# 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 G2 Platform

**August 27, 2014**

**Version 1.11**

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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 G2 platforms as TDM gateways:

Routers supported:

- 2911
- 2921
- 2951
- 3925
- 3945
- 3945E

Voice cards supported:

VIC3-2E/M
VIC3-4FXS/DID
VIC2-4FXO
VIC2-2FXO
SM-NM-ADPTR
NM-HD-2V
NM-HDV2-1T1/E1
NM-HDV2-2T1/E1
VWIC2-1MFT-T1/E1
VWIC2-2MFT-T1/E1
VWIC2-1MFT-G703
VWIC2-2MFT-G703
VWIC3-1MFT-T1/E1
VWIC3-2MFT-T1/E1
VWIC3-4MFT-T1/E1

PVDM cards supported:

PVDM2-ADPTR
PVDM2-8

PVDM2-32
PVDM2-16
PVDM2-48
PVDM3-32
PVDM3-16
PVDM3-48
PVDM3-64
PVDM3-128

### *1.2 Software Requirements*

Configurations in this guide were tested with Cisco IOS 15.2(1)T2ES and 15.3(3)M1ES.

The IOS files can be obtained from:
https://upload.cisco.com/cgi-bin/swc/fileexg/main.cgi?CONTYPES=ATT-Managed-Services

Note: CCO access is required to download these files.

**2900 routers:**

c2900-universalk9-mz.SSA-eng-sp-152-1.T2ES

c2900-universalk9-mz.SSA-eng-sp-153-3.M1.bin

**2951 router (only supported with 15.3(3)M1ES):**
c2951-universalk9-mz.SSA-eng-sp-153-3.M1.bin

**3925/45 routers:**

c3900-universalk9-mz.SSA-eng-sp-152-1.T2ES

c3900-universalk9-mz.SSA-eng-sp-153-3.M1.bin

**3945E router:**

c3900e-universalk9-mz.SSA-eng-sp-152-1.T2ES

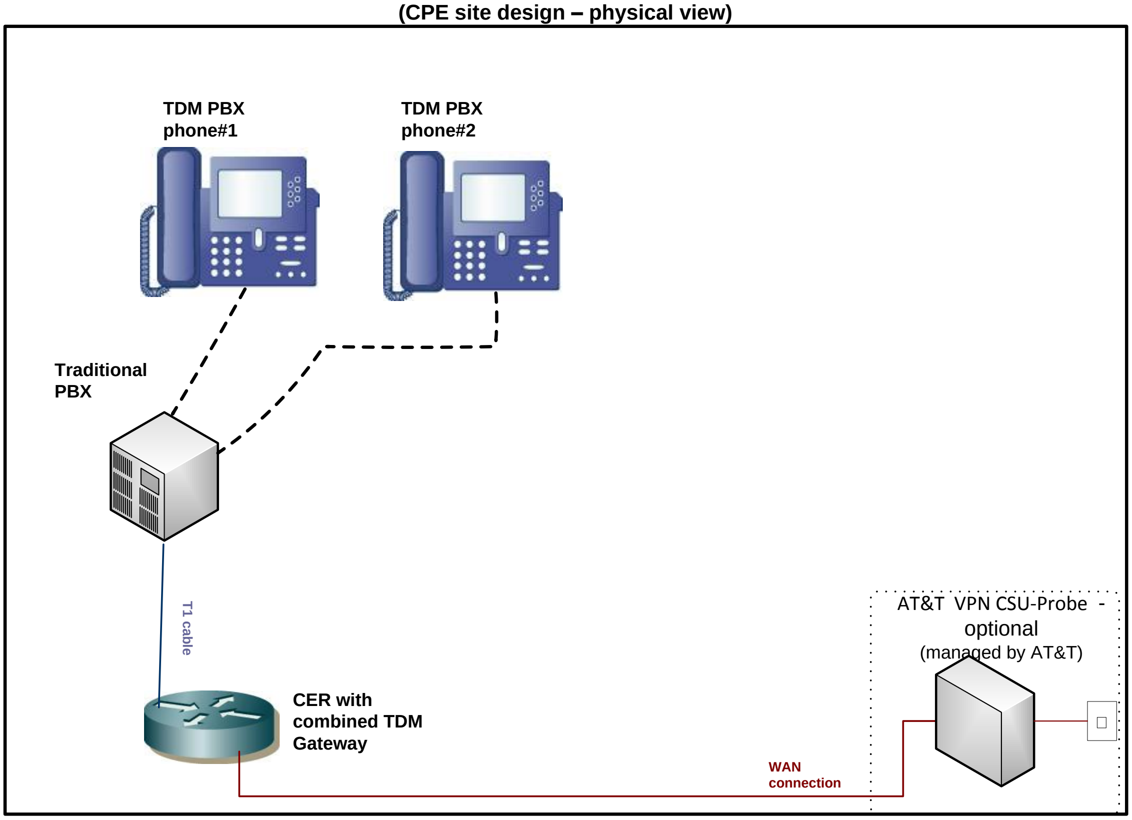
c3900e-universalk9-mz.SSA-eng-sp-153-3.M1.bin

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.

**AT&T BVoIP on AT&T VPN site**
**with VPN CSU-Probe, CER with combined TDM Gateway Router**
**(CPE site design – physical view)**




## *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, you will need to know the number of channels and the codec used.

The ISR G2's support both PVDM2-X and PVDM3-X series modules.

The following are unique characteristics of PVDM3-X:

- The PVDM3 can only be installed on the motherboard of the 29xx or 39xx platform. They cannot be installed on the NM-HDV2 NM.
- The PVDM3-X and PVDM2-X can coexist on the same chassis as long as they are not installed in the same domain (i.e. Motherboard or NM-HDV2).
- The PVDM3s can support a maximum of 64 G711 participants per conference session on any PVDM3-X and maximum of 16 G729 participants per PVDM3-16 and a maximum of 32 G729 participants per PVDM2-32 and above per conference session.

The following are the characteristics of PVDM2-X with ISR G2:

- The ISR G2 motherboard domain can contain either all PVDM2 modules or all PVDM3 modules.
- If a mix of PVDM2s and PVDM3s are detected on the motherboard slots, then the PVDM2s will be deactivated, allowing only the PVDM3s to be used actively
- PVDM2-X module requires a special adaptor cards (PVDM2- ADPTR) to be installed on the motherboard.
- The ISR G2 service module can take an NM-HDV2 with PVDM2-X modules using the network-to-server module adaptor card (SM-NM-ADPTR).

Please use the following link to determine the number of number of type of PVDMs modules to support the number of calls required.  You will need a CCO ID.

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

Note: G729 is a medium complexity codec and G.711 is a high complexity codec.

## *2.3 How to configure VWIC2-1MFT-T1/E1 or VWIC2-2MFT-T1/E1 modules using on- board DSPs*

Routers such as Cisco ISR 29x/39xx can have on-board DSPs and VWIC cards installed in the WIC slots.  By default, DSPs are shared by the cards in the WIC slots. Therefore, the on-board DSPs do not need to be configured with the "dspfarm" command.

Following are the steps to configure the VWIC2-1 or 2 MFT-T1/E1 modules:

**1) Configure the card for the appropriate type:**

**Syntax: card type {t1 | e1}** ***<slot #> <sub slot#>***

Example for T1-

```
card type t1 0 1
```

Example for E1-

```
card type e1 0 1
```

**2) Configure the "network clock participate" for the appropriate wic slot:**

**Syntax: network-clock-participate wic <WIC *slot* #> |**

Example:
For controller 0/1/0 use -

```
network-clock-participate wic 1
```

Note: For controller 0/1/1, the network-clock-participate command is not needed since it is also in wic1 slot.

If you don't use the above and define CAS or PRI channels under the controller card, the router gives an error with a message that network-clock-participate command must be used first.

### 3) <u>Configure the "network clock select":</u>

Specifiy a controller card to use for clocking.

**Syntax: network-clock-select 1 <T1 or E1> <0>/<WIC slot>/<port>**

Example:

```
network-clock-select 1 T1 0/1/0
```

### 4) <u>If dual port controller is used, set "clock source internal":</u>

For a dual port controller card, set the clocking internal on the 2nd port.

Example:

```
controller T1 0/1/1
 clock source internal
```

### 5) <u>Configure the channels under the controller card:</u>

Depending on the interface to the PBX being CAS or PRI, the following commands are used:

***CAS Syntax: ds0-group <ds0-group#> timeslots <time slot range> type <switch type>***

***PRI Syntax: pri-group timeslots <time slot range> <D channel>***

**Example of T1 CAS with 12 channels:**

```
controller T1 2/0
ds0-group 0 timeslots 1-12 type e&m-wink-start
```

Note: Be sure to set the proper CAS switch type to match the PBX. Choices 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 PRI/T1 with 12 channels:**

```
controller T1 2/0
pri-group timeslots 1-12,24
```

**Example PRI/E1 with 10 channels:**

```
controller E1 0/1/0
 pri-group timeslots 1-10,16
```

**6) <u>Verify voice ports are created.</u>**

Once the above is entered, the router will automatically create the following –

**CAS:**

```
voice-port 0/1/0:0
```

Add an interdigit timeout of 5 seconds to the voice port :

```
voice-port 0/1/0:0
  timeouts interdigit 5
```

**PRI:**

For PRI, a serial and voice-port interface will be created :

```
interface Serial0/1/0:23
  isdn switch-type primary-5ess
!
voice-port 0/1/0:23
```

Note: Be sure to set the proper PRI switch type to match the PBX. Choices 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)

Add an interdigit timeout of 5 seconds:

```
voice-port 0/1/0:23
 timeouts interdigit 5
```

Sample digital port configuration (T1 CAS connection to PBX. Contains 24 channels) :

```
network-clock-participate slot 2
network-clock-select 1 T1 2/0
!
voice-card 2
 dsp-farm
!
controller T1 2/0
 framing esf
 linecode b8zs
 no yellow generation
 no yellow detection
 ds0-group 0 timeslots 1-24 type e&m-wink-start
```

## 2.4 How to configure NM-HDV2 cards

NM-HDV2 cards are available in one port modules (NM-HDV2-1T1/E1) and two port modules (NM-HDV2-2T1/E1).

There are three methods to obtain DSP resources for the NM-HDV2 card:

1) **Recommended Method:** DSP resources can be installed in the PVDM banks directly on the NM-HDV2 circuit board. No additional router configuration is required for the NM-HDV2 to use these resources.
2) The NM-HDV2 card can acquire global DSP resources from the on-board PVDMs (installed on the router's mother board). The "dspfarm" command must be configured on the router to allow for the DSP resources to be shared.
3) DSP resources can be acquired from a combination of on-board PVDMs and from the local PVDMs on the NM-HDV2 card. Take for example, a Cisco 2821 with a NM-HDV2-2E1 with a PDVM2-64 (30 medium complexity calls) and an on-board PDVM2-48 (24 calls). By default 30 calls can be supported on the first E1 port. To support the additional 24 calls, the on-board DSP's need to be shared by explicitly configuring the **dspfarm** command.

**Steps to configure NM-HDV2-1 or 2T1/E1 with on-board DSPs:** The NM-HDV2-1 or 2 E1/T1 card needs to be defined with the type of either E1 or T1, before channels can be defined on a particular trunk. To change the definition from e1 to t1, enter "no card type e1 1 1", save and reload and reconfigure.

1) **Define card type**

**Syntax: card type** {**t1 | e1**} *<slot #>* *<subslot #>*

Example:

```
card type e1 1 1
```

2) **Configure "dspfarm" for NM-HDV2 to use global DSP resources.**

```
voice-card 0
 dspfarm
```

3) **Configure the "network clock participate" for the appropriate slot:**

**Syntax: network-clock-participate slot <*slot* #>**

Example:
For controller E1 1/0 use:

```
network-clock-participate slot 1
```

Note: For controller 1/1 , the network-clock-participate command is not needed since it is also in slot 1.

If you don't use the above and define CAS or PRI channels under the controller card, the router gives an error with a message that network-clock-participate command must be used first.

### 4) Configure the "network clock select":

**Syntax: network-clock-select 1 <T1 or E1> <slot#>/port#>**

Specifiy a controller card to use for clocking.

Example:

```
network-clock-select 1 T1 1/0
```

### 5) If dual port controller is used, set "clock source internal":

For a dual port controller card, set the clocking internal on the 2nd port.

Example:

```
controller T1 1/1
 clock source internal
```

### 6) Configure the channels under the controller card:

Depending on the interface to the PBX being CAS or PRI, the following commands are used.

***CAS Syntax: ds0-group <ds0-group#> timeslots <time slot range> type <switch type>***

***PRI Syntax: pri-group timeslots <time slot range> <D channel>***

**Example of T1 CAS with 12 channels:**

```
controller T1 1/0
ds0-group 0 timeslots 1-12 type e&m-wink-start
```

Note: Be sure to set the proper CAS switch type to match the PBX. Choices 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 PRI/T1 with 12 channels:**

```
controller T1 1/0
pri-group timeslots 1-12,24
```

**Example PRI/E1 with 10 channels:**

```
controller E1 1/0
 pri-group timeslots 1-10,16
```

**7) <u>Verify voice ports are created.</u>**

Once step 6 is completed, the router will automatically create the following:

**CAS:**

```
voice-port 1/0:0
```

Add an interdigit timeout of 5 seconds to the voice port :

```
voice-port 1/0:0
  timeouts interdigit 5
```

**PRI:**

For PRI, a serial and voice-port interface will be created.
Set your ISDN switch type under the Serial interface that was created :

```
interface Serial 1/0:23
isdn switch-type primary-5ess
!
voice-port 1/0:23
!
```

Note: Be sure to set the proper PRI switch type to match the PBX. Choices 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)

Add an interdigit timeout of 5 seconds:

```
voice-port 1/0:23
 timeouts interdigit 5
```

**Example:**
For a 2921 router connecting to a PRI port on a PBX with an on-board PDVM2-64 DSP cards and NM-HDV2 card in slot1 without any DSPs. The on-board DSP must be used to be able to define timeslots on the cards.

```
2921#sho run
Building configuration...

!
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 2921-1
!
boot-start-marker
```

```
boot system flash:c2900-universalk9-mz.SPA.152-1.T2ES
boot-end-marker
!
card type e1 1 1
no logging buffered
!
network-clock-participate slot 1 !allow use of network clock a a source
network-clock-select 1 E1 1/1    ! name source of the network clock
!
!
no ipv6 cef
ip source-route
ip cef
!
no ip domain lookup
ip domain name yourdomain.com

!
voice-card 0
 dspfarm             ! To allows the NM-HDV module share on-board dsps.
…
!
controller E1 1/0
 shutdown
!
controller E1 1/1
 pri-group timeslots 1-10,16
!
interface Serial1/1/:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-5ess
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
!
voice-port 0/1/0:15
 timeouts interdigit 5
```

## 2.5 Configuring Analog Voice Ports

Analog voice ports do not require as much configuration as digital ports. It is recommended to configure each port with a station-id name and its corresponding station-id number (aka phone number, DID). It is recommended to put DSPs on the motherboard when using analog ports.

Sample analog port configuration (FXS port) :

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

## *2.6 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 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 Loopback0
  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 Loopback0
  bind media source-interface Loopback0
  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" ""
 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
```

## *2.7 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.

### 2.7.1 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 g79r8 bytes 30
 codec preference 3 g711ulaw
```

### 2.7.2 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>
```

This example matches incoming calls with 5 guiding digits + variable length dial string (T)

### 2.7.3 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
```

### 2.7.4 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
```

### 2.7.5 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
```

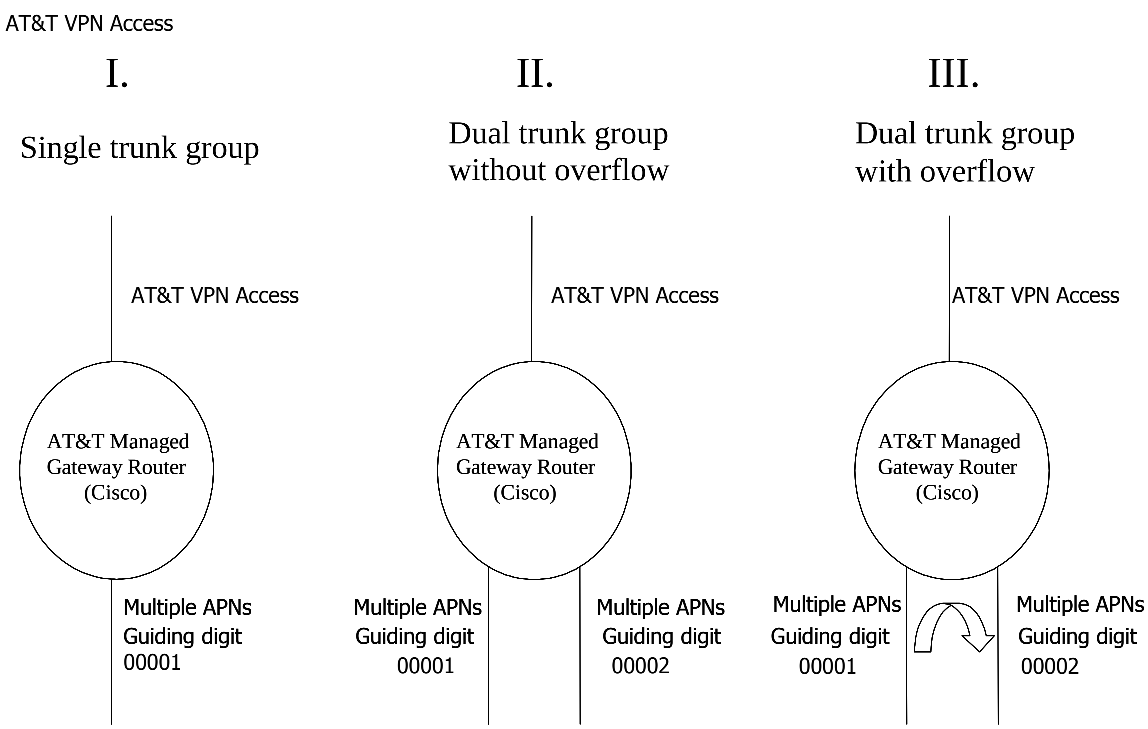
## *2.8 Additional Configuration for AT&T IP Toll-Free*

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
- g79r8 bytes 30
- g711ulaw

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

### 2.8.1 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**!
```

### 2.8.2 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

00001 9143975000 or 00002 9143976000. 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.

### 2.8.2.1 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                  Trunk Group 1
timeouts interdigit 2
music-threshold -70
busyout monitor Serial0/0/0
!
voice-port 0/1/0:1
timeouts interdigit 2               Trunk Group 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
!
```

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
 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
```

T1 PRI port configured into Trunk Group One and Two 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. Points to Trunk Group One.

```
 destination-pattern 00001T
 progress_ind alert enable 8
 direct-inward-dial

!
```

### 2.8.2.2 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***
 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          Trunk Group 1
music-threshold -70
trunk-group TG-One
!
voice-port 0/1/0:1
timeouts interdigit 2          Trunk Group 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
```

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
 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
```

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

```
 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
 !
!
```

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).

## 2.9 *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.

### 2.9.1 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
```

### 2.9.2 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).

#### 2.9.2.1 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
```

## *2.10 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
 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>
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.

## *2.11 Routing*

For routing configuration for the combined CER and TDM Gateway solution, please refer to Chapter 5 in the Customer Edge Router (CER) Customer Configuration Guide for AT&T IP Flexible Reach Service and/or AT&T IP Toll-Free on AT&T VPN as the Underlying Transport Service. http://www.corp.att.com/bvoip/avpn/implementation/ *(login: att, password: attvoip)*.

## *2.12 Configuring Call Detail Records (CDR) Collection*

Following are the commands to configure CDR collection in the TDM Gateway.
The proper key will need to be entered (provided by AT&T in the Customer Router Configuration for Customer-Managed Router and TDM PBX Letter).

```
aaa new-model

aaa group server radius h323
 server 135.89.102.66 auth-port 1812 acct-port 1813
```

```
aaa accounting connection h323 stop-only group radius group h323

gw-accounting aaa
 acct-template callhistory-detail

radius-server host 135.89.102.66 auth-port 1812 acct-port 1813 key 7 <insert key>
radius-server authorization permit missing Service-Type

radius-server vsa send accounting

ip radius source-interface < Loopback Interface for IP Flex Signaling and Media>
```

Note: Even though the signaling protocol is SIP, the accounting processes still think in terms of h323.

Note: To disable logging of Call Detail Records to the TDM Gateway, configure the following from the command line:

```
logging console 3
logging buffered 3
```

# Appendix A: Sample TDM Gateway Configuration

```
Current configuration : 12239 bytes
!
! Last configuration change at 18:42:31 EDST Mon May 2 2011 by cisco
! NVRAM config last updated at 11:15:36 EDST Fri Apr 29 2011 by cisco
!
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname 3925B-Dallas
!
boot-start-marker
boot system flash:c3900-universalk9-mz.SPA.152-1.T2ES
boot-end-marker
!
!
card type e1 0 1
logging buffered 2000000
no logging console
enable password 7 1511021F0725
!
no aaa new-model
clock timezone EST -5 0
clock summer-time EDST recurring
clock calendar-valid
network-clock-participate wic 1
network-clock-select 1 E1 0/1/0
!
no ipv6 cef
ip source-route
ip cef
!
!
no ip domain lookup
ip domain name hawaii
multilink bundle-name authenticated
!
!
voice-card 0
 dspfarm
!
!
voice service voip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
```

```
  bind control source-interface Loopback0
  bind media source-interface Loopback0
  rel1xx disable
  min-se 900 session-expires 900
  sip-profiles 1
!
voice class codec 1
 codec preference 1 g729br8 bytes 30
 codec preference 2 g79r8 bytes 30
 codec preference 3 g711ulaw
!
!
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"

!
!
!
!
license udi pid C3900-SPE100/K9 sn FOC14201W5Y
hw-module pvdm 0/0
!
!
!
username admin password 7 05080F1C2243
username vinny privilege 15 secret 5 $1$R6YO$Fwu2KYGdeFGsbgGJviSGt1
username cisco password 7 030752180500
!
redundancy
!
!
controller E1 0/1/0
 pri-group timeslots 1-12,15-16
 description TAC 737
!
!
!
!
!
interface Loopback0
 ip address 135.16.170.8 255.255.255.255
!
interface GigabitEthernet0/0
 ip address 172.22.16.2 255.255.255.0
 load-interval 30
 duplex full
 speed 100
 no keepalive
```

```
!
interface GigabitEthernet0/1
 no ip address
 duplex full
 speed auto
!
!
interface GigabitEthernet0/2
 no ip address
 duplex auto
 speed auto
!
interface Serial0/1/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
!
ip route 0.0.0.0 0.0.0.0 172.22.16.1
!
!
!
voice-port 0/1/0:15
 timeouts interdigit 5
!
!
voice-port 0/2/0
 timeouts interdigit 5
 station-id name Cornelius Smith
 station-id number 19085424000
!
voice-port 0/2/1
 timeouts interdigit 5
 station-id name Daisy Williams
 station-id number 19085424001
!
!
!
dial-peer voice 1650 voip
 description Outgoing dial-peer Border Element #1, US Calls
 destination-pattern 1..........
 rtp payload-type nse 99
 rtp payload-type nte 100
 session protocol sipv2
 session target ipv4:10.94.2.12
 dtmf-relay rtp-nte
 voice-class codec 1
 fax-relay sg3-to-g3
 fax rate 14400 bytes 48
!
dial-peer voice 1651 voip
 description Outgoing dial-peer Border Element #2, US Calls
```

```
 destination-pattern 1..........
 rtp payload-type nse 99
 rtp payload-type nte 100
 session protocol sipv2
 session target ipv4:10.94.2.13
 dtmf-relay rtp-nte
 voice-class codec 1
 fax-relay sg3-to-g3
 fax rate 14400 bytes 48
!
dial-peer voice 1652 voip
 description Outgoing dial-peer Border Element #1, International Calls
 destination-pattern 011T
 rtp payload-type nse 99
 rtp payload-type nte 100
 session protocol sipv2
 session target ipv4:10.94.2.12
 dtmf-relay rtp-nte
 voice-class codec 1
 fax-relay sg3-to-g3
 fax rate 14400 bytes 48
!
dial-peer voice 1653 voip
 description Outgoing dial-peer Border Element #2, International Calls
 destination-pattern 011T
 rtp payload-type nse 99
 rtp payload-type nte 100
 session protocol sipv2
 session target ipv4:10.94.2.13
 dtmf-relay rtp-nte
 voice-class codec 1
 fax-relay sg3-to-g3
 fax rate 14400 bytes 48
!
dial-peer voice 1654 voip
 description Outgoing Dial Peer Border Element #1, N11 IP Local Only
 preference 1
 destination-pattern [2-9]11
 rtp payload-type nse 99
 rtp payload-type nte 100
 session protocol sipv2
 session target ipv4:10.94.2.12
 dtmf-relay rtp-nte
 voice-class codec 1
 fax-relay sg3-to-g3
 fax rate 14400 bytes 48
!
dial-peer voice 1655 voip
 description Outgoing Dial Peer Border Element #2, N11 IP Local Only
 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 ipv4:10.94.2.13
 dtmf-relay rtp-nte
 voice-class codec 1
 fax-relay sg3-to-g3
 fax rate 14400 bytes 48
!
dial-peer voice 1660 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 voice 1661 pots
 answer-address 7325554000
 destination-pattern 000004000
 port 0/2/0
!
dial-peer voice 1662 pots
 answer-address 7325554001
 destination-pattern 000004001
 port 0/2/1
!
dial-peer voice 1664 pots
 destination-pattern 00000.…
 port 0/1/0:15


!
!
!
dial-peer hunt 1
!
sip-ua
 retry invite 2
 retry notify 2
 retry register 10
 timers notify 100
!
!
!
!
gatekeeper
 shutdown
!
banner exec ^C
% Password expiration warning.
```

```
-----------------------------------------------------------------------

Cisco Configuration Professional (Cisco CP) is installed on this device
and it provides the default username "cisco" for  one-time use. If you
have
already used the username "cisco" to login to the router and your IOS
image
supports the "one-time" user option, then this username has already
expired.
You will not be able to login to the router with this username after you
exit
this session.

It is strongly suggested that you create a new username with a privilege
level
of 15 using the following command.

username <myuser> privilege 15 secret 0 <mypassword>

Replace <myuser> and <mypassword> with the username and password you want
to
use.

-----------------------------------------------------------------------
^C
banner login ^C
-----------------------------------------------------------------------
Cisco Configuration Professional (Cisco CP) is installed on this device.
This feature requires the one-time use of the username "cisco" with the
password "cisco". These default credentials have a privilege level of 15.

YOU MUST USE CISCO CP or the CISCO IOS CLI TO CHANGE THESE  PUBLICLY-KNOWN
CREDENTIALS

Here are the Cisco IOS commands.

username <myuser>  privilege 15 secret 0 <mypassword>
no username cisco

Replace <myuser> and <mypassword> with the username and password you want
to use.

IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE ABLE
TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.

For more information about Cisco CP please follow the instructions in the
QUICK START GUIDE for your router or go to http://www.cisco.com/go/ciscocp
-----------------------------------------------------------------------
^C
!
line con 0
 exec-timeout 600 0
 login local
```

```
line aux 0
line vty 0 4
 exec-timeout 300 0
 privilege level 15
 login local
 transport input telnet
line vty 5 15
 access-class 23 in
 privilege level 15
 login local
 transport input telnet ssh
!
exception data-corruption buffer truncate
scheduler allocate 20000 1000
end

3925B-Dallas#
```

# Appendix B: TDM Gateway Performance

Please use the following Cisco link for router performance information:

http://www.cisco.com/en/US/prod/collateral/routers/ps259/product_data_sheet0900aecd8057f2e0.pdf

# 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 |
| 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 |
| 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 |
| | |