

**AT&T IP Flexible Reach
And IP Toll Free
Cisco Call Manager
Configuration Guide**

**Issue 2.13
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1 Introduction

This document provides a configuration guide to assist Cisco Call Manager administrators in connecting to AT&T IP Flexible Reach Services and IP Toll Free.

2 Special Notes

Emergency 911/E911 Services Limitations

While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide in detail to understand these limitations and restrictions.

Fax Not Supported on G.729 Calls

Customers using the AT&T IP Flexible Reach Service generally use the G.729 Codec for voice calls. When a customer makes a fax call rather than a voice call, IP Flexible Reach service detects this fax tone and converts the call to the G.711 ulaw Codec. This codec provides greater reliability than G.729 for fax calls. This change to G.711 however is not currently supported by Cisco on H.323 trunks.

Unity Voice Mail Access Does Not Work From a TDM PBX Site

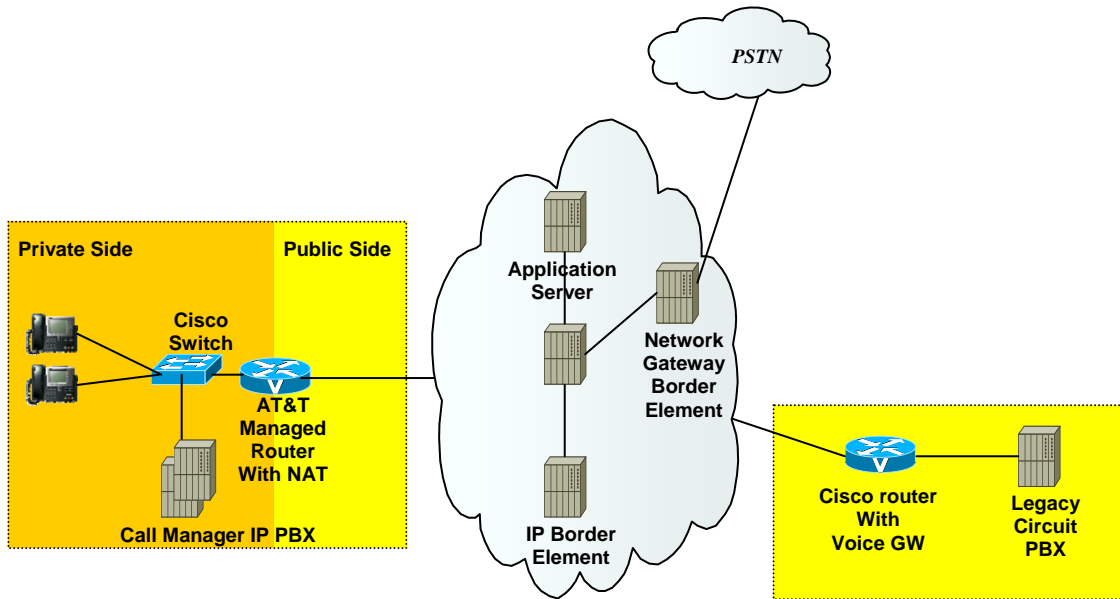
Unity Voice Mail access from a TDM PBX with a SIP gateway in the AT&T managed router (i.e. with IP Long Distance or IP Flexible Reach Service) is not supported. Unity Voice Mail access works fine from a PSTN endpoint.

Certain Unattended Transfers to the PSTN Result in No Ring Back

Certain unattended transfers to PSTN endpoints do not return ring back prior to connect.

3 Overview

This section provides a service overview of the Cisco Call Manager integration with AT&T IP Flexible Reach. The components are shown next.



3.1 Call Manager Site

The Call Manager site consists of the following components.

- Cisco phones (customer managed) – These may be hard phones or soft phones.
- Cisco switch (customer managed) – This is the Cisco switch to provide power to the phones.
- Call Manager IP PBX (customer managed) – This is the Call Manager server.
- AT&T Managed Router (AT&T managed) – This is the router managed by AT&T. The router shall perform packet marking and QOS for voice. This router will support NAT (dynamic NAT for the phones, static NAT for the Call Manager) in the VoPNT VPN configuration. IOS 12.4.6T6 is recommended in order to support the TCP keep alive functions in Call Manager.

3.2 TFTP and DHCP Configuration Guidelines

The guidelines must be followed for TFTP configuration. Phones at the Call Manager must have their TFTP server set to the **private address** of the local Call Manager. This can be configured in the DHCP server used by the Call Manager site phones.

4 Configuration Guide

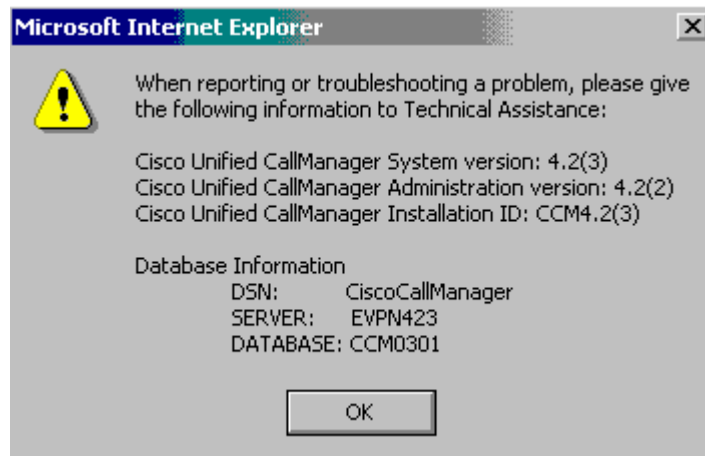
This configuration guide specifies the Call Manager screens that must be configured and updated to support the AT&T IP Flexible Reach.

4.1 Call Manager Version

For IP Flexible Reach, the Call Manager must be running one of the following releases: 4.1.(3), 4.2.(1), or 4.2.(3). The most recent service release is adequate. (See section 5 for IP Toll Free requirements.) You can check the version of Call Manager from the about option on the help menu and then selecting details as shown next.



4.2



Dual AT&T Border Element Connection

This section describes the procedure for connecting to dual AT&T Border Elements.

4.2.1 Gateway

This screen specifies the parameters for connecting to each of the AT&T Border Elements. Two gateways must be configured (i.e. one for each border element). Note that only border element is shown. It is accessed from the Device menu. Key parameters are:

Device Name: This is the IP address of the AT&T Border Element specified in this gateway screen. Two gateway screens are recommended (one for each AT&T Border Element). Sample IP addresses are shown in the screens.

PLEASE CONTACT YOUR CUSTOMER CARE REPRESENTATIVE FOR THE AT&T IP BORDER ELEMENT IP ADDRESSES FOR YOUR SPECIFIC PBX.

Device Pool: Device pool points to a region that specifies the codec and media resources (e.g. conference bridge, transcoder, etc) to be used.

Significant Digits: Number of right most digits that will be extracted from the called number. *For the virtual telephone number (VTN) feature, this field must be set to ALL.*

Wait for Far End H.245 Terminal Capability Set – This field must be unchecked.

Enabled Inbound FastStart: This specifies that the Call Manager can accept the H.323 fast start element in the call setup messages.

Calling Party Presentation – Must set to "Allowed".

Caller ID DN – Set this to the phone number prefix for this site. Call Manager will prepend this string to the extension when the calling number is generated.

Calling Search Space – If the Call Manager phones are assigned to a calling search space, then the calling search space on the gateway form must be set to the same value.

AT&T IP Flexible Reach
Call Manager Configuration Guide

Cisco CallManager 4.1 Administration - Gateway Configuration

File Edit View Favorites Tools Help

Device Information

Device Name* 12.194.180.6

Description 12.194.180.6

Device Pool* WAN

Call Classification* Use System Default

Media Resource Group List 172.16.6.2

Location < None >

AAR Group < None >

Signaling Port* 1720

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Wait for Far End H.245 Terminal Capability Set

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")

MLPP Indication Not available on this device

MLPP Preemption Not available on this device

Call Routing Information

Inbound Calls

Significant Digits* All

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

☒ Redirecting Number IE Delivery - Inbound

☒ Enable Inbound FastStart

The screenshot shows the 'Cisco CallManager 4.1 Administration - Gateway Configuration' window. The 'Outbound Calls' section is highlighted in yellow. It contains several configuration options, some marked with an asterisk to indicate they are required. The settings are as follows:

Configuration Item	Value
Calling Party Selection*	Originator
Calling Party Presentation*	Allowed
Called party IE number type unknown*	National
Calling party IE number type unknown*	National
Called Numbering Plan*	ISDN
Calling Numbering Plan*	ISDN
Caller ID DN	
<input checked="" type="checkbox"/> Display IE Delivery	
<input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound	
<input type="checkbox"/> Enable Outbound FastStart	
Codec For Outbound FastStart*	G711 u-law 64K

* indicates required item

4.2.2 Route Group

The route group specifies a set of alternate routes. If the connection to one AT&T Border element fails, Call Manager will attempt a connection to the other border element. It is accessed from the Route Plan menu. Key fields are:

Route Group Name: This is name of the route group.

Distribution Algorithm: Set this to top down. The first BE will be primary and the second will be backup.

Selected Devices: These are the gateway devices previously configured.

AT&T IP Flexible Reach
Call Manager Configuration Guide

Cisco CallManager 4.1 Administration - Route Group Configuration

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home Search Favorites Media Print

Address E:\CCM41 screenshots\Cisco CallManager 4_1 Administration - Route Group Configuration.mh Go

Route Group: ATTBE
Status: Ready
Update Delete

Route Group Information
Route Group Name* ATTBE
Distribution Algorithm* Top Down

Route Group Member Information
Find Devices to Add to Route Group
Device Name contains Find
Available Devices (select device, then select port below)
Port(s) None Available Add to Route Group

Current Route Group Members
Reverse Order of Selected Devices
Selected Devices* (ordered by highest priority)
12.194.226.70 (All Ports)
12.194.226.198 (All Ports)

Local intranet

4.2.3 Route List

In order for the route group to be used in a route pattern, a route group must be put in a route list. It is accessed from the route plan menu. Key fields are:

Route List Name: The name of the route list.

Selected Groups: This is the route group previously defined for the 2 AT&T Border Elements.

The screenshot shows the 'Cisco CallManager 4.1 Administration - Route List Configuration' window. The window has a menu bar (File, Edit, View, Favorites, Tools, Help) and a toolbar with icons for Back, Forward, Stop, Refresh, Home, Search, Favorites, and Media. The address bar shows the path 'E:\CCM41 screenshots\Cisco CallManager 4_1 Administration - Route List'. The main content area is divided into two sections: 'Route List Information' and 'Route List Member Information'. In the 'Route List Information' section, the 'Route List Name*' field is set to 'ATTBE', the 'Description' field is empty, the 'Cisco CallManager Group*' dropdown is set to 'Default', and the checkbox 'Enable this Route List (change effective on Update; no reset requi' is checked. In the 'Route List Member Information' section, there is an 'Add Route Group' button and a 'Selected Groups*' list box containing 'ATTBE[non-QSIG]'. The list box is labeled '(ordered by highest priority)'. The status bar at the bottom indicates 'Local intranet'.

Route List Information	
Route List Name*	ATTBE
Description	
Cisco CallManager Group*	Default
<input checked="" type="checkbox"/> Enable this Route List (change effective on Update; no reset requi	

Route List Member Information	
<button>Add Route Group</button>	
Selected Groups* (ordered by highest priority)	ATTBE[non-QSIG]

4.2.4 Route Pattern

This screen specifies the called number patterns that should be used to determine which calls are to be sent to the AT&T IP Flexible Reach. It is accessed from the Route Plan menu. Multiple route patterns may need to be configured (e.g. for site to site, US offnet and international offnet calls). Key fields include:

Route Pattern: Specifies the dialing prefix and called number match.

Gateway/Route List: Must be set to the route list configured for the dual AT&T Border Elements.

Prefix Digits: These digits will be pre-pended to the calling party extension. The next section describes how to configure a calling party number translation from an internal extension to a completely different external number.

Discard Digits: Discard the dialed digits before the dot.

Cisco CallManager 4.1 Administration - Route Pattern Configuration

File Edit View Favorites Tools Help

Status: Update completed
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Copy Update Delete

Pattern Definition

Route Pattern* 81.1XXXXXXXXXX

Partition < None >

Description ATT BE

Numbering Plan* North American Numbering Plan

Route Filter < None >

MLPP Precedence Default

Gateway or Route List* ATTBE (Edit)

Route Option
☒ Route this pattern
☐ Block this pattern — Not Selected —

Call Classification* OffNet

☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Prior

☐ Require Forced Authorization Code

Authorization Level 0

☐ Require Client Matter Code

Local intranet

AT&T IP Flexible Reach
Call Manager Configuration Guide

Cisco CallManager 4.1 Administration - Route Pattern Configuration

File Edit View Favorites Tools Help

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

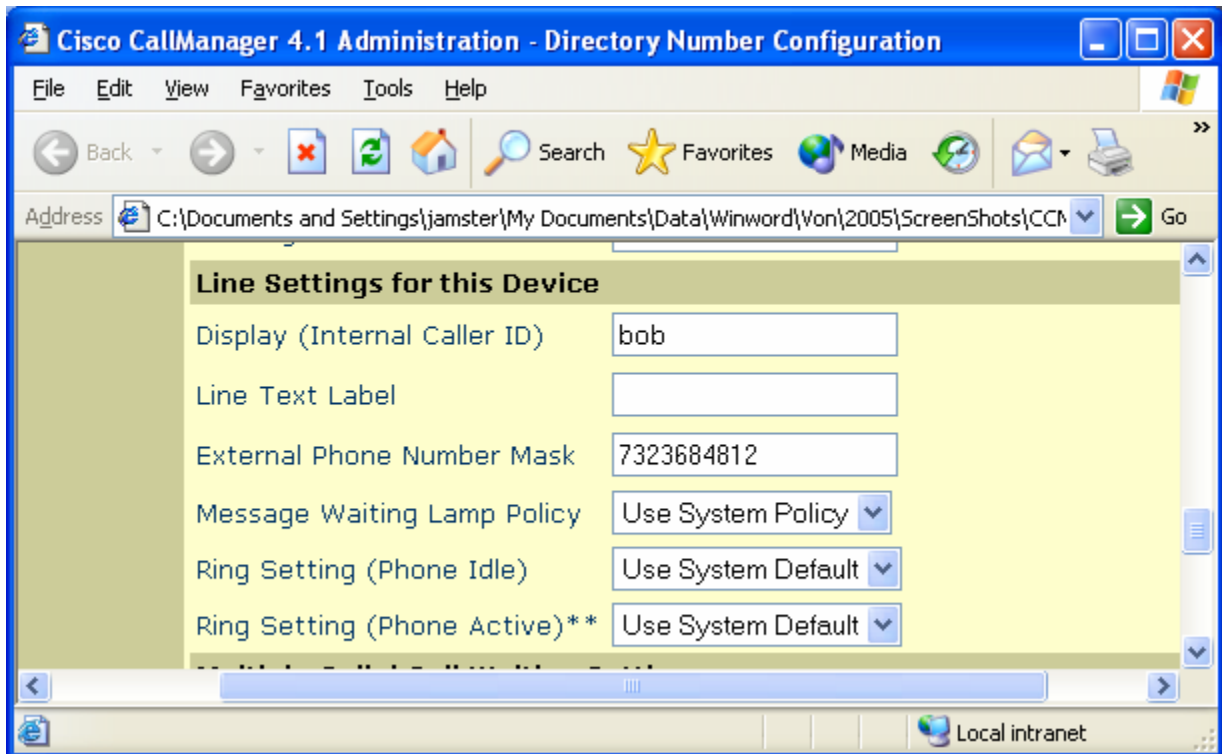
Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text"/>	<input type="text"/>

Local intranet

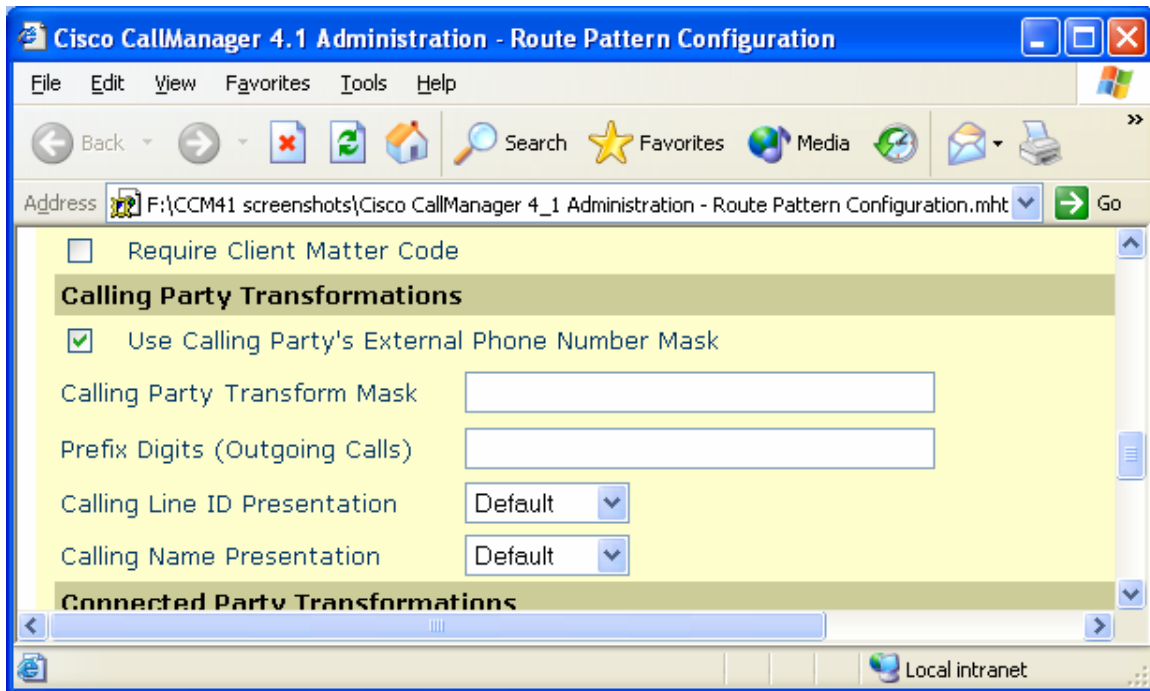
4.3 Calling Party Number Translation Configuration

This section specifies how to configure a calling party number translation from an internal extension to a completely different external number. This configuration is required for the Virtual Telephone Number (VTN) feature.

On the directory number screen for the internal extension, set the **External Phone Number Mask** to the desired external calling party number.



On each route pattern used to route to the AT&T network, check the **Use Calling Party's External Phone Number Mask** field. You must remove the entries in the other calling party transformation fields.

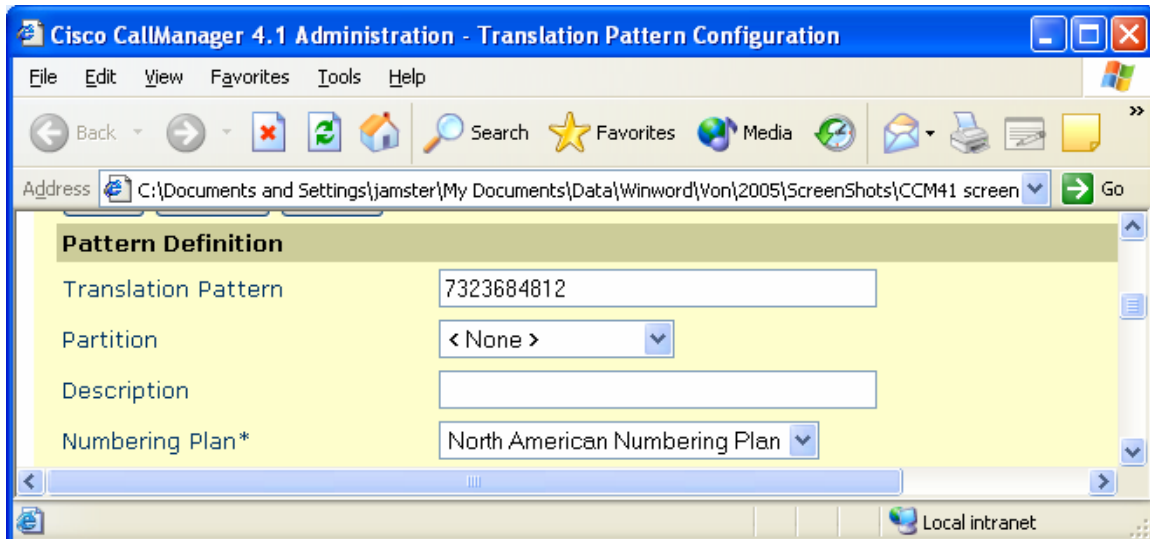


4.4 Incoming Call Routing on Telephone Number

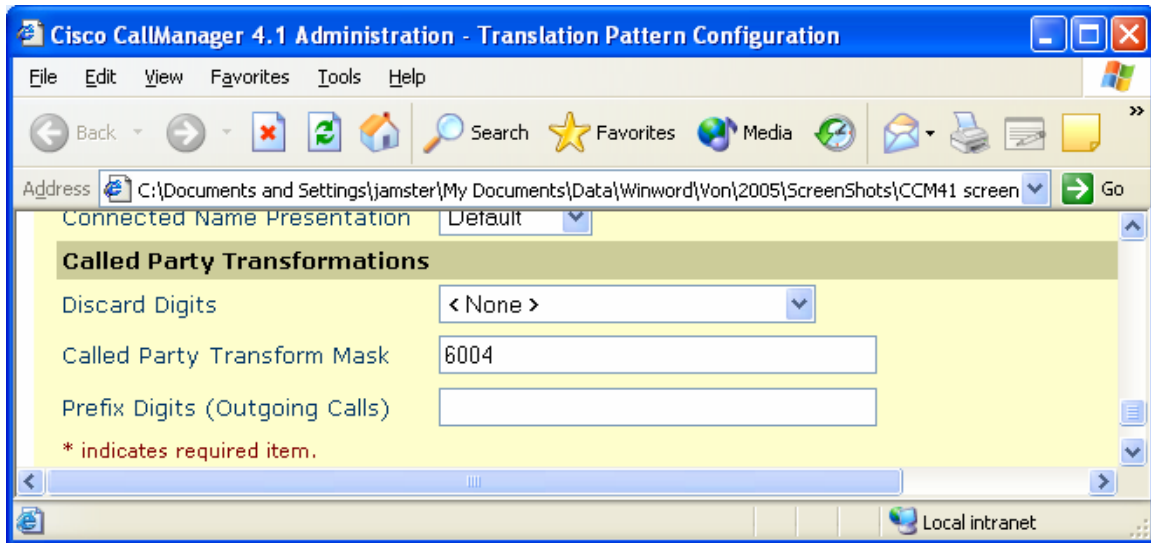
When using the AT&T Virtual Telephone Number feature (VTN), the AT&T network will send the call to the PBX using a full E.164 public number. This number can be mapped to an internal extension on the following screen.

When using a non virtual Telephone Number, the AT&T network will send the call to the PBX using 7 digits or less as requested by the customer.

On the translation pattern screen, put the called number sent from AT&T in the **translation pattern** field and put the internal extension in the **called party transform mask** field. This example maps the public called number "7323684812" to the internal extension "6004" for internal routing.



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Call Manager Configuration Guide



4.5 Region - WAN

The region specifies the codec to be used for devices that point to this region. This particular region is used by devices associated with the AT&T Border Elements. This screen is accessed from the system menu. Key fields are:

Default: Specifies that the G.729 codec to be used with the default region.

WAN: Specifies that the G.729 codec to be used with other devices in this region.

Region Information

Region Name*

Call Information

The maximum audio codec/video bandwidth supported within this region and between 1 other regions are:

Region	Audio Codec	Video Call Bandwidth
Default	<input type="text" value="G.729"/>	<input type="radio"/> None <input checked="" type="radio"/> <input type="text" value="384"/> kbps
WAN (Within this Region)	<input type="text" value="G.729"/>	<input type="radio"/> None <input checked="" type="radio"/> <input type="text" value="384"/> kbps

Items per page First Previous Next Last Page of 1

* indicates required item

Done Local intranet

4.6 Region – Default

The region specifies the codec to be used for devices that point to this region. This particular region is used by devices associated with non AT&T components (i.e. phones). This screen is accessed from the system menu. Key fields are:

Default: Specifies that the G.711 codec to be used with the default region. This means that G.711 will be used between the Call Manager phones at this site.

WAN: Specifies that the G.729 codec to be used with devices in the WAN region. This means that G.729 will be used between the Call Manager phones and any endpoint reach via the AT&T Border Element.

Cisco CallManager 4.1 Administration - Region Configuration

File Edit View Favorites Tools Help

Address E:\CCM41 screenshots\Cisco CallManager 4_1 Administration - Def Region Configuration.mf Go

Region Name* Default

Call Information

The maximum audio codec/video bandwidth supported within this region and between 1 other regions are:

Region	Audio Codec	Video Call Bandwidth
Default (Within this Region)	G.711	<input type="radio"/> None <input checked="" type="radio"/> 384 kbps
WAN	G.729	<input type="radio"/> None <input checked="" type="radio"/> 384 kbps

Items per page 10 First Previous Next Last Page 1 of 1

* indicates required item

Local intranet

4.7 Device Pool – WAN

The device pool specifies parameters to be used by devices that point to this device pool. This particular device pool is used by devices associated with the AT&T Gatekeeper. This screen is accessed from the system menu. Key fields are:

Region: The region specifies the codec to be used for this device pool.

Media Resource Group List: This points to a media resource group that specifies media resources (i.e. conference bridges, transcoders, etc) to be used for this device pool.

Cisco CallManager 4.1 Administration - Device Pool Configuration

File Edit View Favorites Tools Help

Device Pool Name* WAN

Cisco CallManager Group* Default

Date/Time Group* CMLocal

Region* WAN

Softkey Template* Standard User

SRST Reference* Disable

Calling Search Space for Auto-registration <None >

Media Resource Group List 172.16.6.2

Network Hold MOH Audio Source <None >

User Hold MOH Audio Source <None >

Network Locale <None >

User Locale <None >

Connection Monitor Duration* * * 120

Multilevel Precedence and Preemption (MLPP) Information

MLPP Indication* Default

MLPP Preemption* Default

MLPP Domain (e.g., "0000FF")

4.8 Device Pool – Default

The device pool specifies parameters to be used by devices that point to this device pool. This particular device pool is used by devices associated with non AT&T components (i.e. phones). This screen is accessed from the system menu. Key fields are:

Region: The region specifies the codec to be used for this device pool.

Media Resource Group List: This points to a media resource group that specifies media resources (i.e. conference bridges, transcoders, etc) to be used for this device pool.

The screenshot displays the 'Cisco CallManager 4.1 Administration - Device Pool Configuration' window. The interface includes a menu bar (File, Edit, View, Favorites, Tools, Help) and a list of configuration fields. The fields are as follows:

Field	Value
Device Pool Name*	Default
Cisco CallManager Group*	Default
Date/Time Group*	CMLocal
Region*	Default
Softkey Template*	Standard User
SRST Reference*	Disable
Calling Search Space for Auto-registration	< None >
Media Resource Group List	172.16.6.2
Network Hold MOH Audio Source	< None >
User Hold MOH Audio Source	< None >
Network Locale	< None >
User Locale	< None >
Connection Monitor Duration***	
Multilevel Precedence and Preemption (MLPP) Information	
MLPP Indication*	Default
MLPP Preemption*	Default
MLPP Domain (e.g., "0000FF")	

* indicates required item

4.9 Media Resource Group

The media resource group specifies a grouping of media resources (i.e. conference bridge, transcoder, etc). This particular media resource group contains a conference bridge and a transcoder. This screen is accessed from the service/media resource menu. Key fields are:

Selected Media Resources: Points to profiles for conference bridge (CFB) and transcoder (XCODE) resources to be used in this group.

Cisco CallManager 4.1 Administration - Media Resource Group Configuration

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home Search Favorites Media

Address E:\CCM41 screenshots\Cisco CallManager 4_1 Administration - Media Resource Group Configuration.mht Go

Media Resource Group Information

Media Resource Group Name* 172.16.6.2

Description 172.16.6.2

Devices for this Group

Available Media Resources** 135.25.30.136 (CFB)

Selected Media Resources* CCM2 (ANN)
CFB_CCM2 (CFB)
MOH_CCM2 (MOH)
MTP_CCM2 (MTP)
mtp000DBC4d2770 (XCODE)

☐ Use Multicast for MOH Audio (requires at least one multicast MOH resource)

* indicates required item

** Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH), and Transcoders (XCODE)

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4.10 Media Resource Group List

The media resource group list is a grouping of media resource groups. This screen is accessed from the service/media resource menu.

The screenshot shows the 'Cisco CallManager 4.1 Administration - Media Resource Group List Configuration' window. The title bar includes standard Windows window controls. The menu bar contains 'File', 'Edit', 'View', 'Favorites', 'Tools', and 'Help'. The main content area has a yellow background and displays the following information:

- Media Resource Group List: 172.16.6.2 (used by 20 devices)**
- Status: Ready**
- Four buttons: 'Copy', 'Update', 'Delete', and 'Reset Devices'.
- Media Resource Group List Information**
- A text field for 'Media Resource Group List Name*' containing '172.16.6.2'.
- Media Resource Groups for this List**
- A section titled 'Available Media Resource Groups' with an empty list box.
- Below the list box are two small triangular buttons (downward and upward arrows).
- A section titled 'Selected Media Resource Groups*' containing a list box with '172.16.6.2'.
- Below the selected list box is the text '(Groups listed in order of priority)'.

The window has a standard Windows scrollbar on the right side.

4.11 Conference Bridge

The conference bridge specifies the conferencing resources to be used by the Call Manager phone users when the phone's conference button is selected. This screen is accessed from the service/media resource menu.

Cisco CallManager 4.1 Administration - Conference Bridge Configuration

File Edit View Favorites Tools Help

Back Forward Stop Refresh Home Search Favorites Media

Address E:\CCM41 screenshots\Cisco CallManager 4_1 Administration - Conference Bridge Confi Go

Conference Bridge: CFB_CCM2 (172.16.6.2)
Registration: Registered with Cisco CallManager CCM2
IP Address: 172.16.6.2

Status: Ready

Copy Update Delete Reset

Conference Bridge Type Cisco Conference Bridge Software

Host Server CCM2

Conference Bridge Name* CFB_CCM2

Description 172.16.6.2

Device Pool* Default

Location < None >

* indicates required item

Local intranet

4.12 Transcoder

The transcoder specifies the transcoding resources to be used to convert from the G.729 to G.711 codecs for conference calls that include a call leg to the AT&T IP Flexible Reach. This screen is accessed from the service/media resource menu. Key fields include:

Host name: This is a pointer to the transcoding resource. The host name is the string "MTP" followed by the MAC address of the Cisco router or switch with the DSP resources being used for transcoding.

Cisco CallManager 4.1 Administration - Transcoder Configuration

File Edit View Favorites Tools Help

Back Forward Home Search Favorites Media Refresh Print

Address E:\CCM41 screenshots\Cisco CallManager 4_1 Administration - Transcoder Configuration.mht Go

Transcoder Configuration

[Back](#)

Transcoder: mtp000DBC4d2770 (Cisco 3725 on CCME)
Registration: Registered with Cisco CallManager CCM2
IP Address: 172.16.6.16

Status: Ready

Transcoder Type Cisco IOS Media Termination Point

Description

Device Name*

Device Pool* ([View details](#))

Special Load Information (Leave blank to use def

* indicates required item

Done Local intranet

4.13 Phone Configuration

This screen specifies the parameters for the Call Manager phones. It is accessed from the Device menu. Key parameters are:

Device Pool: This phone will use the parameters in the “default” device pool.

Media Resource Group List: This phone will use the resources specifies in the designated Media Resource Group List which include the conference bridge and transcoder.

Cisco CallManager 4.1 Administration - Phone Configuration

File Edit View Favorites Tools Help

Back Forward Home Search Favorites Media Refresh Print

Address E:\CCM41 screenshots\Cisco CallManager 4_1 Administration - Phone Configuration.mht Go

Device Information

MAC Address*	000AF4F85E7D
Description	SEP000AF4F85E7D
Owner User ID	(Select User ID)
Device Pool*	Default (View details)
Calling Search Space	<None>
AAR Calling Search Space	<None>
Media Resource Group List	172.16.6.2
User Hold Audio Source	<None>
Network Hold Audio Source	<None>
Location	<None>
User Locale	<None>
Network Locale	<None>

Local intranet

4.14 Service Parameters

This screen specifies the parameters for the Call Manager. It is accessed from the Service menu.

The full list of service parameters are in the following attachment.



Cisco Unified CallManager 4_2 Administration - Service Parameters Configuration.mht

Key parameters are:

H225 TCP Timer: This specifies the amount of time Call Manager will wait after trying to open an H225 channel with the AT&T Border Element before attempting an alternate route.

Check Progress Indicator Before Establishing Media: This parameter specifies that Call Manager must check the progress indicator in an alerting or progress message prior to establishing a media channel. This parameter must be set to "true".

Max Call Duration: This parameter specifies how long a call should last in minutes. The default is 720 (i.e. 12 hours). AT&T recommends that this parameter be set to "0" (no max call duration limit).

Duplex Streaming: This parameter specifies whether 2 way voice streaming will occur when music on hold is implemented. AT&T requires that the default setting of "true" be used.

5 IP Toll Free Specific Information

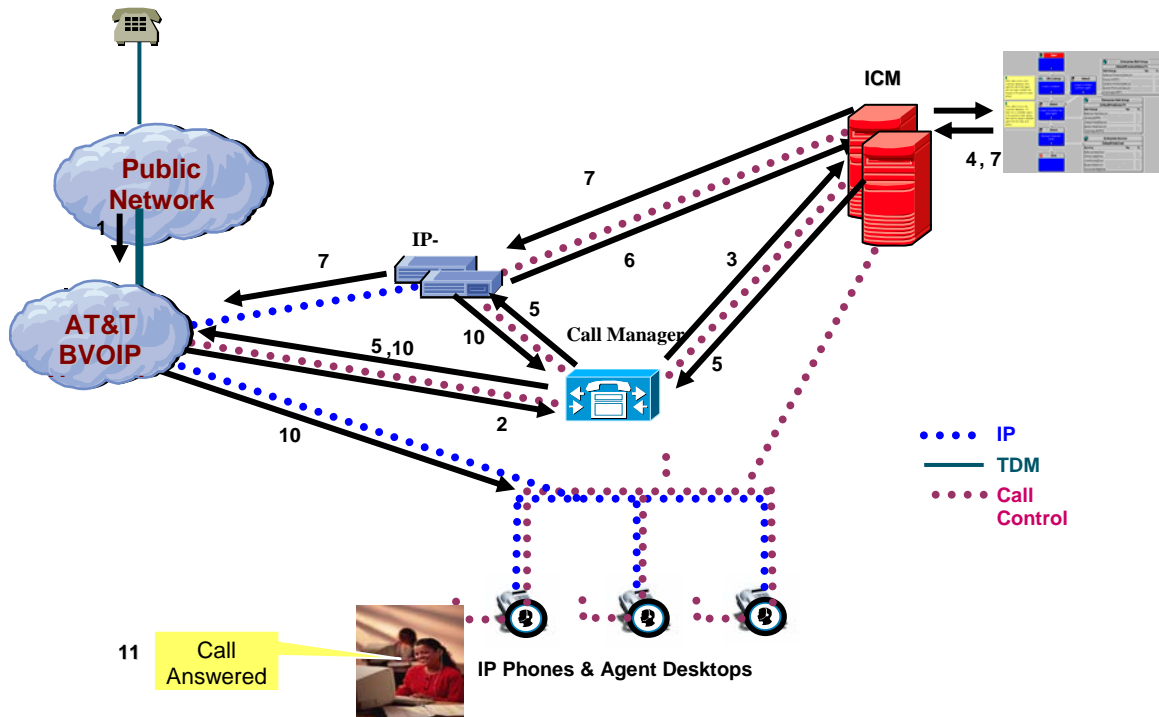
This section provides information specific to IP Toll Free.

5.1 Call Flow with IP Call Center

This section provides a typical call flow for the interaction between the AT&T IP Toll Free Service and the Cisco IP Call Center. **Note that Cisco CVP (Cisco Voice Portal) is NOT currently supported.** The components in this call flow include the following:

- Call Manager - Call Manager is the Cisco IP PBX platform. Call Manager release 4.2.1 is supported for this application.
- IP IVR – IP IVR is the Cisco IVR platform. This platform is used to play voice prompts as needed based on information based from ICM. IP IVR release 4.0.4 is supported for this application.
- ICM – ICM is an IPCC scripting application. This application is used to script the flow of a customer's IPCC application. ICM release 7.0 SR4 is supported for this application.

This call flow is shown in the following figure.



The steps in a typical call flow are described next.

1. Call delivered from PSTN to AT&T BVOIP network.
2. AT&T BVOIP network sends H.323 setup to Cisco CallManager. The AT&T network will always send these calls to Call Manager,
3. Based on the called number, a JTAPI Route Request is sent to the ICM.
4. ICM runs a routing script. Based on the routing script, ICM determine what to do next. For example, if no available agent is found, the IP IVR label is returned from routing script.
5. ICM instructs Cisco CallManager to transfer call to IP IVR, and Cisco CallManager does as instructed.
6. IP IVR notifies ICM that call has arrived.
7. ICM instructs IP IVR to play queue announcements back through BVOIP network.
8. Agent becomes ready (completed previous call or just went ready).
9. ICM sends call data to selected agent screen and instructs the IP IVR to transfer the call to the agent phone.
10. IP IVR transfers the VoIP voice path to selected agent phone.
11. Call is answered by agent.

5.2 Examples for Call Manager Configuration for IPCC

The following sections describe an example set of Call Manager configuration screens that can be used to access the Cisco IPCC application. However, these sections should be used as examples of how IP Toll Free dialed calls can be mapped to a Cisco IPCC application. The customer must consult Cisco documentation to determine the exact configuration they will require.

In addition, this document does not cover configuration for ICM and IP IVR.

5.2.1 CTI Route Points

A computer telephony integration (CTI) route point designates a virtual device that can receive multiple, simultaneous calls for application-controlled redirection. With Cisco IPCC, CTI route points are configured with Call Manager with associated directory numbers. The directory numbers correspond to the numbers received from the AT&T IP Toll Free Service. When a call is received, Call Manager matches the called number with a directory number. If that directory number corresponds to a CTI route point controlled by the JTAPI interface to ICM, then a route request is passed to ICM.

Sample CTI route point and associated directory number screens are shown next. The CTI route point is accessed from the Call Manager Devices menu.

The screenshot displays the 'CTI Route Point Configuration' window in the Cisco Unified CallManager 4.2 Administration interface. The window title is 'Cisco Unified CallManager 4.2 Administration - CTI Route Point Configuration'. The address bar shows the path: 'C:\Documents and Settings\jamster\My Documents\Data\Winword\Won\2007\CiscoCallManager\Cisco Unified CallManager 4_2 Administr...'. The main content area is titled 'CTI Route Point Configuration' and includes several sections:

- Directory Numbers:** A list of directory numbers associated with the route point. It shows 'Line 1 - 000001030004 (no Partition)' and 'Line 2 - Add DN'.
- Device Information:** A section containing fields for 'Device Name*' (DN_000001030004), 'Description' (DN_7323680456), 'Device Pool*' (Default), 'Common Profile' (<None>), 'Calling Search Space' (<None>), 'Location' (<None>), 'Media Resource Group List' (<None>), and 'User Hold Audio Source' (<None>). Each field has a dropdown menu or a text input field.
- Registration:** A section showing 'Registered with Cisco CallManager 172.16.6.2' and 'IP Address: 172.16.6.2'.
- Status:** A section showing 'Status: Ready'.

At the bottom of the window, there is a status bar that says 'Done, but with errors on page.' and a 'Local intranet' icon.

The associated directory number screen is attached next.



Cisco Unified CallManager 4_2 Administration - Directory Number Configuration.mht

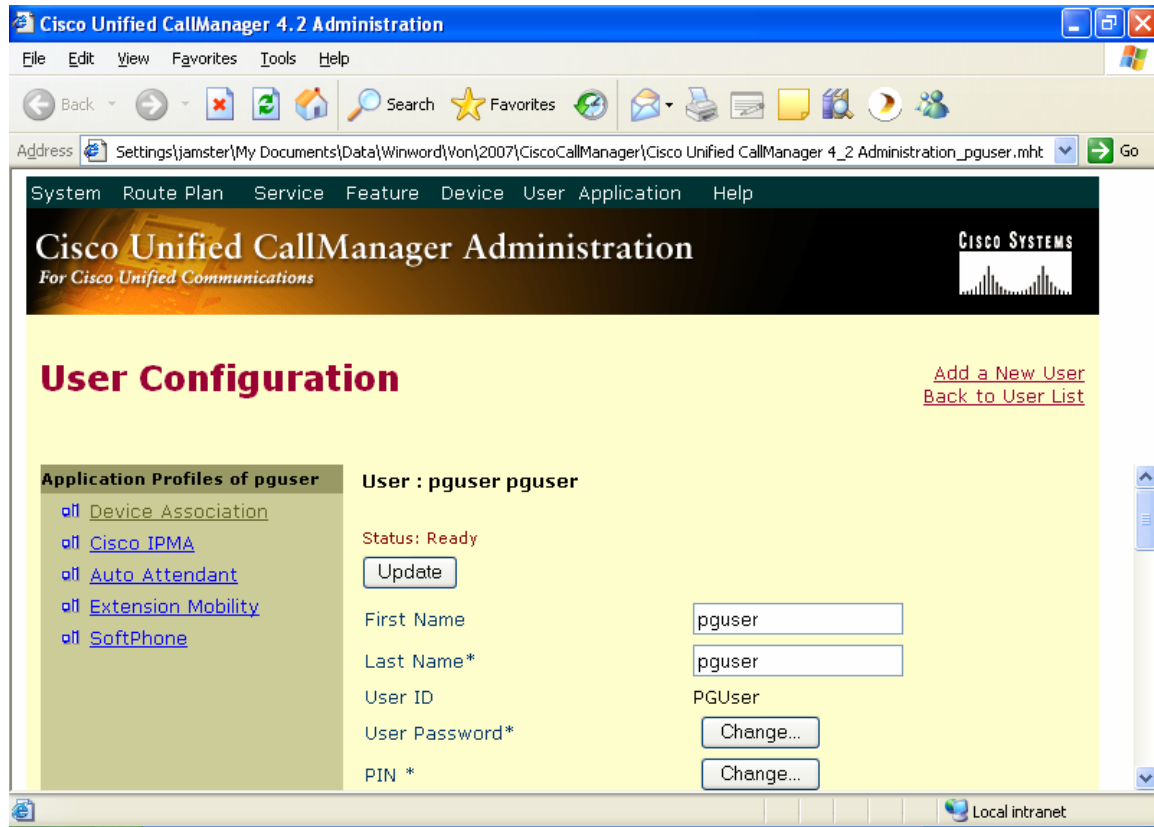
5.2.2 Incoming Call Routing for IP Toll Free

For IP Toll Free, the AT&T network will send the call to the PBX using called number digits requested by the customer. This number will be prefixed with 5 leading zeroes. This number can be mapped to an internal extension or a CTI route point directory number on the translation pattern screen (see section 4.4).

On the translation pattern screen, put the IP Toll Free called number string in the **translation pattern** field and put the internal extension or CTI route point directory number in the **called party transform mask** field.

5.2.3 JTAPI User Definitions

A number of JTAPI users must be defined by adding a user to the global directory. This is done by selecting “add user” from the User menu. A portion of the user screen is shown next.



Based on the call flow described in the previous section, the following three user screens were used to configure a Cisco Call Manager that was tested with IPCC.



Cisco Unified CallManager 4_2 Administration_ivradmin.mht



Cisco Unified CallManager 4_2 Administration_ivruser1.mht



Cisco Unified CallManager 4_2 Administration_pguser.mht

6 Appendix A: Router Version/Configuration for Transcoding

6.1 Router Version

AT&T has tested 12.3.11T for transcoding support with Cisco Call Manager. A sample "show version" output is shown next.

```
Cisco IOS Software, 3700 Software (C3725-IPVOICE-M), Version 12.3(11)T,
RELEASE
SOFTWARE (fc2)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2004 by Cisco Systems, Inc.
Compiled Sat 18-Sep-04 16:10 by eaarmas
```

```
ROM: System Bootstrap, Version 12.2(8r)T2, RELEASE SOFTWARE (fc1)
```

```
v00ccm03-3725 uptime is 5 weeks, 2 hours, 31 minutes
System returned to ROM by power-on
System restarted at 18:14:02 UTC Tue Mar 8 2005
System image file is "flash:c3725-ipvoice-mz.123-11.T.bin"
```

6.2 Transcoding Configuration

The router configuration for transcoding is shown next.

```
! Sets dsp services in voice card in slot 1
voice-card 1
no dspfarm
dsp services dspfarm

! Connection to Call Manager
sccp local FastEthernet0/0
sccp
sccp ccm 172.16.6.2 priority 1
!
! Maximum number of calls for transcoder.
dspfarm transcoder maximum sessions 12
!
```

7 Appendix B: Firewall Rules

If the customer implements a firewall, then the following access control rules shall be required.

- 1) Accept AT&T Border Element traffic from TCP ports >1023
- 2) Accept AT&T Border Elements traffic from UDP ports 16384-32767

AT&T IP Flexible Reach
Call Manager Configuration Guide

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