



AT&T IP Flexible Reach Service and AT&T IP Toll-Free on AT&T VPN Service

TDM Gateway Customer Configuration Guide for AT&T IP Flexible Reach Service and AT&T IP Toll-Free on AT&T VPN Service as the Underlying Transport Service for Cisco ISR 4K Platform

August 31, 2015

Version 1.0

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1 Introduction

This document provides the guidelines and specifications required to correctly configure a TDM Gateway to work with AT&T IP Flexible Reach Service (including Enhanced Features Service) and/or AT&T IP Toll-Free, on AT&T VPN Service (“AT&T VPN”) as the Underlying Transport Service. CERs can be utilized for either one of those services or for both services simultaneously. Please ensure your system set-up is consistent with the recommended specifications provided in this document. AT&T reserves the right to modify or update its guidelines at any time without notice, so please check the following link to confirm having the latest version of this document:

<http://www.corp.att.com/bvoip/avpn/implementation/> (login: att, password: attvoip).

You may also consult with your AT&T technical sales representative.

This document should be used as a general configuration guideline. The customer is solely responsible for determining the appropriate configuration based on their specific environment. The example configurations may be mapped to a variety of vendor implementations.

The configuration examples provided in this document are based upon Cisco IOS features; however, the features are **not** described in their entirety and may vary across hardware platforms and versions of IOS. Please refer to the appropriate Cisco documentation relative to your IOS features.

AT&T BVoIP Calling Plans B and C provide Emergency 911/E911 calling capabilities subject to the following limitations and restrictions:

Emergency 911/E911 Services Limitations and Restrictions –AT&T IP Flexible Reach Service Plan B and C (the local calling Plans) provides 911/E911 calling capabilities as is required by the FCC. Customer is solely responsible for programming its premises equipment to enable a User to originate a 911 call in the domestic U.S. over IP Flexible Reach Service E911/911.

While AT&T IP Flexible Reach Service 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 AT&T Business Voice over IP Services (BVoIP) Service Guide, 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 BVoIP Service Guide for AT&T IP Flexible Reach in detail to understand the limitations and restrictions. Note: Calling Plan A is NOT a local calling Plan, and cannot be used to originate a 911 call.

1.1 Supported Hardware

On AT&T IP Flexible Reach Service and/or AT&T IP Toll-Free on AT&T VPN as the Underlying Transport Service, the Customer-managed TDM Gateway is cascaded behind the Customer Edge Router (CER).

AT&T IP Flexible Reach Service and/or AT&T IP Toll-Free on AT&T VPN support the following Cisco ISR 4K platforms as TDM gateways:

Routers supported:

- 4321
- 4331
- 4351
- 4451

Voice cards supported:

The following T1/E1 NIMs will be supported:

NIM-1MFT-T1/E1	1-port multi-flex trunk voice/clear-channel data T1/E1 module
NIM-2MFT-T1/E1	2-port multi-flex trunk voice/clear-channel data T1/E1 module
NIM-4MFT-T1/E1	4-port multi-flex trunk voice/clear-channel data T1/E1 module
NIM-8MFT-T1/E1	8-port multi-flex trunk voice/clear-channel data T1/E1 module
NIM-1CE1T1-PRI	1-port multi-flex trunk voice/channelized data T1/E1 module
NIM-2CE1T1-PRI	2-port multi-flex trunk voice/channelized data T1/E1 module
NIM-8CE1T1-PRI	8-port multi-flex trunk voice/channelized data T1/E1 module
NIM-2FXS	2-port FXS NIM
NIM-4FXS	4-port FXS NIM
NIM-2FXO	2-port FXO NIM
NIM-4FXO	4-port FXO NIM

The Cisco PVDM4's should be installed directly on the T1/E1 NIM (to turn up the voice channels). There is a single slot for a PVDM on each NIM module. The types supported are show below:

Name	Description
PVDM4-32	32-channel, high-density, voice DSP module
PVDM4-64	64-channel, high-density, voice DSP module
PVDM4-128	128-channel, high-density, voice DSP module
PVDM4-256	256-channel, high-density, voice DSP module

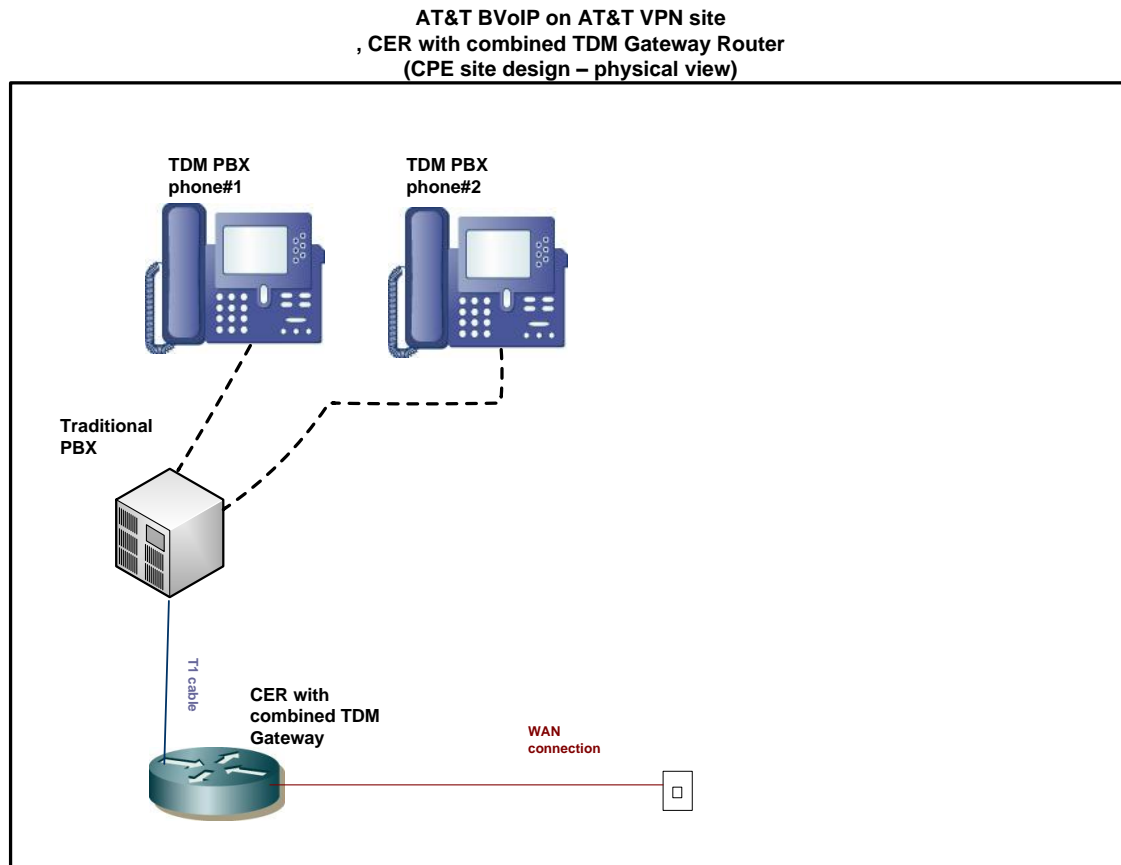
1.2 Software Requirements

Configurations in this guide were tested with Cisco IOS-XE Release 15.4(3)S2 (XE 03.13.02.S).

TDM Gateway will require the UC (Unified Communications) Technology Package License.

2 TDM Gateway Configurations

The following section illustrates a sample network topology diagram for sites with a TDM Gateway.



2.1 Types of Voice Ports on the TDM Gateway

Voice ports are found at the intersection of packet-based networks and traditional telephony networks, and facilitate the passing of voice and call signals between the two networks. Physically, voice ports connect a router to a line from a circuit-switched telephony device in a PBX or the PSTN.

Digital trunking can be accomplished via different signaling types:

1) CAS - Channel Associated Signaling “Robbed Bit Signaling”. T1 uses in-band signaling based on either Super-Frames (bits 6 & 12) or Extended Super-frames (bits 6,12,18 & 24).

2) PRI (U.S.) – Primary Rate Interface; **QSIG/CCS** (Most of World) – QSIG/Common Channel Signaling.

Each of these types will change the available number of channels on the T1 based on the number of channels needed to support the signaling.

Alternatively, if no PBX exists, specific ports on the TDM Gateway router can be directly attached to analog devices (telephones or Fax machines) via FXS or FXO ports.

The following Cisco link provides additional information on how to configure voice ports:

http://www.cisco.com/en/US/docs/ios/voice/voiceport/configuration/guide/15_1/vc_15_1_book.html

2.2 Information on Digital Signal Processors (DSP)

DSPs are necessary for packet Telephony technologies such as AT&T IP Flexible Reach Service and/or AT&T IP Toll-Free on AT&T VPN. You will need to purchase the properly sized Packet Voice DSP Modules (PVDM) which contain DSP’s.

In order to determine the correct PVDM model, you will need to know the number of channels and the codec used. Below is a list of the number of channels supported on the various types of the PVDM4’s categorized by the complexity of the codecs that are supported.

Complexity	PVDM4-32	PVDM4-64	PVDM4-128	PVDM4-256
Low-Complexity Voice	32	64	128	256
Medium-Complexity Voice	24	48	96	192
High-Complexity Voice	16	32	64	128

Low-Complexity Codecs	Medium-Complexity Codecs	High-Complexity Codecs
G.711 Supports clear-channel, and fax and modem pass-through	Fax relay, G.722, G.726, G.729a, and G.729ab.	G.728, G.729, G.729b, Internet Low Bitrate Codec (iLBC), and modem rel

A helpful tool to determine the appropriate type of PVMD4’s to install in the router is the Cisco DSP calculator. Below is a link to the DSP Calculator (requires CCO login):

http://www.cisco.com/web/applicat/dsprecal/dsp_calc.html

2.3 Voice Port Configuration

2.3.1 T1/E1 Digital Voice Ports

Following are the steps to configure the NIM-#MFT-T1/E1 or NIM-#CE1T1-PRI modules.

2.3.1.1 Configure each NIM module for the appropriate type

The NIM-#MFT-T1/E1 or NIM-#CE1T1-PRI modules must be configured for either T1 or E1 with the following command:

Syntax:

```
card type {t1 | e1} <card slot #><card bay #>
```

This command will need to be entered for each NIM module.

Example for T1-

```
card type t1 0 1
```

Example for E1-

```
card type e1 0 1
```

The controller <card slot#><card bay#> <port #> command will be automatically added by the router once the NIM card is installed and set for the appropriate type.

Example for E1-

```
controller E1 0/1/0
```

Example for T1-

```
controller T1 0/1/0
```

2.3.1.2 Configure Clocking

Use the following commands to configure a controller for line clocking. The “clock source primary” command can only be configured on one of the controllers (whichever controller is

configured for “network clock select” command. User should have the ability to configure any single controller as “clock source line”. These commands apply to T1 or E1 controllers.

To configure clocking on the router, enter the following commands:

Syntax:

```
network-clock synchronization automatic  
  
controller <T1/E1>E1 <card slot#><card bay#> <port #>  
clock source line primary  
  
network-clock input-source <priority> controller <T1 or E1> <card slot#><card bay#>  
<port #>
```

Example for E1 port 0/1/0:

```
network-clock synchronization automatic  
  
controller E1 0/1/0  
clock source line primary  
  
network-clock input-source 1 controller E1 0/1/0
```

Additional T1/E1 controllers will use the default clocking (no additional configuration required).

2.3.1.3 Configure T1/E1 Controllers to Match PBX settings

There are two settings on the T1/E1 controllers that should match the setting of the PBX. These settings are framing and linecode. Configure these settings as appropriate.

“Framing” options

T1 Options: **sf** *or* **esf**

E1 Options: **crc4** *or* **no-crc4** *or* **unframed**

“Linecode” options

T1 Options: **ami** or **b8zs**

E1 Options: **ami** or **hdb3**

Sample Configuration for T1 :

```
controller T1 0/2/0
framing esf
linecode b8zs
```

2.3.1.4 Configure the controller card for CAS:

Determine if the interface to the PBX is CAS. If a CAS configuration is required, follow the steps below:

Step 1 – Configure the CAS controller:

Syntax:

```
ds0-group <ds0-group#> timeslots <time slot range> type <switch type>
```

- Ds0-group# is always “0”
- Timeslot range for T1
 - Full T1 (24 channels) is always “1-24” as a default
 - Example: ds0-group 0 timeslots 1-24 type e&m-wink-start
 - Sub-T1 (23 channels or less)
 - Example 12 channels starting at timeslot 1: ds0-group 0 timeslots 1-12 type e&m-wink-start
 - Example – 10 channels starting at timeslot 11: ds0-group 0 timeslots 11-20 type e&m-wink-start
- Timeslot range for E1
 - Full E1 (30 channels) is always “1-31” as a default
 - Example: ds0-group 0 timeslots 1-31 type e&m-wink-start
 - Sub E1 (29 channels or less)
 - Example 12 channels starting at timeslot 1: ds0-group 0 timeslots 1-12 e&m-wink-start
 - Example 10 channels starting at timeslot 11: ds0-group 0 timeslots 11-21 e&m-wink-start

- The **switch type** must be set to match the PBX. Options are:
 - e&m-delay-dial E & M Delay Dial
 - e&m-fgd E & M Type II FGD
 - e&m-immediate-start E & M Immediate Start
 - e&m-lmr E & M land mobil radio
 - e&m-melcas-delay MEL CAS (CEPT) E & M Delay Start
 - e&m-melcas-immed MEL CAS (CEPT) E & M Immediate Start
 - e&m-melcas-wink MEL CAS (CEPT) E & M Wink Start
 - e&m-wink-start E & M Wink Start
 - ext-sig External Signaling
 - fxo-ground-start FXO Ground Start
 - fxo-loop-start FXO Loop Start
 - fxo-melcas MEL CAS (Mercury) FXO
 - fxs-ground-start FXS Ground Start
 - fxs-loop-start FXS Loop Start
 - fxs-melcas MEL CAS (Mercury) FXS
 - none Null Signalling for External Call Control
 - r2-analog R2 ITU Q411
 - r2-digital R2 ITU Q421
 - r2-pulse R2 ITU Supplement 7

Example of T1 CAS:

```
controller T1 0/2/0  
ds0-group 0 timeslots 1-24 type e&m-wink-start
```

Example of E1 CAS:

```
controller E1 0/2/0  
ds0-group 0 timeslots 1-31 type e&m-wink-start
```

For CAS, a voice-port interface will be automatically created by the router for each controller:

Example of CAS voice port created by the router:

```
voice-port 0/2/0:0
```

Step 2 - Add the interdigit timeout command to the CAS voice port :

Syntax :

```
timeouts interdigit 5
```

Example :

```
voice-port 0/2/0:0  
timeouts interdigit 5
```

T1 CAS Sample Configuration :

Example for NIM-2MFT-T1/E1 installed in card slot 0, card bay 1

```
card type t1 0 1  
  
network-clock synchronization automatic  
network-clock input-source 1 controller T1 0/1/0  
!  
!  
controller T1 0/1/0  
clock source line primary  
framing esf  
linecode b8zs  
ds0-group 0 timeslots 1-24 type e&m-wink-start  
  
controller t1 0/1/1  
framing esf  
linecode b8zs  
ds0-group 0 timeslots 1-24 type e&m-wink-start  
  
voice-port 0/1/0:0  
timeouts interdigit 5  
  
voice-port 0/1/1:0  
timeouts interdigit 5
```

2.3.1.5 Configure the controller card for PRI:

Determine if the interface to the PBX is PRI. If a PRI configuration is required, follow the steps below:

Step 1 - The PRI switch-type will need to be configured globally.

The switch type must match the PBX configuration..

Syntax :

```
isdn switch-type <PRI switch type>
```

Choices for **PRI switch type** are:

- primary-4ess Lucent 4ESS switch type for the U.S.
- primary-5ess Lucent 5ESS switch type for the U.S.
- primary-dms100 Northern Telecom DMS-100 switch type for the U.S.
- primary-dpnss DPNSS switch type for Europe
- primary-net5 NET5 switch type for UK, Europe, Asia and Australia
- primary-ni National ISDN Switch type for the U.S.
- primary-ni2c The Cisco NAS-SC switchtype based on NI2C.
- primary-ntt NTT switch type for Japan
- primary-qsig QSIG switch type
- primary-ts014 TS014 switch type for Australia (obsolete)

Step 2 - Configure the controller card for PRI:

Syntax:

```
pri-group timeslots <time slot range>
```

Timeslot range for T1

- *Full T1 (23 channels, 1 channel used for signaling (#24)) is always "1-24" as a default*
 - *Example: pri-group timeslots 1-24*
- *Sub-T1 (22 channels or less, 1 channel used for signaling (#24))*
 - *Example 12 channels starting at channel 1: pri-group timeslots 1-12,*
 - *Example 10 channels starting at channel 11: : pri-group timeslots 11-20*

Timeslot range for E1

- *Full E1 (30 channels, 1 channel used for signaling (#16)) is always "1-31" as a default*
 - *Example: pri-group timeslots 1-31*

- *Sub E1 (29 channels or less where range does not include signaling channel #16)*
 - *Example 12 channels starting at channel 1: pri-group timeslots 1-12,*
 - *Example 5 channels started at channel 6: pri-group timeslots 6-10*
- *Sub E1 (29 channels or less where range includes signaling channel #16)*
 - *Example 20 channels starting at channel 1: pri-group timeslots 1-21*
 - *Example 10 channels starting at channel 11: pri-group timeslots 11-21*

Example PRI/T1:

```
controller T1 1/2/0  
pri-group timeslots 1-24
```

Example PRI/E1:

```
controller E1 0/1/0  
pri-group timeslots 1-31
```

Once the controllers are configured, the router will automatically create a serial interface and a voice port for each controller:

Example :

```
interface Serial0/1/0:23  
!  
voice-port 0/1/0:23
```

Step 3 : Additional Commands to add to the Serial Interface:

```
isdn protocol-emulate <network OR user> *** Information provided by  
customer**
```

Step 4 : Add the interdigit timeout command to the voice port :

Syntax :
timeouts interdigit 5

Example :

```
voice-port 0/2/0:0  
timeouts interdigit 5
```

E1 PRI Sample Configuration :

Example for NIM-1MFT-T1/E1 installed in card slot 0, card bay 1.

```
card type e1 0 1  
  
network-clock synchronization automatic  
network-clock input-source 1 controller e1 0/1/0  
!  
isdn switch-type primary-5ess  
!  
controller E1 0/1/0  
clock source line primary  
pri-group timeslots 1-31  
!  
interface Serial 0/1/0:15  
no ip address  
encapsulation hdlc  
isdn switch-type primary-5ess  
isdn protocol-emulate network  
isdn skipsend-idverify  
  
!  
voice-port 0/1/0:15  
timeouts interdigit 5
```

NOTE - Additional instructions for NIM-XCE1T1-PRI cards:

When configuring the pri-group with a NIM-XCE1T1-PRI card, the option keyword **voice-dsp** must be configured after the timeslot range is defined.

Example PRI with E1:

```
controller E1 0/1/0  
pri-group timeslots 1-31 voice-dsp
```

2.3.2 Analog Voice Ports

Analog voice ports do not require as much configuration as digital ports.

2.3.2.1 FXS

Sample analog FXS port configuration :

```
voice-port 0/3/0
timeouts interdigit 5
station-id name George
station-id number 12013982000
```

2.3.2.2 E&M

For E&M port configuration (which is connected to an analog port on a PBX), the following settings must be configured. These settings must match how the PBX is configured.

operation <2-wire> or <4-wire>

signal <wink-start> or <immediate-start> or <delay-dial>

type <1> or <2> or <3> or <5>

Sample E&M port configuration:

```
voice-port 0/3/0
description E&M port connecting to <fill in description>
timeouts interdigit 5    **Always set to a value of 5**
operation 2-wire
signal wink-start
type 2
station-id name George
station-id number 12013982000
```


2.4 Configuring SIP for TDM PBX

The following configuration is required to configure a router for SIP signaling for use with the AT&T IP Border Elements. **A loopback interface must be configured on the TDM Gateway. The loopback interface must be configured with either the AT&T provided or customer supplied AT&T signaling IP address. If AT&T provided, this address can be found in the "Customer Router Configuration and VQM Shipping Confirmation" letter and is referenced as the IP signaling address.**

In the following sample router configuration, the loopback 0 interface has been configured with the signaling IP address of 12.23.44.27.

Sample Router Configuration:

```
interface Loopback 1
  Ip address 12.23.44.27 255.255.255.255
!
voice service voip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
  bind control source-interface Loopback1
  bind media source-interface Loopback1
  rel1xx disable
  min-se 900 session-expires 900
  sip-profiles 1
!
voice class sip-profiles 1
  request REINVITE sdp-header Attribute modify "a=T38FaxFillBitRemoval:0" "a=Tool:
  GW"
  request INVITE sdp-header Audio-Attribute add "a=ptime:30"
  request REINVITE sdp-header Audio-Attribute add "a=ptime:30"
  response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
  response ANY sdp-header Audio-Attribute add "a=ptime:30"
!
sip-ua
  retry invite 2
  retry notify 2
```

```
retry register 10  
timers notify 100
```

Note: For IPv6, make the following modification to the above configuration. In the following sample configuration, the loopback 1 interface has been configured with the signaling IPv6 address of 2001:1818:16:22::1.

```
interface Loopback1  
  ipv6 address 2001:1818:16:22::1/64 **Each TDM GW gets a unique IPv6 address**
```

2.5 Configuring Dial Peers for TDM Gateway

This section describes how to setup dial peers to work with AT&T IP Flexible Reach Service and/or AT&T IP Toll-Free on AT&T VPN as the Underlying Transport Service. A voice class codec is defined first and then is applied to the appropriate dial peers.

2.5.1 Voice Class Codec

A voice class codec can be used to provide a list of codecs with preferences which the dial peers will refer to. The codec with the lower preference number has the highest preference. For example, preference 1 has a higher preference than preference 2.

```
voice class codec 1  
  codec preference 1 g729br8 bytes 30  
  codec preference 2 g729r8 bytes 30  
  codec preference 3 g711ulaw
```

2.5.2 VoIP Dial Peers for IP Long Distance

VoIP Dial Peers are required for outbound calls (to the AT&T IP Border Elements) and for inbound calls (calls received from the AT&T IP Border Elements). These Dial Peers terminate the VoIP leg of the call.

Incoming calls from the AT&T IP Border Element (IPBE) will use the following digit format:

5 zeros for guiding digits + pbx extension prefix (optional up to 6 digits) + desired number of phone digits

Additional Notes:

- 21 is the total number of digits that the network can deliver to the router. The number of guiding digits + number of PBX extension digits + number of desired phone digits must be less than or equal to 21
- For wildcard dialing, guiding digits will not be signaled.
- The desired number of digits is decided by the PBX extension length.

Following is an example of the digits forwarded by the AT&T IP Border Element with a 7 digit extension:

000004912234 (5 zeros for guiding digits + 7 digit extension)

The incoming dial peer must be configured to match the digits sent by the AT&T IPBE.

If unsure of the format of the digits coming in from the AT&T IPBE, turn on the “debug ccapi inout” command on the router and initiate an inbound call. (Note: It is recommended to turn on debugs during off hours). With this debug, it is possible to view the digits that the AT&T IPBE is sending. This debug will also show if Dial Peers are matching on those digits. Two Dial Peers should be matched. The first should be a VoIP dial peer (to properly terminate the call) and the second is a POTS dial peer (which points to the appropriate digital/analog port).

The AT&T IPBE addresses to be configured in the dial peers, can be obtained from the **“Customer Router Configuration and VQM Shipping Confirmation” letter and is referenced as the AT&T IPBE addresses.**

If a customer needs to enable compressed RTP (cRTP), it is required to configure “no vad” under all VoIP dial peers (outgoing and incoming dial peers). By default, the VAD (Voice Activity Detection) is enabled on all dial peers.

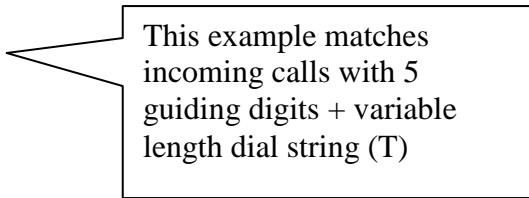
Example configuration:

```
dial-peer voice 1110 voip
description Outgoing Dial Peer To Border Element #1 for US calls
preference 1
destination-pattern 1.....
rtp payload-type nse 99
```

```
rtp payload-type nte 100
session protocol sipv2
session target ipv4:X.X.X.X <Insert IP address of Border Element #1>
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48
!
dial-peer voice 1111 voip
description Outgoing Dial Peer To Border Element #2 for US Calls
preference 1
destination-pattern 1.....
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
session target ipv4:X.X.X.X <Insert IP address of Border Element #2>
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48
!
dial-peer voice 1112 voip
description Outgoing Dial Peer To Border Element #1 for International calls
preference 1
destination-pattern 011T
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
session target ipv4:X.X.X.X <Insert IP address of Border Element #1>
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
```

```
fax rate 14400 bytes 48
!
dial-peer voice 1113 voip
description Outgoing Dial Peer To Border Element #2 for International Calls
preference 1
destination-pattern 011T
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
session target ipv4:X.X.X.X <Insert IP address of Border Element #2>
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48

dial-peer voice 1651 voip
description Incoming Dial Peer From Border Element
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
incoming called-number 00000T
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48
!
dial-peer hunt 1 <required to round robin between Border Elements #1 and #2>
```



Note: For IPV6, make the following modification to the above configuration. The Border Element IP addresses must be changed to IPv6 addresses.

```
session target ipv6:[X:X:X:X:X:X:X] <Insert IP address of Border Element>
```

2.5.3 Additional VoIP Dial Peers for IP Local

```
dial-peer voice 1511 voip
description Outgoing Dial Peer to Border Element #1 for N11 calling
preference 1
destination-pattern [2-9]11
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
session target ipv4:X.X.X.X <Insert IP address of Border Element #1>
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48
!
dial-peer voice 1512 voip
description Outgoing Dial Peer to Border Element #2 for N11 calling
preference 1
destination-pattern [2-9]11
rtp payload-type nse 99
rtp payload-type nte 100
voice-class codec 1
session protocol sipv2
session target ipv4:X.X.X.X <Insert IP address of Border Element #2>
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48
```

Note: For IPV6, make the following modification to the above configuration. The Border Element IP addresses must be changed to IPv6 addresses.

```
session target ipv6:[X:X:X:X:X:X:X] <Insert IP address of Border Element>
```

2.5.4 Additional Dial Peers for Private Dial Plans

```
dial-peer voice 1611 voip
description Outgoing Dial Peer to Border Element#1 for Private Dialing
```

```
preference 1
destination-pattern [2-9]T
rtp payload-type nse 99
rtp payload-type nte 100
voice-class codec 1
session protocol sipv2
session target ipv4:X.X.X.X <Insert IP address of Border Element #1>
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400 bytes 48
!
dial-peer voice 1612 voip
description Outgoing Dial Peer to Border Element#2 for Private Dialing
preference 1
destination-pattern [2-9]T
rtp payload-type nse 99
rtp payload-type nte 100
voice-class codec 1
session protocol sipv2
session target ipv4:X.X.X.X <Insert IP address of Border Element #2>
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400 bytes 48
```

Note: For IPV6, make the following modification to the above configuration. The Border Element IP addresses must be changed to IPv6 addresses.

```
session target ipv6:[X:X:X:X:X:X:X] <Insert IP address of Border Element>
```

2.5.5 POTS Dial Peers

As noted in the previous section, Incoming calls from the AT&T IP Border Element (IPBE) will use the following digit format:

5 zeros for guiding digits + pbx extension prefix (optional up to 6 digits) + desired number of phone digits

The desired number of digits is decided by the PBX extension length.

Following are examples of how to configure the POTS dial peers. POTS dial peers terminate the telephony leg of the call.

```
dial-peer voice 1652 pots
description Dial Peer for FXS analog port
answer-address 7325552000 <E.164 address>
destination-pattern 000002000 <this example is for 5 guiding zeros + 4 digit extension
of 2000>
port 0/3/0
!
dial-peer voice 1671 pots
description Dial Peer for T1 digital port
destination-pattern 00000.... <this example for 5 guiding zeros +4 digit extension >
port 0/1/0:15
```

2.6 Additional Configuration for AT&T IP Toll-Free

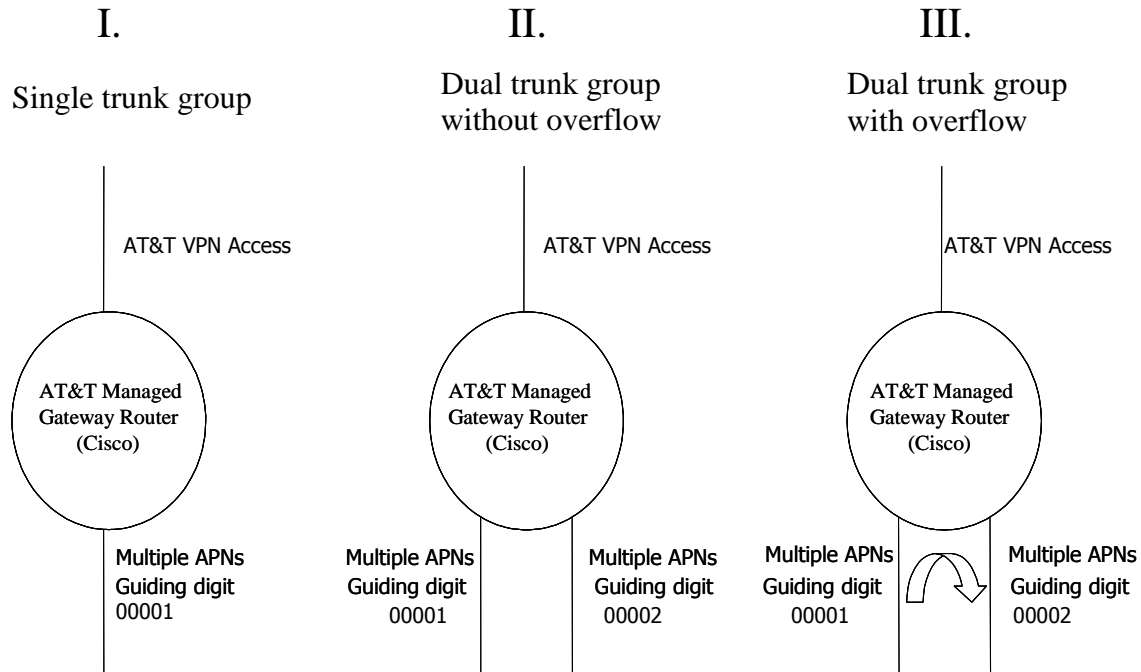
AT&T IP Toll-Free service supports the configuration of different APNs (Action Point Numbers). An APN is similar to a Numbering Plan Area. These APNs can be configured by AT&T to allow the Customer to route different 8YY numbers to different trunk groups on the TDM Gateway site. Additionally, the Customer will have limited flexibility to overflow between the trunk groups.

The three static TDM trunk group arrangements that a customer can choose from are:

- 1) Single trunk group
- 2) Dual trunk group* without overflow
- 3) Dual trunk group* with overflow (overflow in one direction between the trunks).

*Note that a dual trunk group can be setup on either a single or multiple T1 connections to the Customer's TDM PBX.

The Customer can define the number of digits out-pulsed by AT&T to be between 0 and 10 digits. The AT&T network will prefix the out-pulsed digits with guiding digits. A single trunk group will receive guiding digits of "00001". Dual trunk groups will receive guiding digits of "00001" and "00002". See diagram below for an illustration of trunk group arrangements and their corresponding guiding digits:



Codecs supported for AT&T IP Toll-Free service on AT&T VPN service include:

- g729br8 bytes 30
- g729r8 bytes 30
- g711ulaw

It is recommended to configure a “voice-class codec” to the voip dial peers as described in section 2.9.1.

2.6.1 Single Trunk Group Configuration

The following incoming dial peers must be added to the router’s TDM Gateway configuration for AT&T IP Toll-Free service. As previously stated, a single trunk group configuration will send guiding digits of “00001”. Therefore, dial-peer voice 1501 is added in the following sample configuration to match on those guiding digits. These configurations assume the CAS and PRI configurations are already in place (see sections 2.5 and 2.6).

Sample Configuration for Single Trunk Group Configuration

```
dial-peer voice 1501 voip
description Incoming Dial Peer From Border Element
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
incoming called-number 00001T
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48

!
dial-peer voice 1650 pots
destination-pattern 00001T
port 0/1/0:0 **Points to the appropriate voice port – CAS or PRI**!
```

2.6.2 Dual Trunk Group

With the dual trunk group option, the AT&T network can send incoming calls with guiding digits of 00001 or 00002. An example of incoming call digits could be 000019143975000 or 000029143976000. The router can handle the calls differently based on these incoming digits.

Note that in all dual trunk group sample configurations shown, the dial peer 1501 is added to match on guiding digits of “00001” and “00002”.

The customer can choose between dual trunk groups without overflow or with overflow. The following examples will illustrate how to configure these two options with CAS and PRI ports.

2.6.2.1 Dual Trunk Group Without Overflow

This sample configuration is for a single T1 CAS to the customer PBX with 2 trunk groups. Trunk Group 1 has 10 channels and Trunk Group 2 has 14 channels.

Sample Configuration for Dual Trunk Group without Overflow – CAS port

```
controller T1 0/1/0
framing esf
linecode b8zs
ds0-group 0 timeslots 1-10 type e&m-wink-start           ! 10 channels in one trunk group
ds0-group 1 timeslots 11-24 type e&m-wink-start          ! 14 channel in 2nd trunk group
!
```

```
!  
voice-port 0/1/0:0  
timeouts interdigit 2  
music-threshold -70  
busyout monitor Serial0/0/0  
!  
voice-port 0/1/0:1  
timeouts interdigit 2  
music-threshold -70  
busyout monitor Serial0/0/0  
!  
!  
dial-peer voice 1501 voip  
description Incoming Dial Peer From Border Element – Trunk Group 1 & 2  
rtp payload-type nse 99  
rtp payload-type nte 100  
session protocol sipv2  
incoming called-number 0000[1,2]T  
dtmf-relay rtp-nte  
voice-class codec 1  
fax-relay sg3-to-g3  
fax rate 14400 bytes 48  
!  
!  
dial-peer voice 1650 pots  
destination-pattern 00001T  
port 0/1/0:0  
!  
dial-peer voice 1651 pots  
destination-pattern 00002T  
port 0/1/0:1  
!  
!
```

Trunk Group 1

Trunk Group 2

This dial peer configured to match on guiding digits of 00001. Points to voice port 0/1/0:0 (Trunk Group 1)

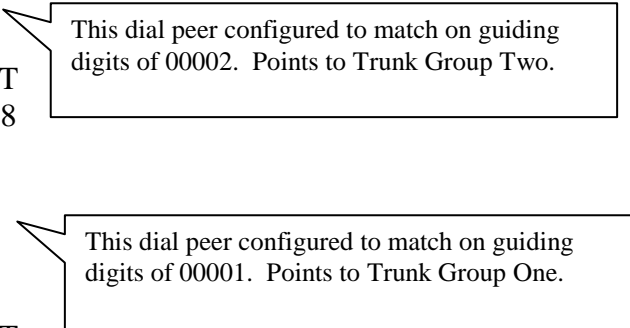
This dial peer configured to match on guiding digits of 00002. Point to voice port 0/1/0 :1 (Trunk Group 2)

Sample Configuration of Dual Trunk Group Without Overflow – PRI port:

```
trunk group TG-One  
description "IP Toll-Free DS0s - Critical Use"  
hunt-scheme least-used  
!  
trunk group TG-Two  
description "IP Toll-Free DS0s - Normal Use"  
hunt-scheme least-used  
!  
controller T1 0/1/0  
pri-group timeslots 1-24  
trunk-group TG-One timeslots 1-21
```

T1 PRI port configured into Trunk Group One and Two based on timeslots.

```
trunk-group TG-Two timeslots 22-23
!
dial-peer voice 1501 voip
description Incoming Dial Peer From Border Element – Trunk Group 1 & 2
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
incoming called-number 0000[1,2]T
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48
!
dial-peer voice 1650 pots
trunkgroup TG-Two
preference 1
destination-pattern 00002T
progress_ind alert enable 8
direct-inward-dial
!
dial-peer voice 1651 pots
trunkgroup TG-One
preference 1
destination-pattern 00001T
progress_ind alert enable 8
direct-inward-dial
!
```



2.6.2.2 Dual Trunk Group with overflow

In the dual trunk group with overflow setup, if the first trunk group is fully used, the next incoming call will terminate on the channels assigned to second trunk group.

The following sample configuration illustrates how to setup dual trunk with overflow on a CAS port. In this setup, the customer is using CAS signaling and has 10 channels in trunk group 1 and 14 channels in trunk group 2. The trunk group called TG-Two (Trunk Group Two) will overflow into channels defined for TG-One.

Sample Configuration for Dual Trunk Group with overflow – CAS port

```
controller T1 0/1/0
framing esf
linecode b8zs
ds0-group 0 timeslots 1-10 type e&m-wink-start *** 10 channels in Trunk Group 1***
```

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```
ds0-group 1 timeslots 11-24 type e&m-wink-start ***14 channel in Trunk Group 2***
!
!
voice-port 0/1/0:0
timeouts interdigit 2
music-threshold -70
trunk-group TG-One
!
voice-port 0/1/0:1
timeouts interdigit 2
music-threshold -70
trunk-group TG-Two
!
! Then we have 2 inbound pots (from the PBX) dial peers – one for each trunk group
!
dial-peer voice 1501 voip
description Incoming Dial Peer From Border Element – Trunk Group 1 & 2
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
incoming called-number 0000[1,2]T
dtmf-relay rtp-nte
voice-class codec 1
fax-relay sg3-to-g3
fax rate 14400 bytes 48
!
!
dial-peer voice 1650 pots
destination-pattern 00001T
trunkgroup TG-One
!
dial-peer voice 1651 pots
destination-pattern 00002T
trunkgroup TG-One
trunkgroup TG-Two
```

Trunk Group 1

Trunk Group 2

This dial peer configured to match on guiding digits of 00001. Points to Trunk Group 1 only.

This dial peer configured to match on guiding digits of 00002 only. Points to both Trunk Group One and Two (used for overflow).

The following sample configuration illustrates how to setup dual trunk with overflow on a PRI port. In this setup, the Customer is using T1 port with PRI signaling and has 21 channels in trunk group 1 and 2 channels in trunk group 2. The trunk group called TG-Two will overflow into channels defined for TG-One.

Sample Configuration of Dual Trunk Group with Overflow – PRI port:

```
trunk group TG-One
```

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```
description "IP Toll-Free DS0s - Critical Use"
hunt-scheme least-used
!
trunk group TG-Two
description "IP Toll-Free DS0s - Normal Use"
hunt-scheme least-used
!
controller T1 0/1/0
  pri-group timeslots 1-24
  trunk-group TG-One timeslots 1-21
  trunk-group TG-Two timeslots 22-23
!
voice-port 0/1/0:23
music-threshold -70
busyout monitor Serial 0/0/0
busyout action shutdown
!
dial-peer voice 1501 voip
description Incoming Dial Peer From Border Element – Trunk Group 1 & 2
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
incoming called-number 0000[1,2]T
dtmf-relay rtp-nte
voice-class codec 1
  fax-relay sg3-to-g3
  fax rate 14400 bytes 48
!
!
dial-peer voice 1650 pots
trunkgroup TG-Two 1
destination-pattern 00002T
!
dial-peer voice 1651 pots
trunkgroup TG-One
trunkgroup TG-Two
destination-pattern 00001T
!
!
```

T1 PRI port configured into two trunk group based on timeslots.

This dial peer configured to match on guiding digits of 00002. Points to Trunk Group Two.

This dial peer configured to match on guiding digits of 00001 only. Points to both Trunk Group One and Two (used for overflow).

2.7 Additional Configuration for Enhanced Features Service

AT&T IP Flexible Reach Service with Enhanced Features is supported for TDM PBX. Additional configuration must be added to the TDM Gateway in order to utilize this service.

2.7.1 Configure Dial Peers to Utilize Feature Access Codes

Additional dial peers must be added to the TDM Gateway to utilize Feature Access Codes (FAC) that are used to activate and deactivate features by the Enhanced Features subscriber (i.e. end user). FAC will only work with PRI interfaces; CAS interfaces are supported for Enhanced Features, but do not support FAC.

Example configuration:

```
dial-peer voice 120 voip
preference 1
destination-pattern [*][0-9]T
session target ipv4:<Insert IP address of Border Element #1>
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
dtmf-relay rtp-nte
fax rate 14400
voice-class codec 1

dial-peer voice 130 voip
preference 1
destination-pattern [*][0-9]T
session target ipv4: <Insert IP address of Border Element #2>
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
dtmf-relay rtp-nte
fax rate 14400
voice-class codec 1
```

Note: For IPV6, make the following modification to the above configuration. The Border Element IP addresses must be changed to IPv6 addresses.

```
session target ipv6:[X:X:X:X:X:X:X] <Insert IP address of Border Element>
```

2.7.2 Configure Caller ID

Caller ID must be configured properly in order to utilize AT&T IP Flexible Reach Service with Enhanced Features. The TDM Gateway must be setup to use a valid Telephone Number that is provisioned for that site as the Caller ID. If a site is setup for IP Long Distance only, the Caller ID should be set to the PBX Telephone Number (which is provided by the customer during the service ordering process).

2.7.2.1 Analog Ports

Caller ID can be configured on analog ports using the following command under the voice port. Example is shown below:

```
voice-port 0/3/0
station-id number 12013982000
```

2.8 Additional Configuration for Modems

Modem calls are supported with AT&T's IP Flexible Reach service using the G.711 codec. It is recommended to configure separate G.711 inbound and outbound dial peers to support the modem calls. For inbound modem calls, inbound dial peers should be created for each modem TN or in some cases pattern-matching can be used to cover a broader range of TNs. For outbound modem calls, outbound dial peers should be created for the specific TNs that will be dialed or in some cases pattern-matching can be used to cover a broader range of TNs.

Outbound dial-peer example to specific destination TNs: Following is an example for outbound calls to a bank of modems with a destination TN range of 732-333-2210 to 732-333-2219.

```
dial-peer voice 2000 voip
description outgoing voice call to AT&T facing AT&T network
destination-pattern 1732333221[1-9]
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
session target ipv4:135.25.29.74 <IP BE #1 IP address>
dtmf-relay rtp-nte
codec g711ulaw
no vad
!
dial-peer voice 2001 voip
description outgoing voice call to AT&T facing AT&T network
destination-pattern 1732333221[1-9]
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
```


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```
session target ipv4:135.25.29.84 <IP BE #2 IP address>
dtmf-relay rtp-nte
codec g711ulaw
no vad
```

Outbound dial-peer example to multiple destination TNs: Following is an example for outbound calls to multiple destinations from a modem TN of 7326520003. In this example, the G.711 dial peer will be chosen based on the CALLING FROM TN (7326520003).

```
!
voice translation-rule 100
rule 1 /^/ /11111/
!
voice translation-rule 101
rule 1 /^11111\(.*$\)/ /\1/
!
voice translation-profile MODEM
translate called 100
!
voice translation-profile MODEM_STRIP
translate called 101
!
dial-peer voice 2000 pots
translation-profile incoming MODEM
answer-address 7326520003
direct-inward-dial
port 0/1/0:23
!
dial-peer voice 2001 voip
translation-profile outgoing MODEM_STRIP
destination-pattern 11111T
translate-outgoing calling 1
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
session target ipv4:12.194.172.10 <IP BE #1 IP address>
dtmf-relay rtp-nte
codec g711ulaw
fax rate 14400
!
dial-peer voice 2002 voip
translation-profile outgoing MODEM_STRIP
destination-pattern 11111T
translate-outgoing calling 1
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
session target ipv4:12.194.176.10 <IP BE #2 IP address>
```

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```
dtmf-relay rtp-nte  
codec g711ulaw  
fax rate 14400
```

Inbound dial peer example with FXS port: Following is an example for inbound calls to modems with TNs of 732-555-6671 and 732-555-6672 (assuming a 4 digit extension).

```
!  
dial-peer voice 1670 voip  
description Incoming Dial Peer From Border Element  
rtp payload-type nse 99  
rtp payload-type nte 100  
session protocol sipv2  
incoming called-number 00000667[1-2]  
dtmf-relay rtp-nte  
codec g711ulaw  
!  
dial-peer voice 1800 pots  
answer-address 7325556671  
destination-pattern 000006671  
port 0/3/0  
!  
dial-peer voice 1801 pots  
answer-address 7325556672  
destination-pattern 000006672  
port 0/3/1
```

Inbound dial peer example with T1 PRI port: Following is an example for inbound calls to modems with a TN range of 732-555-4410 and 732-555-4419 (assuming a 4 digit extension).

```
dial-peer voice 1670 voip  
description Incoming Dial Peer From Border Element  
rtp payload-type nse 99  
rtp payload-type nte 100  
session protocol sipv2  
incoming called-number 00000441.  
dtmf-relay rtp-nte  
codec g711ulaw  
!  
dial-peer voice 1671 pots  
description Dial Peer for T1 digital port  
destination-pattern 00000441.
```

```
forward-digits 4  
port 0/1/0:15
```

Note: When adding new inbound/outbound modem TNs to an existing configuration: Outbound dial peers may need to be updated when new outbound dialed TNs are added if not already covered by existing outbound dial peer configuration. Inbound dial peers may also need to be updated when new modem TNs are added to the IP Flexible Reach site if not already covered by existing inbound dial peer configuration. Please make sure to check the existing dial peers to determine if additional configuration needs to be applied to the router.

Note: For IPV6, make the following modification to the above configuration. The Border Element IP addresses must be changed to IPv6 addresses.

```
session target ipv6:[X:X:X:X:X:X:X] <Insert IP address of Border Element>
```

2.9 Routing Configuration

The loopback interface for signaling and media must be distributed on the CER via the BGP network statement:

Following is sample information on how to configure the routing for an IPV6 signaling/media address.

```
interface Loopback1  
  ipv6 address X:X:X:X:X:X:/64  **Each TDM GW gets a unique IPv6 address**  
  
router bgp <Customer AS number>  
  bgp router-id <Router-ID Number>  
  no bgp default ipv4-unicast  
  bgp log-neighbor-changes  
  neighbor <PER IPv4 address> remote-as <remote AS>  
  neighbor <PER IPv6 address> remote-as <remote AS>  
  !  
  address-family ipv4  
  neighbor <PER IPv4 address> activate  
  no auto-summary  
  no synchronization  
  exit-address-family
```

```
address-family ipv6
network <Enter Loopback 1 network> /64
neighbor <PER IPv6 address> activate
no synchronization
exit-address-family
```

Example:

```
interface Loopback1
  ipv6 address 2001:1818:16:22::1/64 **Each TDM GW gets a unique IPv6 address**

router bgp 65000
  bgp router-id 192.168.6.12
  no bgp default ipv4-unicast
  bgp log-neighbor-changes
  neighbor 2001:1890:8E00:C::1 remote-as 13979
  neighbor 192.168.150.5 remote-as 13979
!
address-family ipv4
  neighbor 192.168.150.5 activate
  no auto-summary
  no synchronization
exit-address-family
!
address-family ipv6
  network 2001:1818:16:22::/64 (IPV6 Network for Loopback 1)
  neighbor 2001:1890:8E00:C::1 activate
  no synchronization
exit-address-family
!
```

Appendix A: Sample TDM Gateway Configuration

```
Building configuration...

Current configuration : 10253 bytes
!
! Last configuration change at 16:24:57 EDT Tue May 19 2015
!
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
no platform punt-keepalive disable-kernel-core
platform hardware throughput level 100000
!
hostname 4331-Brig
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.13.02.S.154-3.S2-
ext.SPA.bin
boot system bootflash:isr4300-universalk9.03.13.01.S.154-3.S1-
ext.SPA.bin
boot-end-marker
!
!
!
card type e1 0 1
enable password 7 110A1016141D
!
no aaa new-model
clock timezone EST -5 0
clock summer-time EDT recurring
no ip source-route
!
!
!
!
!
!
!
```

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```
no ip domain lookup
!
subscriber templating
multilink bundle-name authenticated
!
!
!
!
!
isdn switch-type primary-5ess
!
!
trunk group PSTN
  hunt-scheme longest-idle
!
!
voice service pots
  no direct-inward-dial isdn
!
voice service voip
  ip address trusted list
    ipv4 135.25.29.74 255.255.255.255
    ipv4 192.168.0.50 255.255.255.255
    ipv4 135.16.206.41 255.255.255.255
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback
  pass-through g711ulaw
  sip
    bind control source-interface Loopback2
    bind media source-interface Loopback2
    relxx disable
    min-se 900 session-expires 900
    sip-profiles 1
  !
voice class codec 1
  codec preference 1 g729br8 bytes 30
  codec preference 2 g729r8 bytes 30
  codec preference 3 g711ulaw
!
!
!
voice class sip-profiles 3
  request REINVITE sdp-header Attribute modify
"a=T38FaxFillBitRemoval:0" ""
  request INVITE sdp-header Audio-Attribute add "a=ptime:30"
  request REINVITE sdp-header Audio-Attribute add "a=ptime:30"
  response ANY sdp-header Audio-Attribute modify "a=ptime:20"
"a=ptime:30"
  response ANY sdp-header Audio-Attribute add "a=ptime:30"
!
!
voice-card 0/1
!
```

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```
!  
license udi pid ISR4331/K9 sn FDO183806MP  
license accept end user agreement  
license boot level uck9  
!  
username vinny privilege 15 secret 5 $1$vaq0$a65vyjFfeK0bDhqGb5A0d1  
username cisco password 7 01100F175804  
!  
redundancy  
mode none  
!  
controller E1 0/1/0  
clock source line primary  
pri-group timeslots 1-10,16  
!  
!  
!  
!  
interface Loopback2  
ip address 135.16.206.66 255.255.255.255  
!  
interface GigabitEthernet0/0/0  
description - (WAN) Trunk To South-Station  
mtu 1514  
no ip address  
load-interval 30  
negotiation auto  
hold-queue 4096 in  
hold-queue 4096 out  
!  
interface GigabitEthernet0/0/0.2704  
description - Gig-E Link To SFCHXRPE4 - Gog0/10/0/7.2704  
encapsulation dot1Q 2704  
ip address 195.18.34.117 255.255.255.252  
ipv6 address 2001:506:15:274::1/64  
no cdp enable  
!  
interface GigabitEthernet0/0/1  
description - (LAN) Trunk To Metropark  
no ip address  
load-interval 30  
speed 1000  
no negotiation auto  
hold-queue 1024 in  
hold-queue 1024 out  
!  
interface GigabitEthernet0/0/1.86  
description - LAN subinterface  
encapsulation dot1Q 86  
ip address 10.2.86.1 255.255.255.0  
no cdp enable
```

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```
h323-gateway voip interface
h323-gateway voip bind srcaddr 10.2.86.1
!
!
interface GigabitEthernet0/0/2
no ip address
shutdown
negotiation auto
!
interface Service-Engine0/1/0
!
interface Serial0/1/0:15
encapsulation hdlc
isdn switch-type primary-5ess
isdn protocol-emulate network
isdn skipsend-idverify
no cdp enable
!
!
interface Service-Engine0/2/0
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
router bgp 65000
bgp router-id 192.168.0.174
bgp log-neighbor-changes
neighbor 2001:506:15:274::2 remote-as 13979
neighbor 195.18.34.118 remote-as 13979
!
address-family ipv4
network 10.2.86.0 mask 255.255.255.0
network 135.16.206.66 mask 255.255.255.255
network 172.48.128.0 mask 255.255.192.0
network 192.168.0.174 mask 255.255.255.255
no neighbor 2001:506:15:274::2 activate
neighbor 195.18.34.118 activate
neighbor 195.18.34.118 allowas-in
exit-address-family
!
address-family ipv6
network 2001:506:16:100::174/128
network 2001:506:16:174::/64
neighbor 2001:506:15:274::2 activate
neighbor 2001:506:15:274::2 allowas-in
exit-address-family
```


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```
!  
ip forward-protocol nd  
no ip http server  
no ip http secure-server  
ip route 10.1.55.0 255.255.255.0 10.10.199.20  
!  
!  
!  
!  
control-plane  
!  
!  
voice-port 0/1/0:15  
!  
!  
!  
no mgcp package-capability res-package  
no mgcp timer receive-rtcp  
mgcp behavior rsip-range tgcp-only  
mgcp behavior comedia-role none  
mgcp behavior comedia-check-media-src disable  
mgcp behavior comedia-sdp-force disable  
!  
mgcp profile default  
!  
!  
dial-peer voice 1110 voip  
  description Outgoing Dial Peer To Border Element #1 for US calls  
  preference 1  
  destination-pattern 1.....  
  rtp payload-type nse 99  
  rtp payload-type nte 100  
  session protocol sipv2  
  session target ipv4:135.25.29.74  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  fax-relay sg3-to-g3  
  fax rate 14400 bytes 48  
!  
dial-peer voice 1111 voip  
  description Outgoing Dial Peer To Border Element #2 for US calls  
  preference 1  
  destination-pattern 1.....  
  rtp payload-type nse 99  
  rtp payload-type nte 100  
  session protocol sipv2  
  session target ipv4:135.25.30.74  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  fax-relay sg3-to-g3  
  fax rate 14400 bytes 48
```

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```
!  
dial-peer voice 1112 voip  
  description Outgoing Dial Peer To Border Element #1 for International  
  calls  
  preference 1  
  destination-pattern 011T  
  rtp payload-type nse 99  
  rtp payload-type nte 100  
  session protocol sipv2  
  session target ipv4:135.25.29.74  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  fax-relay sg3-to-g3  
  fax rate 14400 bytes 48  
!  
dial-peer voice 1113 voip  
  description Outgoing Dial Peer To Border Element #1 for International  
  calls  
  preference 1  
  destination-pattern 011T  
  rtp payload-type nse 99  
  rtp payload-type nte 100  
  session protocol sipv2  
  session target ipv4:135.25.30.74  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  fax-relay sg3-to-g3  
  fax rate 14400 bytes 48  
!  
dial-peer voice 1650 voip  
  description Outgoing Dial Peer to Border Element #1 for N11 calling  
  preference 1  
  destination-pattern [2-9]11  
  rtp payload-type nse 99  
  rtp payload-type nte 100  
  session protocol sipv2  
  session target ipv4:135.25.29.74  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  fax-relay sg3-to-g3  
  fax rate 14400 bytes 48  
!  
dial-peer voice 1650 voip  
  description Outgoing Dial Peer to Border Element #2 for N11 calling  
  preference 1  
  destination-pattern [2-9]11  
  rtp payload-type nse 99  
  rtp payload-type nte 100  
  session protocol sipv2  
  session target ipv4:135.25.30.74
```

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```
voice-class codec 1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400 bytes 48
!
dial-peer voice 1670 voip
description Incoming Dial Peer From Border Element
rtp payload-type nse 99
rtp payload-type nte 100
session protocol sipv2
incoming called-number 00000T
voice-class codec 1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400 bytes 48
no vad
!
dial-peer voice 1653 pots
description Dial Peer for Digital port
destination-pattern 00000.....
port 0/1/0:15
!
!
!
!
dial-peer hunt 1
gateway
timer receive-rtp 180
!
sip-ua
retry invite 2
retry notify 2
retry register 10
timers notify 100
!
!
line con 0
exec-timeout 300 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 600 0
login local
!
network-clock synchronization automatic
ntp source Loopback0
ntp server 135.16.205.66
!
end
```

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```
4331-Brig#term lenm_ _ 24  
4331-Brig#
```

Appendix C: Acronyms

Acronym	Translation
AT&T VPN	AT&T Virtual Private Network
CAS	Channel Associated Signaling
CCG	Customer Configuration Guide
CCS	Common Channel Signaling
CEF	Cisco Express Forwarding
CER	Customer Edge Router
CLI	Command Line Interface
COS	Class of Service
CPE	Customer Premise Equipment
CPU	Central Processing Unit
DID	Direct Inward Dial
DSP	Digital Signal Processors
DTMF	Dual Tone Multi Frequency
E&M	Ear & Mouth
FXO	Foreign Exchange Office
FXS	Foreign Exchange Station
GSM FR	Global System for Mobile communications Full Rate
HDV	High Density Voice
HWIC	High-speed WAN Interface Card
IETF	Internet Engineering Task Force
IOS	Internetwork Operation System
IP	Internet Protocol
IPBE	Internet Protocol Border Element
ISR	Integrated Services Router
ITU-T	International Telecommunication Union - Telecommunications
LAN	Local Area Network
LD	Long Distance
MOW	Most Of World
MTU	Maximum Transmission Unit
NIM	Network Interface Module
NM	Network Module
OAM	Operation Administration & Maintenance
PBX	Private Branch Exchange
POTS	Plain Old Telephone Service
PRI	Primary Rate Interface
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
PVDM	Packet Voice DSP Module
QSIG	Q Signaling
RC	Receive

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RFC	Request for Comment
RT	Real Time
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SIP	Session Initiation Protocol
TAC	Technical Assistance Center
TDM	Time Division Multiplexing
TN	Telephone Number
UDP	User Datagram Protocol
VAD	Voice Activity Detection
VNI	Voice Network Infrastructure
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
VQM	Voice Quality Monitor
WAN	Wide Area Network
WIC	WAN Interface Card

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