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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:

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 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.

<table>
<thead>
<tr>
<th>Call Scenario</th>
<th>SIP message detail</th>
<th>Result</th>
</tr>
</thead>
</table>

This table shows how the ptime is negotiated in the different call scenarios:
<table>
<thead>
<tr>
<th>Hopoff to PSTN (Sonus)</th>
<th>The Cisco gateway sends no ptime in the invite; Sonus sends max ptime of 20 in 18x and OK. Cisco ignores the max ptime.</th>
<th>For G.729: 30 msec G729 payload from the Cisco gateway, 20 msec from Sonus</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>For G.711: 20 msec G711 payload from the Cisco gateway, 20 msec from Sonus</td>
<td>For G.729: 30 msec G719 payload from the Cisco gateway, 20 msec from Sonus</td>
</tr>
<tr>
<td>Hopon from PSTN (Sonus)</td>
<td>Sonus sends max ptime of 30 in invite. Cisco ignores the max ptime. Cisco sends no ptime of 20 in 18x and OK.</td>
<td>For G.729: 30 msec G729 payload from the Cisco gateway, 20 msec from Sonus</td>
</tr>
<tr>
<td></td>
<td>For G.711: 20 msec G711 payload from the Cisco gateway, 20 msec from Sonus</td>
<td>For G.729: 30 msec G729 payload from the Cisco gateway, 20 msec from Sonus</td>
</tr>
<tr>
<td>Hopoff to PSTN (NSN)</td>
<td>The Cisco gateway sends no ptime in invite; NSN sends ptime of 20 in 18x and OK. Cisco accepts ptime = 20.</td>
<td>For G.729: 20 msec G729 payload from the Cisco gateway, 20 msec from Sonus</td>
</tr>
<tr>
<td></td>
<td>For G.711: 20 msec G711 payload from the Cisco gateway, 20 msec from Sonus</td>
<td>For G.729: 20 msec G729 payload from the Cisco gateway, 20 msec from Sonus</td>
</tr>
<tr>
<td>Hopon from PSTN (NSN)</td>
<td>NSN sends ptime of 20 in invite. Cisco accepts the ptime. Cisco sends ptime of 20 in 18x and OK.</td>
<td>For G.729: 20 msec G729 payload from the Cisco gateway, 20 msec from NSN</td>
</tr>
<tr>
<td></td>
<td>For G.711: 20 msec G711 payload from the Cisco gateway, 20 msec from Sonus</td>
<td>For G.729: 20 msec G729 payload from the Cisco gateway, 20 msec from NSN</td>
</tr>
</tbody>
</table>

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.

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.
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:

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:

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

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

#### 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) **Configure the “network clock select”:**

Specify 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) **If dual port controller is used, set “clock source internal”:**

For a dual port controller card, set the clocking internal on the 2<sup>nd</sup> port.

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

5) **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 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-nl2c
- 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.

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 #>

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`

Specify 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 2\textsuperscript{nd} 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:**

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

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

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

7) **Verify voice ports are created.**

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

**CAS:**

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

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

```plaintext
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:

```plaintext
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:

```plaintext
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$71BR$DZaq7SAQzZs9JxO4VIHd./
network-clock-participate slot 1 lallow 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 Serial1/1/1:15
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)**

```plaintext
voice rtp send-receive
!
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.

```plaintext
voice class codec 1
  codec preference 1 g729br8 bytes 30
  codec preference 2 g79r8 bytes 30
```
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>
2.8.3 Additional VolP 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>
dtmaf-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>
dtmaf-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>
dtmaf-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

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

```plaintext
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:

```plaintext
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:**

```plaintext
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

```plaintext
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:**

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

**PRI Example:**

```plaintext
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).

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>aaa new-model</td>
<td></td>
</tr>
<tr>
<td>aaa group server radius h323</td>
<td></td>
</tr>
<tr>
<td>server 135.89.102.66 auth-port 1812 acct-port 1813</td>
<td></td>
</tr>
<tr>
<td>aaa accounting connection h323 stop-only group radius group h323</td>
<td></td>
</tr>
<tr>
<td>gw-accounting aaa</td>
<td></td>
</tr>
<tr>
<td>acct-template callhistory-detail</td>
<td></td>
</tr>
<tr>
<td>radius-server host 135.89.102.66 auth-port 1812 acct-port 1813 key 7</td>
<td>&lt;insert key&gt;</td>
</tr>
<tr>
<td>radius-server authorization permit missing Service-Type</td>
<td></td>
</tr>
<tr>
<td>radius-server vsa send accounting</td>
<td></td>
</tr>
<tr>
<td>ip radius source-interface &lt; Loopback Interface for IP Flex Signaling and Media&gt;</td>
<td></td>
</tr>
</tbody>
</table>

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
## DSP Farm Configuration

```plaintext
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.

<table>
<thead>
<tr>
<th>Router platform</th>
<th>Maximum simultaneous calls</th>
<th>Maximum call setups per second</th>
</tr>
</thead>
<tbody>
<tr>
<td>2811</td>
<td>69</td>
<td>1</td>
</tr>
<tr>
<td>2821</td>
<td>112</td>
<td>1</td>
</tr>
<tr>
<td>3825</td>
<td>280</td>
<td>3</td>
</tr>
<tr>
<td>3845</td>
<td>325</td>
<td>7</td>
</tr>
</tbody>
</table>
### Appendix C: Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Translation</th>
</tr>
</thead>
<tbody>
<tr>
<td>AT&amp;T VPN</td>
<td>AT&amp;T Virtual Private Network</td>
</tr>
<tr>
<td>CAS</td>
<td>Channel Associated Signaling</td>
</tr>
<tr>
<td>CCG</td>
<td>Customer Configuration Guide</td>
</tr>
<tr>
<td>CCS</td>
<td>Common Channel Signaling</td>
</tr>
<tr>
<td>CEF</td>
<td>Cisco Express Forwarding</td>
</tr>
<tr>
<td>CER</td>
<td>Customer Edge Router</td>
</tr>
<tr>
<td>CLI</td>
<td>Command Line Interface</td>
</tr>
<tr>
<td>COS</td>
<td>Class of Service</td>
</tr>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>CPU</td>
<td>Central Processing Unit</td>
</tr>
<tr>
<td>DID</td>
<td>Direct Inward Dial</td>
</tr>
<tr>
<td>DSP</td>
<td>Digital Signal Processors</td>
</tr>
<tr>
<td>DTMF</td>
<td>Dual Tone Multi Frequency</td>
</tr>
<tr>
<td>E&amp;M</td>
<td>Ear &amp; Mouth</td>
</tr>
<tr>
<td>FXO</td>
<td>Foreign Exchange Office</td>
</tr>
<tr>
<td>FXS</td>
<td>Foreign Exchange Station</td>
</tr>
<tr>
<td>GSM FR</td>
<td>Global System for Mobile communications Full Rate</td>
</tr>
<tr>
<td>HDV</td>
<td>High Density Voice</td>
</tr>
<tr>
<td>HWIC</td>
<td>High-speed WAN Interface Card</td>
</tr>
<tr>
<td>IETF</td>
<td>Internet Engineering Task Force</td>
</tr>
<tr>
<td>IOS</td>
<td>Internetwork Operation System</td>
</tr>
<tr>
<td>IP</td>
<td>Internet Protocol</td>
</tr>
<tr>
<td>IPBE</td>
<td>Internet Protocol Border Element</td>
</tr>
<tr>
<td>ISR</td>
<td>Integrated Services Router</td>
</tr>
<tr>
<td>ITU-T</td>
<td>International Telecommunication Union - Telecommunications</td>
</tr>
<tr>
<td>LAN</td>
<td>Local Area Network</td>
</tr>
<tr>
<td>LD</td>
<td>Long Distance</td>
</tr>
<tr>
<td>MOW</td>
<td>Most Of World</td>
</tr>
<tr>
<td>MTU</td>
<td>Maximum Transmission Unit</td>
</tr>
<tr>
<td>NM</td>
<td>Network Module</td>
</tr>
<tr>
<td>OAM</td>
<td>Operation Administration &amp; Maintenance</td>
</tr>
<tr>
<td>PBX</td>
<td>Private Branch Exchange</td>
</tr>
<tr>
<td>POTS</td>
<td>Plain Old Telephone Service</td>
</tr>
<tr>
<td>PRI</td>
<td>Primary Rate Interface</td>
</tr>
<tr>
<td>PSAP</td>
<td>Public Safety Answering Point</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>PVDM</td>
<td>Packet Voice DSP Module</td>
</tr>
<tr>
<td>QSIG</td>
<td>Q Signaling</td>
</tr>
<tr>
<td>RC</td>
<td>Receive</td>
</tr>
<tr>
<td>RFC</td>
<td>Request for Comment</td>
</tr>
<tr>
<td>Acronym</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>--------------------------</td>
</tr>
<tr>
<td>RT</td>
<td>Real Time</td>
</tr>
<tr>
<td>RTCP</td>
<td>Real Time Control Protocol</td>
</tr>
<tr>
<td>RTP</td>
<td>Real Time Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>TAC</td>
<td>Technical Assistance Center</td>
</tr>
<tr>
<td>TDM</td>
<td>Time Division Multiplexing</td>
</tr>
<tr>
<td>TN</td>
<td>Telephone Number</td>
</tr>
<tr>
<td>UDP</td>
<td>User Datagram Protocol</td>
</tr>
<tr>
<td>VAD</td>
<td>Voice Activity Detection</td>
</tr>
<tr>
<td>VNI</td>
<td>Voice Network Infrastructure</td>
</tr>
<tr>
<td>VoIP</td>
<td>Voice over Internet Protocol</td>
</tr>
<tr>
<td>VPN</td>
<td>Virtual Private Network</td>
</tr>
<tr>
<td>VQM</td>
<td>Voice Quality Monitor</td>
</tr>
<tr>
<td>WAN</td>
<td>Wide Area Network</td>
</tr>
<tr>
<td>WIC</td>
<td>WAN Interface Card</td>
</tr>
</tbody>
</table>
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