecall-signaling passthru ⁴

¹ This command enables the basic IP-to-IP CISCO UBE feature for SIP calls.
² Prack Processing is turned off to simplify the call flow.
³ This command allows for SIP error messages to pass-through end-to-end without modification through CISCO UBE.

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Note: Testing was conducted at AT&T labs.
privacy-policy passthru

voice class codec 1
codec preference 1 g711ulaw

voice class sip-profiles 2
request INVITE sip-header Diversion add "Diversion: <sip:7323204097@2 001:1818:15:22::2>" 6
request INVITE sip-header From modify "(<sip:[A-Za-z].)@(.*))" "<sip:7323204097@2" 7

voice iec syslog

voice statistics type iec

voice translation-rule 1111
  rule 1 /^\((.*)/)\+1&/ 8

voice translation-rule 3333
  rule 1 /^\+(.*)\+11\/ \+1/ 9

voice translation-rule 4444
  rule 1 /^368\((.*)/)\+1732&/ 10

voice translation-profile AddPlusOne 11
  translate called 1111

voice translation-profile RemovePlusN11 12
  translate called 3333

voice translation-profile AddNPA 13
  translate called 4444

interface GigabitEthernet0/0
description Facing Lync2010 Environment
ip address 10.60.60.20 255.255.255.0
duplex auto

4 This command must be enabled at a global level to maintain integrity of SIP signaling across SIP end-points.
5 G.711 is the supported CODEC.
6 This command adds a Diversion header with a fixed calling party number (CPN) so that calls can be forwarded by to 8YY and NPA5551212 endpoints if needed. The CPN (7323204097 is an example) must be one of the AT&T assigned TNs for this Lync site. The right hand side (2001:1818:15:22::2) must be the IP address of the AT&T facing side of the CISCO UBE.
7 If the user part of the From header contains a string that begins with an alphabetic character (e.g. Microsoft Lync user name) instead of an E.164 number, this command replaces the user part with a CPN. The CPN (7323204097 is an example) must be one of the AT&T assigned TNs for this Lync site. This scenario will occur if an Lync user is conferencing in a 3rd party on the AT&T IP Flexible Reach Network.
8 This rule adds a “+1” to a called number.
9 This rule removes a “+” from a called number of the form +N11.
10 This rule adds +1732 to a 7 digit number that starts with “368”. This type of rule is required for non virtual TNs for which AT&T IP Flexible Reach sends a maximum of 7 digits to the customer premises.
11 Translation Profile for adding a “+1”
12 Translation Profile for removing a “+” from a N11 number
13 Translation Profile for adding “+1NPA” to a 7 digit number.
speed auto
!
interface GigabitEthernet0/1
description Facing AT&T Customer Edge Router
no ip address
duplex auto
speed 100
ipv6 address 2001:1818:15:22::2/64
!
!
dial-peer voice 2000 voip
description "Incoming – AT&T to CUBE match on [2-9]........"
session protocol sipv2
incoming called-number [2-9]........ 14
voice-class codec 1
voice-class sip asserted-id pai
dtmf-relay rtp-nte
fax-relay sg3-to-g3 15
fax rate 14400
no vad 16
!
dial-peer voice 2001 voip 17
description "Incoming – CUBE to Lync match on [2-9]........ – virtual TNs"
translation-profile outgoing AddPlusOne 18
destination-pattern [2-9]........
session protocol sipv2
session target ipv4:10.60.60.15 19
session transport tcp 20
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip block 183 sdp absent
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
!
dial-peer voice 2002 voip 21
description "Incoming - CUBE to Lync match on 368.... – non virtual TN"
translation-profile outgoing AddNPA
destination-pattern 368....
session protocol sipv2
session target ipv4:10.60.60.15
session transport tcp
voice-class codec 1

14 Match on a called number starting with 2 through 9.
15 Fax is not supported but this command and the following command are here for potential future use.
16 Disables voice activity detection (VAD).
17 Dial peer for matching virtual TN’s for which AT&T IP Flexible Reach sends 10 digits to the customer premises.
18 Add a “+1” to the called number before sending to Lync.
19 IP Address of the Microsoft Lync Mediation Server. Any port other than the default of 5060 must be specified in the format of “session target ipv4:<IP Address>:<Port>” (eg. session target ipv4:10.60.60.15:5068)
20 Support of TCP on calls to Lync.
21 Dial peer for matching for non virtual TNs for which AT&T IP Flexible Reach sends a maximum of 7 digits to the customer premises.

Note: Testing was conducted at AT&T labs.
voice-class sip asserted-id pai
voice-class sip block 183 sdp absent 22
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
!
dial-peer voice 1003 voip
description "Outgoing – CUBE to AT&T match on +[1-9]11"
translation-profile outgoing RemovePlusN11
destination-pattern +[1-9]11
session protocol sipv2
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip early-offer forced
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
!
dial-peer voice 1001 voip
description "Outgoing – Lync to CUBE matching on +T"
session protocol sipv2
session transport tcp
incoming called-number +T
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip early-offer forced
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
no vad
!
dial-peer voice 1000 voip
description "Outgoing – CUBE to AT&T matching on +T"
destination-pattern +T
session protocol sipv2
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip early-offer forced
voice-class sip profiles 2 23
no voice-class sip block 183
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
no vad
!

Remove 183 messages without SDP from Lync to eliminate ringback problems on certain PSTN originated calls.
Add diversion header with fixed Calling Party Number (CPN).

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Note: Testing was conducted at AT&T labs.
protocol mode dual-stack
!
end

24 CISCO UBE will attempt 2 invites to a dial peer (e.g. AT&T IP Flexible Reach Border Element) before routing to an alternate dial peer (e.g. other AT&T IP Flexible Reach Border Element).

Note: Testing was conducted at AT&T labs.
Configuring Microsoft Lync Server 2010

The following items are useful references relating to Microsoft Lync Server 2010.

- The technical library for Microsoft Lync Server 2010 communications software
- Microsoft Lync Server 2010 home page
  - [http://lync.microsoft.com/En-us/Pages/default.aspx](http://lync.microsoft.com/En-us/Pages/default.aspx)
- Cumulative Updates for Microsoft Lync Server 2010 – Microsoft Support
  - [http://support.microsoft.com/kb/2493736](http://support.microsoft.com/kb/2493736)
- Microsoft Lync Server 2010 Protocol Workloads Poster Download
- Configuring Media Bypass for Microsoft Lync 2010

The screenshots in this section show the Microsoft Lync Server 2010 configurations used in the lab for feasibility testing. Production configurations will vary based on many factors including IP Addresses, TCP/UDP ports, site DID’s, addition of leading digits for an outside line, normalization rules, etc.

The screenshot examples below include:

- Lync Server 2010 Topology Builder – Adding Cisco UBE as a PSTN Gateway
- Lync Server 2010 Topology Builder – Assigning the Cisco UBE PSTN Gateway to a Microsoft Mediation Server
- Lync Server 2010 Control Panel – Dial Plan
- Lync Server 2010 Control Panel – Voice Policy
- Lync Server 2010 Control Panel – Voice Route
- Lync Server 2010 Control Panel – Trunk Configuration

Note: Testing was conducted at AT&T labs.
Add the Cisco UBE as a PSTN Gateway in Topology Builder. This illustrates adding a Cisco UBE with IP Address 10.60.60.20, listening on port 5060 and using TCP for the SIP Transport Protocol.

Figure 1: Topology Builder - Adding Cisco UBE as a PSTN Gateway

Note: Testing was conducted at AT&T labs.
Assign the newly created PSTN Gateway to a Mediation Server. Note: This example uses a collocated mediation server listening on port 5068. The dialpeer in the Cisco UBE would have to be configured as such. (i.e. session target ipv4:10.60.60.15:5068)

Figure 2: Topology Builder – Assigning the Cisco UBE PSTN Gateway to the Mediation Server

Note: Testing was conducted at AT&T labs.
This example user has been enabled for Enterprise Voice on the Lync 2010 Server. Note that the account telephone number follows the E.164 format as show in the Line URI field.

Figure 3: Control Panel - Lync Server User

Note: Testing was conducted at AT&T labs.
Note: Testing was conducted at AT&T labs.
The Global Dial Plan was used in this example and has various Associated Normalization Rules.

- 4 Digit – Prefix of +1732320 added to the dialed 4 digit string.
- 7 Digit – Prefix of +1732 added to the dialed 7 digit string.
- 10 Digit – Prefix of +1 added to the dialed 10 digit string.
- 11 Digit – Prefix of + added to the dialed 11 digit string.
- International – Prefix of + added to the dialed string starting with 011.
- N11 – Prefix of + added to the dialed 3 digit string
Figure 4: Control Panel - Dial Plan

**Note:** Testing was conducted at AT&T labs.
Normalization Rule details shows a prefix of +1732 added to the dialed 7 digit string.

Figure 5: Control Panel - Dial Plan Normalization Rule

Note: Testing was conducted at AT&T labs.
Global Voice Policy associates the PSTN Usage Records with Associated Routes.

Figure 6: Control Panel - Voice Policy

Note: Testing was conducted at AT&T labs.
Voice Route example showing the dialed patterns that will route to the Cisco UBE with the IP Address of 10.60.60.20.

Figure 7: Control Panel - Voice Route

Note: Testing was conducted at AT&T labs.
Translation rules can be applied to calls destined to the PSTN Gateway (i.e. Cisco UBE). In this example, translations were not applied.

![Microsoft Lync Server 2010 Control Panel](image)

Figure 8: Trunk Configuration

<table>
<thead>
<tr>
<th>Translation rule</th>
<th>State</th>
<th>Pattern to match</th>
<th>Translation pattern</th>
</tr>
</thead>
<tbody>
<tr>
<td>PBX Translation Rule</td>
<td>Committed</td>
<td>^+1732333\d(4)$</td>
<td>$1</td>
</tr>
<tr>
<td>PSTN Translation Rule</td>
<td>Committed</td>
<td>^+1\d(10)$</td>
<td>$1</td>
</tr>
<tr>
<td>International Translation Rule</td>
<td>Committed</td>
<td>^+(011\d+)$</td>
<td>$1</td>
</tr>
<tr>
<td>N11</td>
<td>Committed</td>
<td>^+(\d{2})\d$</td>
<td>$1</td>
</tr>
</tbody>
</table>

Note: Testing was conducted at AT&T labs.
Network Configuration – Global Screen

Note: Testing for this certification was done with Media Bypass disabled. However, subsequent testing has been performed with Media Bypass enabled and is therefore supported. Details on configuring Media Bypass can be found at the following url: http://technet.microsoft.com/en-us/library/gg413028%28v=ocs.14%29.aspx

Figure 9 Network Configuration
Network Configuration – Edit Global Settings

Figure 10 Network Configuration

Note: Testing was conducted at AT&T labs.
### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>BVoIP</td>
<td>AT&amp;T Business Voice over IP</td>
</tr>
<tr>
<td>CPN</td>
<td>Calling Party Number</td>
</tr>
<tr>
<td>cRTP</td>
<td>Compressed Real Time Protocol</td>
</tr>
<tr>
<td>DID</td>
<td>Direct Inward Dialing</td>
</tr>
<tr>
<td>NAT</td>
<td>Network Address Translation</td>
</tr>
<tr>
<td>NPA</td>
<td>Numbering Plan Area (i.e. area code)</td>
</tr>
<tr>
<td>RTP</td>
<td>Real Time Protocol</td>
</tr>
<tr>
<td>UBE</td>
<td>Unified Border Element</td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted at AT&T labs.
Important Information

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Note: Testing was conducted at AT&T labs.