AT&T IP Flexible Reach Service on AT&T VPN Service

TDM Gateway
Customer Configuration Guide for
AT&T IP Flexible Reach Service
on AT&T VPN Service
as the Underlying Transport Service
for Cisco ISR G1 Platform

August 27, 2013

Version 1.5

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1 Introduction

This Customer Configuration Guide ("CCG") provides recommended guidelines for configuring the Customer-managed TDM Gateway for use with AT&T IP Flexible Reach Service (including Enhanced Features Service), on AT&T VPN Service ("AT&T VPN") as the Underlying Transport Service. TDM Gateways 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 be sure you have the latest version of this document (http://www.corp.att.com/bvoip/avpn/implementation/ (login: att, password: attvoip)). You may also wish to 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.

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

<u>Emergency 911/E911 Services Limitations and Restrictions</u> –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 Service Guide for AT&T IP Flexible Reach Service found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Service Guide 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 over AT&T VPN, the customer managed TDM Gateway is cascaded behind the Customer Edge Router (CER).

AT&T IP Flexible Reach Service over AT&T VPN service supports the following Cisco ISR G1 platforms as TDM gateways with an IOS of 12.4.(15)T13:

Routers supported:

- 2811
- 2821
- 3825
- 3845

Voice cards supported:

- VWIC2-1MFT-T1/E1
- VWIC2-2MFT-T1/E1
- VIC2-4FXS/DID
- VIC3-FXS/DID
- VIC2-4FXO
- NM-HDV2-1T1/E1
- NM-HDV2-2T1/E1

PVDM cards supported:

- PVDM2-8
- PVDM2-16
- PVDM2-32
- PVDM2-48
- PVDM2-64

1.2 Ptime Negotiation with AT&T PSTN Gateways

Ptime negotiation is important because it will determine your bandwidth per call. The PSTN hop-on / hop-off gateway used will determine the ptime negotiation for the codec.

AT&T IP Flexible Reach Service uses two types of PSTN hop-on/hop-off gateways in our network: Sonus and Nokia/Siemens (aka: NSN). The customers TN (Telephone Number) will determine which PSTN gateway is used for a particular call.

1. Your traffic might enter or exit via NSN gateways if you are located in the states listed below: AR
2. CA
3. IL
4. IN
5. KS
6. MI
7. MO
8. NV
9. OH
10. OK
11. TX
12. WI

The ptime parameter that is negotiated can vary depending on the PSTN gateway. Ptime is defined as the amount of media which can be encapsulated in each RTP packet, expressed in time, milliseconds (ms).

To negotiate codec and ptime with the AT&T PSTN gateways, AT&T recommends that "voice class codec" be applied to the dial peers on a TDM gateway. The purpose of the "voice class codec" is to prioritize the codec and ptime in order to achieve the maximum bandwidth savings. The preferences are listed in order of least bandwidth per call to highest bandwidth per call.

```
voice class codec 1
 codec preference 1 g729br8 bytes 30
 codec preference 2 g79r8 bytes 30
 codec preference 3 g711ulaw
```

When using the AT&T recommended "voice class codec" configuration on a TDM gateway, it is important to be aware of the following behavior with the AT&T PSTN gateways.

This table shows how the ptime is negotiated in the different call scenarios:

Call Scenario	SIP message detail	Result

Hopoff to PSTN (Sonus)	The Cisco gateway sends no ptime in the invite; Sonus sends max ptime of 20 in 18x and OK. Cisco ignores the max ptime.	For G.729: 30 msec G729 payload from the Cisco gateway, 20 msec from Sonus For G.711: 20 msec G711 payload from the Cisco gateway, 20 msec from Sonus
Hopon from PSTN (Sonus)	Sonus sends max ptime of 30 in invite. Cisco ignore the max ptime. Cisco sends no ptime of 20 in 18x and OK.	For G.729: 30 msec G729 payload from the Cisco gateway, 20 msec from Sonus For G.711: 20 msec G711 payload from the Cisco gateway, 20 msec from Sonus
Hopoff to PSTN (NSN)	The Cisco gateway sends no ptime in invite; NSN sends ptime of 20 in 18x and OK. Cisco accepts ptime = 20.	For G.729: 20 msec G729 payload from the Cisco gateway, 20 msec from Sonus For G.711: 20 msec G711 payload from the Cisco gateway, 20 msec from Sonus
Hopon from PSTN (NSN)	NSN sends ptime of 20 in invite. Cisco accepts the ptime. Cisco sends ptime of 20 in 18x and OK.	For G.729: 20 msec G729 payload from the Cisco gateway, 20 msec from NSN For G.711: 20 msec G711 payload from the Cisco gateway, 20 msec from Sonus

Some additional useful information follows:

1. The Cisco TDM gateway does not support max ptime from the Sonus PSTN gateway.
2. The Cisco TDM gateway does not send ptime in the invite when a Voice Class Codec is configured for an outgoing dial-peer.
3. The Cisco TDM gateway uses its configured codec bytes associated with the negotiated codec when the ptime is not negotiated. The configured codec bytes on the Cisco TDM gateway are 30 msec for G.729 and 20 msec for G.711.

4. The Sonus PSTN Gateway uses 20 msec codec bytes when the Cisco TDM gateway does not send ptime (the Invite or 18x/200 OK).

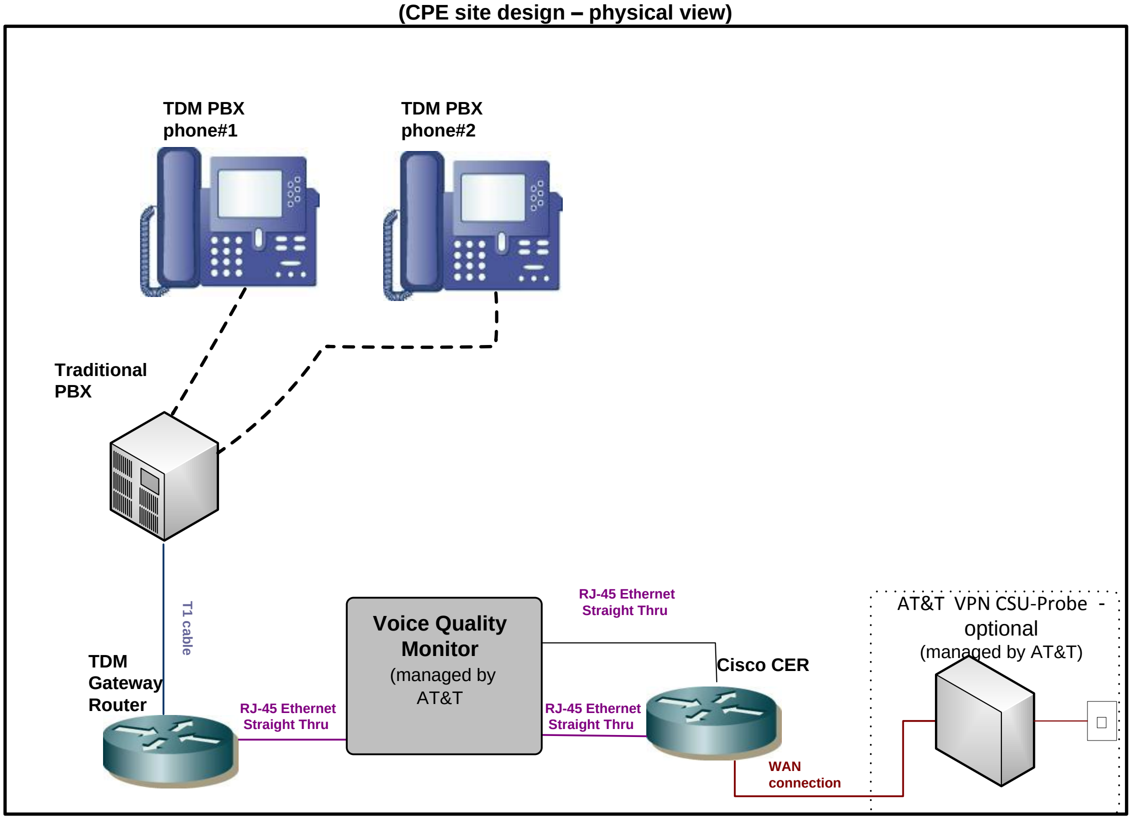
2 TDM Gateway Configurations

2.1 Supported Topologies

2.1.1 Cascaded TDM Gateway

The following is a sample network topology diagram for a site with a TDM Gateway. In this design the Customer Edge Router (CER) and TDM Gateway router are separate routers.

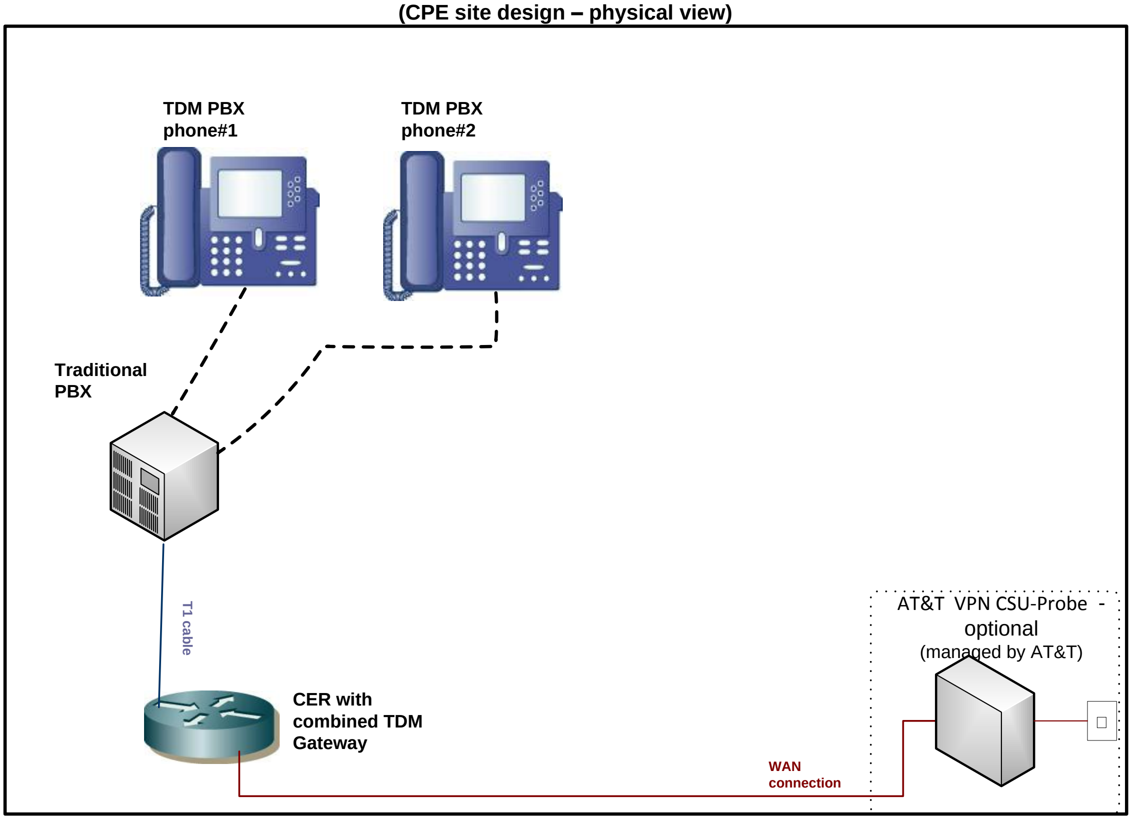
AT&T IP Flexible Reach on AT&T VPN site
with VPN CSU-Probe, Voice Quality Monitor, CER and TDM Gateway Router
(CPE site design – physical view)



2.1.2 CER combined with TDM Gateway

Following is a sample diagram of a network topology for a site with a CER combined with a TDM gateway.

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



2.2 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/vp_overview_ps6350_TSD_Products_Configuration_Guide_Chapter.html

2.3 Information on Digital Signal Processors (DSP)

DSPs are necessary for packet Telephony technologies such as AT&T IP Flexible Reach. You will need to purchase the properly sized PVDM module (see chart below) which contains DSP's. In order to determine the correct PVDM, you will need to know the number of channels and the codec used.

The table below includes a description of the number of channels and complexity supported:

	Description [1]	**Number of DSPs**	**Maximum Channels in G.711**	**Maximum Channels in High Complexity Codecs** G.729A, G.729AB, G.726, G.711, Clear-Channel, GSMFR, FAX Relay/PassThrough, Modem PassThrough	**Maximum Channels in Medium Complexity Codecs** All MC Codecs and also G.723, G.728, G.729, G.729B, GSMEFR
PVDM2-8	8-Channel Packet Fax/Voice DSP Module	1	8	4	4
PVDM2-16	16-Channel Packet Fax/Voice DSP Module	1	16	6	8
PVDM2-32	32-Channel Packet Fax/Voice DSP Module	2	32	12	16

	Description [1]	**Number of DSPs**	**Maximum Channels in G.711**	**Maximum Channels in High Complexity Codecs** G.729A, G.729AB, G.726, G.711, Clear-Channel, GSMFR, FAX Relay/PassThrough, Modem PassThrough	**Maximum Channels in Medium Complexity Codecs** All MC Codecs and also G.723, G.728, G.729, G.729B, GSMEFR
PVDM2-48	48-Channel Packet Fax/Voice DSP Module	3	48	18	24
PVDM2-64	64-Channel Packet Fax/Voice DSP Module	4	64	24	32

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

Routers such as Cisco ISR 28xx/38xx 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-fgd
- e&m-immediate-start
- e&m-lmr
- e&m-wink-start
- ext-sig
- fgd-eana
- fgd-os
- fxo-ground-start
- fxo-loop-start
- fxs-ground-start
- fxs-loop-start
- none

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) Verify voice ports are created.

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
- primary-5ess
- primary-dms100
- primary-net5
- primary-ni
- primary-ni2c
- primary-ntt
- primary-qsig
- primary-ts014

Add an interdigit timeout of 5 seconds and set your ISDN switch type :

```
voice-port 0/1/0:23
 timeouts interdigit 5
 isdn switch-type primary-5ess
```

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
```

2.5 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) 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.

<u>Steps to configure</u> 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) <u>Define card type</u>

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

Example:

```
card type e1 1 1
```

2) <u>Configure “dspfarm” for NM-HDV2 to use global DSP resources.</u>

```
voice-card 0
 dspfarm
```

3) <u>Configure the “network clock participate” for the appropriate slot:</u>

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-fgd
- e&m-immediate-start
- e&m-lmr

- e&m-wink-start
- ext-sig
- fgd-eana
- fgd-os
- fxo-ground-start
- fxo-loop-start
- fxs-ground-start
- fxs-loop-start
- none

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
- primary-5ess
- primary-dms100
- primary-net5
- primary-ni
- primary-ni2c
- primary-ntt
- primary-qsig
- primary-ts014

Add an interdigit timeout of 5 seconds and set your ISDN switch type :

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

Example:
For a 2821 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.

```
2821#sho run
Building configuration...

Current configuration : 11776 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 2821-1
!
boot-start-marker
boot-end-marker
!
card type e1 1 1
no logging buffered
!
username cisco privilege 15 secret 5 $1$71BR$DZaq7SAQzZs9JxO4VlHd./
network-clock-participate slot 1 !allow use of network clock a a source
no network-clock-participate aim 0
no network-clock-participate aim 1
network-clock-select 1 E1 1/1    ! name source of the network clock
no aaa new-model
ip subnet-zero
```

```
!
ip cef
!
no ip domain lookup
ip domain name yourdomain.com
no ftp-server write-enable
!
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
 isdn outgoing display-ie
 no cdp enable
!
voice-port 0/1/0:15
 timeouts interdigit 5
```

2.6 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
```

2.7 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. In this configuration, the loopback 0 interface is used for all SIP signaling and media traffic. **Loopback 0 must be the AT&T registered IP address (this address can be found in the "Customer Router Configuration and VQM Shipping Confirmation" letter and is referenced as the IP signaling address)**

```
voice rtp send-recv
!
voice service voip
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco
 sip
  bind control source-interface Loopback0
  bind media source-interface Loopback0
  rel1xx disable
  min-se  900
!
sip-ua
 retry invite 2
 retry notify 2
 retry register 10
 timers notify 100
```

2.8 Configuring Dial Peers for TDM Gateway

This section describes how to setup dial peers to work with AT&T IP Flexible Reach Service over AT&T VPN. A voice class codec is defined first and then is applied to the appropriate dial peers.

2.8.1 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
```

2.8.2 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.**

Example config:

```
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)

2.8.3 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
```

2.8.4 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
```

2.8.5 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
```

2.9 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).

There are different configurations to configure Caller ID for analog and digital ports.

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
```

2.9.2.2 T1/E1 Digital Ports

Caller ID can be configured on digital ports using the two steps as shown below:

Step 1: Create a translation pattern.

Use a translation pattern in order to change the calling number to a valid Telephone Number or main line number.

Syntax:

```
translation-rule 1
Rule 1 <extension> <TN>     **use this rule if TDM PBX is not configured to send full TN**
Rule 2 null <Main Line Telephone Number>
Rule 3 any <Main Line Telephone Number>
```

Example:

For extension range of 3000-3999, change the caller ID to a valid TN of 9085558888
For all other calls, change the caller ID to the Main Line Telephone Number of 9085551000

```
translation-rule 1
Rule 1 3… 9085558888
Rule 2 null 9085551000
Rule 3 any 9085551000
```

Step 2: Apply the translation rule to all voice ports.

CAS Example:

```
voice-port 0/2/0:0
translate calling 1
```

PRI Example:

```
voice-port 0/2/0:23
translate calling 1
```

2.10 Configuring CDR Collection for combined CER/TDM Gateway

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.

Appendix A: Sample TDM Gateway Configuration

```
Building configuration...

Current configuration : 9565 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname 2821
!
boot-start-marker
boot system flash:c2800nm-spservicesk9-mz.124-15.T13a.bin
boot-end-marker
!
card type t1 1 1
enable password 7 1511021F0725
!
no aaa new-model
!
resource policy
!
network-clock-participate slot 1
network-clock-switch never never
network-clock-select 1 T1 1/0
ip tcp path-mtu-discovery
!
!
ip cef
!
!
no ip domain lookup
ip domain name hawaii
ip ssh authentication-retries 5
ip ssh version 2
isdn switch-type primary-5ess
!
!
voice-card 0
 dspfarm
!
voice-card 1
```

```
 dspfarm
!
!
voice rtp send-recv
!
voice service voip
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco
 sip
  bind control source-interface Loopback0
  bind media source-interface Loopback0
  rel1xx disable
  min-se  900
!
!
voice class codec 20
 codec preference 1 g729br8 bytes 30
 codec preference 2 g729r8 bytes 30
 codec preference 3 g711ulaw
!
!
!
voice class h323 1
 h225 timeout tcp establish 3
!
!
!
!
!
!
!
!
!
application
 service autoatt flash:its-CISCO.2.0.1.0.tcl
  paramspace english language en
  paramspace english index 0
  paramspace english location flash:
 !
!
!
!
username admin privilege 15 password 7 11080D111B13091F
!
!
!
controller T1 1/0
```

```
 framing esf
 linecode b8zs
 ds0-group 3 timeslots 1-24 type e&m-wink-start
!
!
!
!
!
!
!
interface Loopback0
 ip address 135.16.170.2 255.255.255.255
!
!
interface GigabitEthernet0/0
 ip address 172.22.16.2 255.255.255.0
 duplex full
 speed 100
!
interface GigabitEthernet0/1
 no ip address
 load-interval 30
 shutdown
 duplex full
 speed 100
!
!
!
ip route 0.0.0.0 0.0.0.0 172.22.16.1
!
ip http server
no ip http secure-server
!
!
!
control-plane
!
!
!
!
voice-port 0/3/0
 timeouts interdigit 5
 station-id name Joe
 station-id number 19085424000
!
voice-port 0/3/1
```

```
 timeouts interdigit 5
 station-id name Bob
 station-id number 19085424001
!
voice-port 1/0:0
 timeouts interdigit 5
!
!
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 20
 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 20
 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 20
 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 20
 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 20
 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 20
 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 20
 fax-relay sg3-to-g3
```

```
 fax rate 14400 bytes 48
!
dial-peer voice 1661 pots
 answer-address 7325554000
 destination-pattern 000004000
 port 0/3/0
!
dial-peer voice 1662 pots
 answer-address 7325554001
 destination-pattern 000004001
 port 0/3/1
!
dial-peer voice 1664 pots
 destination-pattern 00000….
 port 1/0:0

!
!
!
dial-peer hunt 1
!
sip-ua
 retry invite 2
 retry notify 2
 retry register 10
 timers notify 100
!
!
!
line con 0
 exec-timeout 0 0
 password 7 070C285F4D06
line aux 0
line vty 0 4
 exec-timeout 300 0
 password 7 1416061F00052838
 login local
 transport input ssh
!
scheduler allocate 20000 1000
!
end
```

Appendix B: TDM Gateway Performance

The recommended maximum number of simultaneous calls and calls per second for cascaded (LAN connected) IOS gateway routers are shown below. This assumes G.729 codec is being used. The maximum rates were achieved while keeping the CPU utilization below 75% and dedicating the CER to IOS gateway usage.

Router platform	**Maximum simultaneous calls**	**Maximum call setups per second**
2811	69	1
2821	112	1
3825	280	3
3845	325	7

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