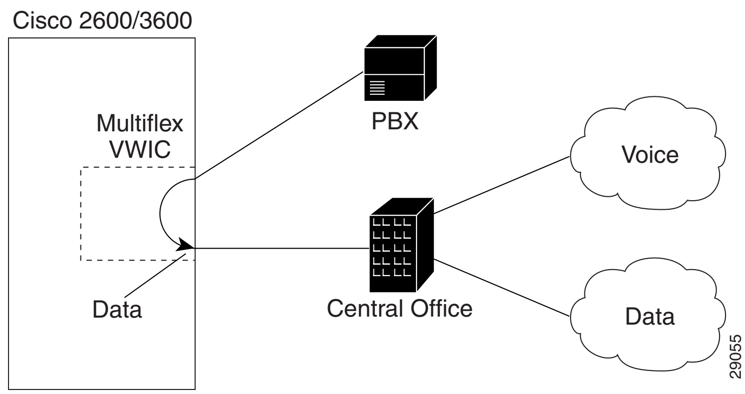
Configuring 1- and 2-Port T1/E1 Multiflex Voice/WAN Interface Cards on Cisco 2600 and 3600 Series Routers

This document explains how you can configure 1- and 2-port T1 and E1 Multiflex Voice/WAN interface cards (VWICs) on Cisco 2600 and 3600 routers and includes the following sections:

Feature Overview

Cisco T1/E1 Multiflex Voice/WAN interface cards (VWICs) support voice and data applications in Cisco 2600 and 3600 series routers. The VWICs offer WAN interface card (WIC) and voice interface card (VIC) functionality in a variety of applications for enterprises and for service providers who supply customer premises equipment.

Figure 1 T1/E1 Multiflex VWIC Applications, VWIC Ports Assigned to PBX and CO (No WAN Connectivity)



Multiflex VWICs support the following applications:

- **Data:** As WICs for T1/E1 applications, including fractional use, the T1 version integrates a fully managed data service unit/channel service unit (DSU/CSU), and the E1 version includes a fully managed DSU.
- **Packet Voice:** As VICs included with the Digital T1 Packet Voice Trunk Network Module to provide T1 connections to private branch exchanges (PBXs) and central offices (COs), the T1 VWICs enable packet voice over IP (VoIP) applications.
- **Multiplexed Voice and Data:** 2-port T1/E1 VWICs can provide Drop-and-Insert multiplexing services with integrated DSU/CSUs. For example, when used with a Digital T1 Packet Voice Trunk Network Module, Drop and Insert allows you to take 64Kb DS0 channels from one T1 and digitally cross-connect them to 64Kb DS0 channels on another T1. Drop and Insert, sometimes called *Time-Division Multiplexing (TDM) Cross-Connect*, uses circuit switching and does not use the digital signal processors (DSPs) that VoIP technology employs.

The following Multiflex VWICs are available:

- 1-port T1 Multiflex Trunk Interface (VWIC-1MFT-T1)
- 1-port E1 Multiflex Trunk Interface (VWIC-1MFT-E1)
- 2-port T1 Multiflex Trunk Interface (VWIC-2MFT-T1)
- 2-port E1 Multiflex Trunk Interface (VWIC-2MFT-E1)
- 2-port T1 Multiflex Trunk Interface with Drop and Insert (VWIC-2MFT-T1-DI)
- 2-port E1 Multiflex Trunk Interface with Drop and Insert (VWIC-2MFT-E1-DI)

Multiflex VWIC features include:

- Drop-and-Insert capabilities that allow individual 64Kb DS0 channels to be transparently passed, uncompressed, between two ports on the same Multiflex VWIC without passing through a digital signal processor (DSP):
  - By using this method, the channel traffic is sent between a PBX and central office (CO) or other telephony device.
  - In addition, Drop and Insert can cross-connect a telephony switch (from the CO or PSTN) to a channel bank to provide external analog connectivity.

Note T1/E1 channels can be used either for Drop and Insert or VoIP, but not *both*.

- Physical-layer alarm forwarding feature between the ports on 2-port cards
- T1/E1 or fractional T1/E1 network interfaces
- Per-channel T1/E1 data rates of 64 or 56 kbps for WAN services (Frame Relay or leased line)

Table 1 shows the possible hardware configurations for the Multiflex VWICs.

Table 1 Multiflex VWIC Support

VWIC and Application	Cisco 2600 Series	Cisco 3620 and 3640	Cisco 3660
1-port Data only	T1 or E1 VWIC in a chassis slot	T1 or E1 VWIC in a 1- or 2-port network module (NM-1E2W, NM-2E2W, NM-1E1R2W)	Planned for future availability
2-port Data only	T1 or E1 VWIC in a chassis slot. *Does not provide packet voice.* Can provide two physical WAN connections with both ports supporting up to full T1/E1 speeds.	Planned for future availability	Planned for future availability
2-port Drop-and-Insert	T1 or E1 D&I VWIC in a chassis slot. *Does not provide packet voice.* Provides WAN connections and digital cross-connect.	T1 D&I VWIC in a Digital T1 Packet Voice Trunk Network Module. Provides voice connections and digital cross-connect. *Does not provide WAN connections.* T1 or E1 D&I VWIC in a 1- or 2-port network module (NM-1E2W, NM-2E2W, NM-1E1R2W). Provides WAN connections and digital cross-connect.	T1 D&I VWIC in a Digital T1 Packet Voice Trunk Network Module. Provides voice connections and digital cross-connect. *Does not provide WAN connections.*
1- or 2-port Voice only, no WAN connections	T1 VWIC in a Digital T1 Packet Voice Trunk Network Module	T1 VWIC in a Digital T1 Packet Voice Trunk Network Module	T1 VWIC in a Digital T1 Packet Voice Trunk Network Module

Benefits

T1/E1 Multiflex VWICs reduce networking life-cycle costs in the following ways:

- Allow an efficient transition from data-only applications to both multiplexed voice and data and packet voice applications.
- Are easier to deploy and manage than single-purpose interfaces.

T1/E1 Multiflex VWICs provide the following benefits of multifunction support for LAN-to-LAN routing, multiplexed voice and data, and voice:

- Eliminate costly external CSUs or DSUs.
- Eliminate the need for Drop-and-Insert multiplexers.
- Simplify remote network management by allowing a single management tool, such as CiscoView or CiscoWorks, to support router, CSU/DSU, Drop-and-Insert multiplexer functions.
- Increase T1/E1 port density supported on Cisco 2600 series routers and permit two T1/E1 connections in a single WIC slot.

Restrictions

The following restrictions apply to T1/E1 Multiflex VWIC configurations:

- On Cisco 3660 platforms, Multiflex VWICs are supported only when they are installed in a Digital T1 Packet Voice Trunk Network Module.
- On all Cisco 2600 and 3600 platforms, Digital T1 Packet Voice Trunk Network Modules only support T1 Multiflex VWICs.
- E1 VWICs are not supported on Cisco 3660 platforms.
- Cisco 3620 and 3640 combination network modules allow the installation of either a 1-port VWIC or a 2-port Drop-and-Insert VWIC.
- Drop-and-Insert capability is supported only between two ports on the same multiflex card.
- When installed in a Cisco 2600 chassis slot, DSP resources for packet voice are not available to the Multiflex VWICs with Drop and Insert.

See Table 1 on page 3 for summary information.

Related Features and Technologies

Digital T1 Packet Voice Trunk Network Modules requires 2-port T1 Multiflex VWIC for operation. For more information about these modules, see *Configuring Digital T1 Packet Voice Trunk Network Modules on Cisco 2600 and 3600 Series Routers*.

Related Documents

The following documents provide additional information about installing and configuring T1/E1 Multiflex VWICs:

- *Cisco Network Modules Hardware Installation Guide*
- *Configuring Digital T1 Packet Voice Trunk Network Modules on Cisco 2600 and 3600 Series Routers*

The following Cisco IOS Release 12.0 documents provide information that can help you use T1/E1 Multiflex VWICs:

- *Voice, Video, and Home Applications Configuration Guide*
- *Voice, Video, and Home Applications Command Reference*
- *Network Protocols Configuration Guide, Part 1*
- *Cisco IOS Interface Configuration Guide*
- *Cisco IOS Interface Command Reference*
- *Dial Solutions Configuration Guide*

Supported Platforms

This feature is supported on the following platforms:

- Cisco 2610
- Cisco 2611
- Cisco 2612
- Cisco 2613
- Cisco 2620
- Cisco 2621
- Cisco 3620
- Cisco 3640
- Cisco 3661
- Cisco 3662

Supported Standards, MIBs, and RFCs

T1/E1 Multiflex VWICs support the standards, MIBs, and RFCs listed in this section.

T1 Compliance (Partial List)

- ANSI T1.403
- US (UL 1950, T1)
- FCC Part 68
- CS-03
- Canada (CSA 950, T1)
- US (FCC Part 15 Class B, T1)
- U.K. (BS6301, EN60950, EN41003)
- Canada (CSA C108.8 Class A, T1)
- Bellcore---AT&T Accunet (62411)
- ATT 54016
- Japan (VCCI Class 2, VCCI:V-3/97.04, T1, JATE Green Book, IEC950)

E1 Compliance (Partial List)

- Australia (TS 016, AS/NZS 3548:1995)
- Germany (TUV GS, EN60950)
- Germany (VDE 0878 part 3 and 30)
- France (NFC98020, EN60950, EN41003)
- Sweden (SS447-2-22, SS636334, EN60950)

- UK (NTR4)
- Europe (EN55022 Class B, EN55102-1, EN55102-2, CTR12, EN60950, EN50082-1:1992, EN55022:1994)
- CCITT/ITU G.704, I.431
- ETSI NET5, ETS300156
- TBR4
- CTR-13
- ETS 300011
- ITU I.431

RFC

RFC 1406

MIB

- T1 CSU MIB

Other Standards

- ANSI T1.40
- AT&T Publication 62411

Prerequisites

T1/E1 Multiflex VWICs require specific service, software, and hardware:

- Obtain T1 or E1 service from your service provider.
- Install Cisco IOS Software Release 12.0(5)XK, 12.0(7)T or a later release.
- If you are installing Multiflex VWICs in a Digital T1 Packet Voice Trunk Network Module, see the following documents for more information about the module:
  - *Cisco 2600 and 3600 Series Network Module Hardware Installation Guide*
  - *Configuring Digital T1 Packet Voice Trunk Network Modules on Cisco 2600 and 3600 Series Routers*

Note You can install one Digital T1 Packet Voice Trunk Network Module in a Cisco 2600 series router or a Cisco 3620 router. A Cisco 3640 router can support three modules, and you can install as many as six modules in a Cisco 3660 router.

- Install the T1 or E1 Multiflex VWIC by following the instructions in *Cisco Network Modules Hardware Installation Guide*.
- If you are using Drop and Insert with a Digital T1 Packet Voice Trunk Network Module, install at least one other network module or WAN interface card to provide the connection to the IP LAN or WAN.

- Establish a working IP network. For more information about configuring IP, see “IP Overview,” “Configuring IP Addressing,” and “Configuring IP Services” chapters in the Cisco IOS Release 12.0 *Network Protocols Configuration Guide, Part 1*.
- Complete your company’s dial plan.
- Establish a working telephony network based on your company's dial plan.

Voice, Video, and Home Applications Configuration Guide and *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.0 provide information about setting up voice networks.

Configuration Tasks

Configuring T1/E1 Multiflex VWICs includes the following tasks:

- Setting up voice cards (voice only) and T1/E1 controllers.
- Configuring serial and LAN interfaces.
- Setting up voice ports (voice services applying only to T1 Multiflex VWICs installed in Digital T1 Packet Voice Trunk Network Module).
- Configuring voice dial peers (voice services applying only to T1 Multiflex VWICs installed in Digital T1 Packet Voice Trunk Network Modules).

For detailed information about configuring a T1 Multiflex VWIC that is installed in a Digital T1 Packet Voice Trunk Network Module, see *Configuring Digital T1 Packet Voice Trunk Network Modules on Cisco 2600 and 3600 Series Routers*.

Configuring Voice Card and Controller Settings

This section includes the following sections:

- Configuring voice cards and DS0 groups, only for T1 Multiflex VWICs installed in Digital T1 Packet Voice Trunk Network Modules when voice services are required
- Configuring T1 or E1 controllers
- Configuring Drop and Insert for T1 or E1

Configuring Voice Cards and DS0 Groups

Follow the steps below if you are configuring T1 Multiflex VWICs installed in Digital T1 Packet Voice Trunk Network Modules for voice. Repeat Steps 2 and 3 for each voice card.

Step	Command	Purpose
1	`Router#` **`configure terminal`**	Enter global configuration mode.
2	`Router(config)#` **`voice-card`** *`slot`*	Enter voice card interface configuration mode and specify the slot location by using a value from 0 to 5, depending upon your router.
3	`Router(config-voice-ca)#` **`codec complexity`** `{`**`high`** `|` **`medium`**`}`	Specify the codec complexity based on the codec standard you are using. High-complexity codecs support lower call density than do medium-complexity codecs. The number of channels supported is based on the number of packet voice data modules (PVDMs) installed and the codec complexity. Here is a guideline: • In high-complexity codec mode, up to six voice or fax calls can be completed per PVDM-12, using the following codecs: G.711, G.726, G.729, G.729 Annex B, G.723.1, G.723.1 Annex A, G.728, and fax relay. • In medium-complexity codec mode, up to twelve voice or fax calls can be completed per PVDM-12, using the following codecs: G.711, G.726, G.729 Annex A, G.729 Annex B with Annex A, and fax relay All voice cards in a router must use the same codec complexity setting. The keyword that you specify for **codec complexity** affects the choice of codecs available using the **codec** dial-peer configuration command. See Step 7 in "Configuring Voice Dial Peers" on page 17. You cannot change codec complexity while DS0 groups are defined. If they are already set up, use the **no ds0-group** command before resetting the codec complexity. For more information about the **ds0-group** command, see Step 5.
4	`Router(config)#` **`controller T1`** *`slot/port`*	Enter controller configuration mode for the VWIC. Valid values for *slot* are 0 through 5 and for *port* are 0 and 1.

Step	Command	Purpose
5	`Router(config-controller)#` **`ds0-group`** *`ds0-group-no`* **`timeslots`** *`timeslot-list`* **`type`** `{`**`e&m-immediate`** `|` **`e&m-delay`** `|`**`e&m-wink`** `|` **`fxs-ground-start`** `|` **`fxs-loop-start`** `|` **`fxo-ground-start`** `|` **`fxo-loop-start`**`}`	(Voice only) This command defines the T1 channels for use by compressed voice calls as well as the signaling method the router uses to connect to the PBX or CO. Set up DS0 groups after you have specified codec complexity in voice-card configuration, as shown in Step 3. If you modify the **codec complexity** command parameters, you must first remove any existing DS0 groups, then reinstate them after the change to the codec complexity. *ds0-group-no* is a value from 0 to 23 that identifies the DS0 group. **Note** The **ds0-group** command automatically creates a logical voice port that is numbered as follows: *slot*/*port*:*ds0-group-no*. Although only one voice port is created, applicable calls are routed to any channel in the group. *timeslot-list* is a single number, numbers separated by commas, or a pair of numbers separated by a hyphen to indicate a range of timeslots. For T1, allowable values are from 1 to 24. To map individual DS0 timeslots, define additional groups. The system maps additional voice ports for each defined group. See Step 2 of "Configuring Voice Ports" on page 15. The signaling method selection for **type** depends on the connection that you are making: • The E&M interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The wink and delay settings both specify confirming signals between the transmitting and receiving ends, whereas the immediate setting stipulates no special offhook/onhook signals. • The FXO interface is for connection of a central office (CO) to a standard PBX interface where permitted by local regulations; the interface is often used for off-premises extensions. • The FXS interface allows connection of basic telephone equipment and PBXs.

Configuring T1 and E1 Controllers

Follow this procedure to configure T1 and E1 controllers. Skip Steps 1 and 2 if you are already in controller configuration mode.

Repeat the steps following Step 2 for each controller.

1	`Router#` **`configure terminal`**	Skip this step if you are already in controller configuration mode. Enter global configuration mode.
2	`Router(config)#` **`controller`** `{`**`T1`** `|` **`E1`**`}`*`slot`*`/`*`port`*	Skip this step if you are already in controller configuration mode. Enter controller configuration mode for the T1 or E1 controller at the specified *slot/port* location.
3	`Router(config-controller)#` **`loopback`** `{`**`diagnostic`** `|` **`local`** `{`**`payload`**`|`**`line`**`}|`**`remote`**`{`**`iboc`** `|`**`esf`** `{`**`payload`** `|` **`line`**`}}`	(Optional, T1 only, testing) This command generates a local loopback test at the line or payload level or a remote loopback. For details, *Configuring Digital T1 Packet Voice Trunk Network Modules on Cisco 2600 and 3600 Series Routers*.
4	`Router(config-controller)#` **`clock source`** `{`**`line`** `[`**`primary`**`]` `|` **`internal`**`}`	Specify the clock source. The **line** keyword specifies that the clock source is derived from the active line—rather than from the free-running internal clock. This is the default setting and is generally more reliable. These rules apply to clock sourcing: • When both ports are set to line clocking with no primary specification, port 0 is the default primary clock source and port 1 is the default secondary clock source. • When both ports are set to line and one port is set as the primary clock source, the other port is by default the backup or secondary source and is loop-timed. • If one port is set to **clock source line** or **clock source line primary** and the other is set to **clock source internal**, the internal port recovers clock from the clock source line port if the clock source line port is up. If it is down, then the internal port generates its own clock. • If both ports are set to **clock source internal**, there is only one clock source—internal.

5	`Router(config-controller)#` **`framing`** `{`**`sf`** `|` **`esf`**`}`
	or
	
	`Router(config-controller)#` **`framing`** `{`**`crc4`** `|` **`no-crc4`**`}` `[`**`australia`**`]`
		Set the framing to SuperFrame (SF) or Extended SuperFrame (ESF) format, according to service provider requirements.
		
		Set the framing to cyclic redundancy check 4 (CRC4) or no CRC4, according to service provider requirements. The **australia** optional keyword specifies Australian Layer 1 Homologation for E1 framing.
		
		Note If you will be configuring drop and insert, the T1 framing under the controllers involved (where the tdm-groups are configured), needs to be the same. If different framing types are used, the signaling bits may not be understood properly when a channel from one controller is dropped and inserted into a channel from another controller.

6	`Router(config-controller)# linecode {b8zs	ami	hdb3}`	Set the line encoding according to your service provider's instructions. Bipolar-8 zero substitution (B8ZS), available only for T1 lines, encodes a sequence of eight zeros in a unique binary sequence to detect line coding violations. Alternate mark inversion (AMI), available for T1 or E1 lines, represents zeros using a 01 during each bit cell, and ones are represented by 11 or 00, alternately, during each bit cell. AMI requires that the sending device maintain ones density. Ones density is not maintained independent of the data stream. For E1, set the line coding to either AMI or high-density bipolar 3 (HDB3), the default.
7	`Router(config-controller)# line-termination {75-ohm	120-ohm}`	(E1 only) Enter a line-termination value. This command specifies the impedance (amount of wire resistance and reactivity to current) for the E1 termination. Impedance levels are maintained to avoid data corruption over long-distance links. Specify **120-ohm** to match the balanced 120-ohm interface. This is the default. **75-ohm** is for an unbalanced BNC 75-ohm interface.
8	`Router(config-if)# fdl {att	ansi	both}`	(T1 interfaces only) This command sets the Facility Data Link (FDL) exchange standard for the CSU controllers. The FDL is a 4-Kbps channel used with the Extended SuperFrame (ESF) framing format to provide out-of-band messaging for error-checking on a T1 link. You typically leave this setting at the default, **ansi,** which follows the ANSI T1.403 standard for extended superframe facilities data link exchange support. Changing it allows improved management in some cases but can cause problems if your setting is not compatible with that of your service provider. **att** selects the AT&T TR54016 standard for extended superframe facilities data link exchange support. **both** enables both of the above standards.

9	Router(config-controller)# **cablelength long** {**gain26** | **gain36**} {**-15db** | **-22.5db** | **-7.5db** | **0db**} or **cablelength short** {**133** | **266** | **399** | **533** | **655**}	(T1 interfaces only) The cable length setting must conform to the actual cable length you are using. For example, if you attempt to enter the **cablelength short** command on a long-haul T1 link, the command is rejected. To set a cable length longer than 655 feet for a T1 link, enter the **cablelength long** command: • **gain26** specifies the decibel pulse gain at 26. This is the default pulse gain. • **gain36** specifies the decibel pulse gain at 36. • **-15db** specifies the decibel pulse rate at -15 decibels. • **-22.5db** specifies the decibel pulse rate at -22.5 decibels. • **-7.5db** specifies the decibel pulse rate at -7.5 decibels. • **0db** specifies the decibel pulse rate at 0 decibels. This is the default pulse rate. To set a cable length 655 feet or less for a T1 link, enter the **cablelength short** command. There is no default for **cablelength short**: • **133** specifies a cable length from 0-133 feet. • **266** specifies a cable length from 134-266 feet. • **399** specifies a cable length from 267-399 feet. • **533** specifies a cable length from 400-533 feet. • **655** specifies a cable length from 534-655 feet. If you do not set the cable length, the system defaults to a setting of **cablelength long gain26 0db**.

Configuring Drop and Insert

Perform the steps in this section if you are setting up Drop and Insert. If not, proceed to "Configuring Serial Interfaces" on page 14.

1	Router(config-controller)# **tdm-group** *tdm-group-no* **timeslots** *timeslot-list* **type** [**e&m** | **fxs** [**loop-start** | **ground-start**] **fxo** [**loop-start** | **ground-start**]	Enter this command to set up TDM channel groups for the Drop-and-Insert function with a 2-port Multiflex VWIC. *tdm-group-no* is a value from 0 to 23 for T1 and from 0 to 30 for E1; it identifies the group. *timeslot-list* is a single number, numbers separated by commas, or a pair of numbers separated by a hyphen to indicate a range of timeslots. The valid range is from 1 to 24 for T1. For E1, the range is from 1 to 31. The signaling method selection for **type** depends on the connection that you are making. The **fxs** and **fxo** options allow you to specify a ground-start or loop-start line. The Cisco IOS Release 12.0 *Voice, Video, and Home Applications Command Reference* includes additional information about these options. **Note** The group numbers for controller groups must be unique. For example, a TDM group should not have the same ID number as a DS0 group or channel group.

2	`Router(config-controller)#` **`channel-group`** *`channel-group-no`* **`timeslots`** *`timeslot-list`* `[`**`speed`** `[`**`48`**`|`**`56`**`|`**`64`**`]]`	(Optional) Enter this command to set up channel groups for WAN data services with a 2-port Multiflex Drop-and-Insert VWIC. *channel-group-no* is a value from 0 to 23 for T1 and from 0 to 30 for E1; because there can be only one channel group on a 1- or 2-port Multiflex VWIC, 0 is always the value. *timeslot-list* is a single number, numbers separated by commas, or a pair of numbers separated by a hyphen to indicate a range of timeslots. The valid range is from 1 to 24 for T1. For E1, the range is from 1 to 31. The optional **speed** setting defaults to 56 Kbps for T1 and 64 Kbps for E1. **Note** Although the CLI displays **48** as a speed option, it is not supported.
3	`Router(config-controller)#` **`no shutdown`**	Activate the controller.
4	`Router(config-controller)#` **`exit`**	Exit controller configuration mode. Skip the next step if you are not setting up Drop and Insert.
5	`Router(config)#` **`connect`** *`id`* `{`**`T1`** `|` **`E1`**`}` *`slot/port-1 tdm-group-no-1`* `{`**`T1`** `|` **`E1`**`}` *`slot/port-2 tdm-group-no-2`*	This global configuration command sets up the connection between two T1 or E1 TDM groups of timeslots on the WVIC—for Drop and Insert. *id* is a name for the connection. Identify each controller by its *slot/port* location. *tdm-group-no-1* and *tdm-group-no-2* identify the TDM group numbers (from 0 to 23 or 30) on the specified controller. The groups were set up in Step 1. **Note** When configuring drop and insert, the T1 framing under the controllers involved (where the tdm-groups are configured), needs to be the same. If different framing types are used, the signaling bits may not be understood properly when a channel from one controller is dropped and inserted into a channel from another controller. See the "Configuration Examples" section on page 21 for sample Drop and Insert configurations.

Verifying Voice Card and Controller Settings

Step 1 Enter the **show running-config** command to display the current voice-card setting. If no codec complexity is shown, the default of medium complexity is set. The following example shows an excerpt from the command output:

```
Router# show running-config
.
.
hostname router-alpha

voice-card 1
 codec complexity high
.
.
.
```

Step 2 The privileged EXEC **show controllers t1/e1** command displays the status of T1 or E1 controllers and displays information about clock sources and other settings for the ports:

```
Router# show controller T1 1/0

T1 1/0 is up.
  Applique type is Channelized T1
  Cablelength is short 133
  Description: T1 WIC card Alpha
  No alarms detected.
  Framing is ESF, Line Code is B8ZS, Clock Source is Line Primary.
  Data in current interval (1 seconds elapsed):
     0 Line Code Violations, 0 Path Code Violations
     0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
     0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
```

Step 3 The privileged EXEC **show connection all** command displays the status of T1 or E1 TDM controller groups and how they are set up:

```
Router# show connection all

ID   Name               Segment 1           Segment 2           State
========================================================================
1    Test              -T1 1/0 01          -T1 1/1 02          ADMIN UP
```

Configuring Serial Interfaces

The way you set up serial and LAN interfaces depends on your application. This document does not explain all possible serial interface configuration options, nor does it show LAN interface configuration. For complete information, see the Cisco IOS Release 12.0 *Cisco IOS Interface Configuration Guide* and the *Cisco IOS Interface Command Reference*.

If you are not planning voice support, proceed to “Configuration Examples” on page 21.

To configure VoIP, you must at least set up IP addresses for serial interfaces. When a user dials enough digits to match a configured destination pattern, the telephone number is mapped to an IP host through the dial plan mapper. The IP host has a direct connection to either the destination telephone number or a PBX that completes the call to the configured destination pattern.

Note For information about monitoring serial interfaces in order to trigger a busy-out condition on a voice port when an interface is down, see “Configuring Voice Ports” on page 15.

Step	Command	Purpose
1	`Router# configure terminal`	Enter global configuration mode.
2	`Router(config)# interface serial` *`slot/port:channel-group`*	Enter interface configuration mode for a serial interface that you specify by slot and port. The *:channel-group* portion of the command is only required for channelized T1 or E1 interfaces; its value is always 0 for Multiflex VWIC support. (For setting up channelized interfaces, see *Dial Solutions Configuration Guide* for Cisco IOS Release 12.0.)
3	`Router(config-if)# ip address` *`ip-address mask`*	Assign the IP address and subnet mask to the interface.

Verifying Serial Interface Configuration

To verify serial interface configuration, enter the privileged EXEC command **show interfaces serial**, which shows the status of all serial interfaces or of a specific serial interface, as in the following example. You can use this command to check the encapsulation, IP addressing, and other settings:

```
Router #show interface serial0/0:0
Serial0/0:0 is up, line protocol is up
  Hardware is QUICC Serial
  Internet address is 1.156.1.1/24
  MTU 1500 bytes, BW 1536 Kbit, DLY 20000 usec,
     reliability 255/255, txload 1/255, rxload 1/255
  Encapsulation HDLC, loopback not set
  Keepalive not set
  Last input 00:00:00, output 00:00:00, output hang never
  Last clearing of "show interface" counters never
  Input queue: 0/75/0 (size/max/drops); Total output drops: 0
  Queueing strategy: weighted fair
  Output queue: 0/1000/64/0 (size/max total/threshold/drops)
     Conversations  0/1/256 (active/max active/max total)
     Reserved Conversations 0/0 (allocated/max allocated)
  5 minute input rate 1000 bits/sec, 1 packets/sec
  5 minute output rate 1000 bits/sec, 1 packets/sec
     637 packets input, 64736 bytes, 0 no buffer
     Received 181 broadcasts, 0 runts, 5 giants, 0 throttles
     3617 input errors, 1506 CRC, 1646 frame, 0 overrun, 0 ignored, 0 abort
     682 packets output, 67213 bytes, 0 underruns
     0 output errors, 0 collisions, 1070 interface resets
     0 output buffer failures, 0 output buffers swapped out
     13 carrier transitions
     Timeslot(s) Used:1-24, Transmitter delay is 0 flags
```

Configuring Voice Ports

Follow these steps to set up voice ports to support the local and remote stations. This procedure applies only to T1 Multiflex VWICs installed in Digital T1 Packet Voice Trunk Network Modules when voice services are required.

This section does not show all the commands that you can use. To learn more, see *Voice, Video, and Home Applications Configuration Guide* and *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.0.

Step	Command	Purpose
1	`Router#` **`configure terminal`**	Enter global configuration mode.

Step	Command	Purpose
2	`Router(config)#` **`voice-port`** *`slot/port:ds0-group-no`*	Enter voice-port configuration mode. *slot* is the router location where the voice module is installed. Valid entries are from 0 to 3. *port* indicates the Multiflex VWIC location. Valid entries are 0 or 1. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1 card. For more information about DS0 groups, see Step 5 of "Configuring Voice Card and Controller Settings" on page 7. **Note** This **voice-port** command syntax does not apply to analog voice network modules and voice interface cards. Specify voice interface cards by using *slot*/*subunit*/*port*, designating the router slot for the voice network module, the location of the voice interface card in the network module, and the port on the voice interface card.
3	`Router(config-voice-port)#` **`busyout monitor interface`** *`interface number`*	(Optional) This command allows you to specify a LAN or WAN interface that will be monitored, and, when it is down, trigger a busyout (offhook) state on the voice port. This allows rerouting of calls. For example, if you specify *Serial 1/0* as the interface and number, the voice port sends a busyout signal when that interface is down. You can issue the command repeatedly to specify as many interfaces, virtual interfaces, and subinterfaces as are required for a voice port. For example, if you issue the command three times so that three interfaces are monitored, the voice port only goes into busyout state when all three interfaces are down. When any one of the interfaces is operational, the busyout state is removed.
4	`Router(config-voice-port)#` **`comfort-noise`**	(Optional) This parameter is enabled by default. It creates subtle background noise to fill silent gaps during calls when VAD is enabled on voice dial peers. If comfort noise is not generated, the silence can be unsettling to callers.
5	`Router(config-voice-port)#` **`echo-cancel enable`**	(Optional) This setting is enabled by default. Echo cancellation adds to the quality of voice transmissions by adjusting the echo that occurs on the interface due to impedance mismatches. Some echo is reassuring; echo over 25 milliseconds can cause problems.
6	`Router(config-voice-port)#` **`echo-cancel coverage`** `{`**`16`** `|` **`24`** `|`**`32`** `|` **`8`**`}`	(Optional) This command adjusts the echo canceller by the specified number of milliseconds; the default is 16.
7	`Router(config-voice-port)#` **`connection`** `{`**`plar`** `|`**`trunk`**`}` *`string`*	(Optional) This command sets up a connection mode for the voice port. **plar** specifies a private line automatic ring down (PLAR) connection, which rings a remote telephone when the dial peer goes off hook. **trunk** specifies a straight tie-line connection to a PBX. *string* specifies the remote telephone number or significant start digits of the number. See "Configuration Examples" on page 21 for sample PLAR and trunk configurations.

Step	Command	Purpose
8	`Router(config-voice-port)#` **`timeouts interdigit`** *`seconds`*	(Optional) This command sets the number of seconds the system waits—after the caller has input the initial digit—for a subsequent digit of the dialed string. If the timeout ends before the destination is identified, a tone sounds and the call ends. The default value is 10 seconds, and the timeout can be set from 0 to 120 seconds. **Note** Changes to the default for this command normally are not required. Other timing settings may also be needed. For more information, see the Cisco IOS Release 12.0 *Voice, Video, and Home Applications Configuration Guide*.
9	`Router(config-voice-port)#` **`exit`**	Exit voice-port configuration mode. Repeat Steps 2 through 9 for each DS0 group you create.

Verifying Voice Ports

Follow this procedure to verify voice-port configuration. To learn more, see Cisco IOS Release 12.0 *Voice, Video, and Home Applications Command Reference*.

Important command output is shown in bold.

To verify the voice-port configuration, enter the privileged EXEC **show voice port** *slot*/*port*:*ds0-group* command. The following sample output from the command shows explanatory information after the << characters:

```
cisco-router# show voice port 1/0:1

receEive and transMit Slot is 1, Sub-unit is 0, Port is 1  << voice-port 1/0:1
 Type of VoicePort is E&M
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Region Tone is set for US
```

Configuring Voice Dial Peers

Follow these steps to set up voice dial peers to support the local and remote stations. This procedure applies only to T1 Multiflex VWICs installed in Digital T1 Packet Voice Trunk Network Modules when voice services are required.

This section does not show all the commands that you might need to enter. To learn more, see *Voice, Video, and Home Applications Configuration Guide* and *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.0.

Step	Command	Purpose
1	`Router#` **`configure terminal`**	Enter global configuration mode.
2	`Router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Enter dial-peer configuration mode and define a local dial peer that will connect to the plain old telephone service (POTS) network. *number* is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647. **pots** indicates a peer using basic telephone service.
3	`Router(config-dialpeer)#` **`destination-pattern`** *`string`* `[`**`T`**`]`	Configure the dial peer's destination pattern, so that the system can reconcile dialed digits with a telephone number. *string* is a series of digits that specify the E.164 or private dialing plan phone number. Valid entries are the digits 0 through 9 and the letters A through D. The plus symbol (+) is not valid. You can enter the following special characters: • The star character (*) that appears on standard touch-tone dial pads can be in any dial string but not as a leading character (for example, *650). • The period (.) acts as a wildcard character. • The comma (,) can be used only in prefixes and inserts a one-second pause. When the timer (T) character is included at the end of the destination pattern, the system collects dialed digits as they are entered—until the interdigit timer expires (10 seconds, by default)—or the user dials the termination of end-of-dialing key (default is #). **Note** The timer character must be a capital T.
4	`Router(config-dialpeer)#` **`prefix`** *`string`*	(Optional) Include a dial-out prefix that the system enters automatically instead of people dialing it. *string* is a value from 0 to 9, and you can use a comma (,) to indicate a pause. **Note** There are other digit manipulation commands available to handle such situations as prefixes for special services, ignoring some digits, and dialing in to remote PBXs as though they are local.
5	`Router(config-dialpeer)#` **`port`** *`slot/port:ds0-group-no`*	This command associates the dial peer with a specific logical interface. *slot* is the router location where the voice module is installed. Valid entries are from 0 to 3. *port* indicates the voice interface card location. Valid entries are 0 or 1. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1 card.

Step	Command	Purpose
6	`Router(config)#` **`dial-peer voice`** *`number`* **`voip`**	Enter dial-peer configuration mode and define a remote VoIP dial peer. *number* is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647. **voip** indicates a VoIP peer using voice encapsulation on the IP network.
7	`Router(config-dialpeer)#` **`codec`** {**`g711alaw`** | **`g711ulaw`** | **`g723ar53`** | **`g723ar63`** | **`g723r53`** | **`g723r63`** | **`g726r16`** | **`g726r24`** | **`g726r32`** | **`g728`** | **`g729r8`** [**`pre-ietf`**] | **`g729br8`**} [*`bytes`*]	The voice-card configuration **codec complexity** command sets the codec options that are available when you enter this command (see Step 3 of "Configuring Voice Card and Controller Settings" on page 7). If you do not set codec complexity, **g729r8** with IETF bit-ordering is used. If you set codec complexity to **high**, the following options are available: • **g711alaw**—G.711 A Law 64,000 bps • **g711ulaw**—G.711 u Law 64,000 bps • **g723ar53**—G.723.1 Annex A 5,300 bps • **g723ar63**—G.723.1 Annex A 6,300 bps • **g723r53**—G.723.1 5,300 bps • **g723r63**—G.723.1 6,300 bps • **g726r16**—G.726 16,000 bps • **g726r24**—G.726 24,000 bps • **g726r32**—G.726 32,000 bps • **g728**—G.728 16,000 bps • **g729r8**---G.729 8,000 bps (default) • **g729br8**—G.729 Annex B 8,000 bps If you set codec complexity to **medium**, the following options are valid: • **g711alaw**—G.711 A Law 64,000 bps • **g711ulaw**—G.711 u Law 64,000 bps • **g726r16**—G.726 16,000 bps • **g726r24**—G.726 24,000 bps • **g726r32**—G.726 32,000 bps • **g729r8**—G.729 Annex A 8,000 bps • **g729br8**—G.729 Annex B with Annex A 8,000 bps The optional *bytes* parameter sets the number of voice data bytes per frame. Acceptable values are from 10 to 240 in increments of 10 (for example, 10, 20, 30, and so on). Any other value is rounded down (for example, from 236 to 230). If you specify **g729r8**, then Internet Engineering Task Force (IETF) bit-ordering is used. For interoperability with a Cisco 2600, 3600, or AS5300 router running a Cisco IOS release earlier than Release 12.0(5)T or12.0(4)XH, you *must* specify the additional keyword **pre-ietf** after **g729r8**.

Step	Command	Purpose
8	Router(config-dialpeer)# **vad**	(Optional) This setting is enabled by default and activates voice activity detection (VAD). VAD allows the system to reduce unnecessary voice transmissions caused by unfiltered background noise.
9	Router(config-dialpeer)# **dtmf-relay** [**cisco-rtp**] [**h245-signal**] [**h245-alphanumeric**]	(Optional) Dual-tone multifrequency (DTMF) describes the tone that sounds in response to a keypress on a touch-tone phone. DTMF tones are compressed at one end of a call and decompressed at the other end. If a low-bandwidth codec, such as a G.729 or G.723 is used, the tones can sound distorted. The **dtmf-relay** command transports DTMF tones generated after call establishment out-of-band by using a method that transmits with greater fidelity than is possible in-band for most low-bandwidth codecs. Without DTMF relay, calls established with low-bandwidth codecs can have trouble accessing automated phone menu systems, such as voice mail and interactive voice response (IVR) systems. A signaling method is supplied only if the remote end supports it, and the options are: Cisco proprietary (**cisco-rtp**), standard H.323 (**h245-alphanumeric**), and H.323 standard with signal duration (**h245-signal**).
10	Router(config-dialpeer)# **fax-rate** {**2400** | **4800** | **7200** | **9600** | **12000** | **14400** | **disable** | **voice**}	(Optional) Specify the transmission speed of a fax to be sent to this dial peer. **disable** turns off fax transmission capability, and **voice** specifies the highest possible fax speed supported by the voice rate.
11	Router(config-dialpeer)# **destination-pattern** *string* [**T**]	See Step 3 in this procedure.
12	Router(config-dialpeer)# **session target** {**ipv4:*destination-address*** | **dns:**[**s.** | **d.** | **e.** | **u.**] *host-name*}	Configure the IP session target for the dial peer. **ipv4:***destination-address* indicates IP address of the dial peer. **dns:***host-name* indicates that the domain name server will resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device. There are also wildcards available for defining domain names with the keyword by using source, destination, and dialed information in the host name. For complete command syntax information, see *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.0.

Verifying Voice Dial Peers

Follow the procedure below to verify dial-peer configuration. To learn more about these commands, see *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.0.

Important command output is shown in bold.

Enter the privileged EXEC **show dial-peer voice** command. The following text is sample output from the command for a POTS dial peer:

```
router# show dial-peer voice 1
VoiceEncapPeer1
```

```
        tag = 1, dest-pat = \Q+14085551000',
        answer-address = \Q',
        group = 0, Admin state is up, Operation state is down
        Permission is Both,
        type = pots, prefix = \Q',
        session-target = \Q', voice-port =
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is "10"
        Last Disconnect Text is ""
        Last Setup Time = 0
```

The following text is sample output from the **show dial-peer voice** command for a VoIP dial peer:

```
Router# show dial-peer voice 10
VoiceOverIpPeer10
        tag = 10, dest-pat = \Q',
        incall-number = \Q+14087',
        group = 0, Admin state is up, Operation state is down
        Permission is Answer,
        type = voip, session-target = \Q',
        sess-proto = cisco, req-qos = bestEffort,
        acc-qos = bestEffort,
        fax-rate = voice, codec = g729r8,
        Expect factor = 10,Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is "10"
        Last Disconnect Text is ""
        Last Setup Time = 0
```

Configuration Examples

This section includes three sample configurations to illustrate different scenarios:

- Drop and Insert where PSTN and VoIP services are provided through the same service provider line.
- Drop and Insert where PSTN and data services are provided through the same service provider line.
- Drop and Insert where PSTN, data, and VoIP services are provided through the same service provider line.

Note For additional examples, see *Configuring Digital T1 Packet Voice Trunk Network Modules on Cisco 2600 and 3600 Series Routers.*

Drop-and-Insert technology is one way to integrate old PBX technologies with VoIP. It allows you to take 64-Kb DS0 channels from one T1 or E1 and digitally cross-connect them to 64Kb DS0 channels on another T1 or E1.

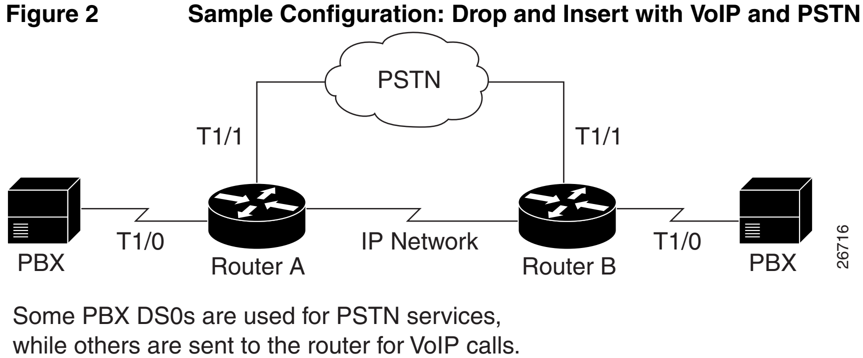
Drop and Insert allows individual 64Kb DS0 channels to be transparently passed, uncompressed, between T1/E1 ports without DSP processing. Channel traffic is sent between a PBX and CO switch or other telephony device, allowing the use, for example, of some PBX channels for long-distance service through the PSTN while the router compresses others for interoffice VoIP calls. In addition, Drop and Insert can cross-connect a telephony switch (from the CO or PSTN) to a channel bank to provide external analog connectivity.

Keep the following considerations in mind:

- Drop and Insert works only between ports on the same Multiflex VWIC.
- The VWIC can either be in a standalone WIC slot on a Cisco 2600 series router, integrated into a Digital T1 Packet Voice Trunk Network Module VWIC slot, or installed in a Cisco 3600 series 2-port network module (NM-1E2W, NM-2E2W, NM-1E1R2W).
- When the VWIC module is installed in the VWIC slot of a Digital T1 Packet Voice Trunk Network Module, the T1 ports do not provide WAN connectivity (for example. Frame-Relay, PPP, and so on) in addition to voice and Drop and Insert.
- WAN and Drop-and-Insert capabilities are supported when the VWIC is in a chassis WIC slot on a Cisco 2600 series router.

Drop and Insert with VoIP and PSTN Services

Figure 2 Sample Configuration: Drop and Insert with VoIP and PSTN Services



This example in this section shows configuration for Drop and Insert when a 2-port Multiflex VWIC is installed in a Digital T1 Packet Voice Trunk Network Module VWIC slot and VoIP is configured. WAN connections must be provided by separate links.

hostname RTR-A	hostname RTR-B
!	!
voice-card 1	voice-card 1
codec complexity high	codec complexity high
!	!
controller T1 1/0	controller T1 1/0
framing esf	framing esf
linecoding b8zs	linecoding b8zs
ds0-group 1 timeslots 1-12 type e&m-wink	ds0-group 1 timeslots 1-12 type e&m-wink
tdm-group 2 timeslots 13-24 type e&m	**tdm-group 2 timeslots 13-24 type e&m**
!	!
controller T1 1/1	controller T1 1/1
framing esf	framing esf
linecoding b8zs	linecoding b8zs
clock source line primary	clock source line primary
tdm-group 3 timeslots 13-24 type e&m	**tdm-group 3 timeslots 13-24 type e&m**
!	!
voice-port 1/0:1	voice-port 1/0:1
!	!
dial-peer voice 1 voip	dial-peer voice 1 voip
destination-pattern 4...	destination-pattern 5….
codec g723r63	codec g723r63
dtmf-relay h245-alpha	dtmf-relay h245-alpha
session target ipv4:209.165.200.253	!
session target ipv4:209.165.200.252	!
!	!
dial-peer voice 2 pots	dial-peer voice 2 pots
destination-pattern 5...	destination-pattern 4….
prefix 5	prefix 4
port 1/0:1	port 1/0:1
!	!
interface serial 0/0	interface serial 0/0
encapsulation ppp	encapsulation ppp
ip address 209.165.200.252 255.255.255.224	ip address 209.165.200.253 255.255.255.224
!	!
connect tdm1 T1 1/0 2 T1 1/1 3	**connect tdm1 T1 1/0 2 T1 1/1 3**

Clock Sources

In this example, two clock sources are available on each router's Multiflex VWIC ports: one from the PBX and one from the PSTN central office (CO). However, the clock sources must be the same, so the system adjusts to this need.

The **primary** keyword of the **clock source** command, applied to T1 1/1, means that the PSTN is providing the clock source. The T1 1/0 port connected to the PBX is automatically put into *looped-time* mode, which means that the port takes the clocking received on its Rx (receive) pair and regenerates it back on its Tx (transmit) pair. While it is receiving clocking, it does not drive the on-board clock. It is "spoofing" the port so that the connected PBX does not detect clocking that is out of synchronization, which is indicated by *slips*. The router detects the slips as controlled and does not force the port to fail.

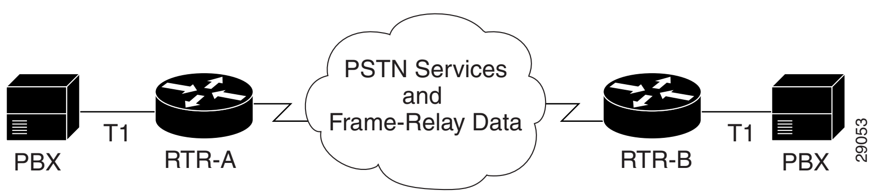
Additional Considerations

Here are some additional key points for consideration:

- The **tdm-group 2 timeslots 13-24 type e&m** command defines Drop and Insert by setting up the timeslots from each T1 that will be used in the digital cross-connect. The **type** keyword is optional, but its use is specific to the Drop and Insert feature.
  - — If you include the **type** keyword with a signaling type, the Drop-and-Insert cross-connect ensures that the specified signaling (on-hook and off-hook) is passed between the DS0s. It also uses the signaling bits to signal busy-out if one of the T1s goes down.
  - — If you do not use the **type** keyword, the Drop-and-Insert cross-connect is clear channel and does not interpret any signaling.
- The **connect tdm1 T1 1/0 2 T1 1/1 3** command activates the Drop-and-Insert digital cross-connect between the T1s. The **tdm1** portion of the command is just a name for the cross-connect, and the name can be any word, number, or series of letters.
- You can verify Drop-and-Insert connections by using the **show connection** command.

Drop and Insert with Data and PSTN Services

Figure 3 Sample Configuration: Drop and Insert with Data and PSTN Voice Services



This example in this section shows configuration for Drop and Insert when a 2-port Multiflex VWIC is installed in a Cisco 2600 series chassis slot or in a WIC slot of a Cisco 3600 series network module. Frame Relay data and PSTN voice calls travel between the PBXs, but no VoIP or VoIP over Frame Relay information is carried.

Clock Sources

As in the previous example, two clock sources are available on each router's Multiflex VWIC ports: one from the PBX and one from the PSTN central office (CO). However, the clock sources must be the same, so the system adjusts to this need.

The primary clock source is T1 or E1 1/0, connected to the PSTN, and its clock is a reference for T1 or E1 1/1. If T1 1/0 fails, the clock source to drive T1 or E1 1/1 is generated from the line to the PBX.

Additional Considerations

The **channel-group 0** command is configured in such a way that the service provider can send Frame-Relay link management information (LMI) on T1 channels 13 through 24 (17 through 31 on E1) for Frame-Relay data services. This command automatically creates interface serial 1/0:0.

Interface serial 1/0:0 is where all WAN and Layer 3 protocol details are configured, for example, Frame Relay encapsulation or IP addresses.

T1 Configuration

```
hostname RTR-A
!
controller T1 1/0
framing esf
linecoding b8zs
clock source line primary
tdm-group 1 timeslots 1-12
channel-group 0 timeslots 13-24
!
controller T1 1/1
framing esf
linecoding b8zs
clock source line
tdm-group 2 timeslots 1-12
!
interface serial 1/0:0
encapsulation frame-relay
!
interface serial 1/0:1.1
ip address 209.165.200.252 255.255.255.224
frame-relay interface-dlci 100 br
!
interface ethernet 0
ip address 209.165.200.250 255.255.255.224
!
router eigrp 1
network 209.165.200.224
!
connect tdm1 T1 1/0 1 T1 1/1 2
```

```
hostname RTR-B
!
controller T1 1/0
framing esf
linecoding b8zs
clock source line primary
tdm-group 1 timeslots 1-12
channel-group 0 timeslots 13-24
!
controller T1 1/1
framing esf
linecoding b8zs
clock source line
tdm-group 2 timeslots 1-12
!
interface serial 1/0:0
encapsulation frame-relay
!
interface serial 1/0:1.1
ip address 209.165.200.253 255.255.255.224
frame-relay interface-dlci 100 br
!
interface ethernet 0
ip address 209.165.201.1 255.255.255.224
!
router eigrp 1
network 209.165.200.224
network 209.165.201.0
!
connect tdm1 T1 1/0 1 T1 1/1 2
```

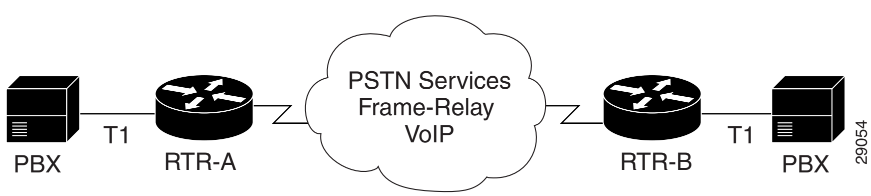
E1 Configuration.

```
hostname RTR-A
!
controller E1 1/0
framing crc4
linecoding hdb3
clock source line primary
tdm-group 1 timeslots 1-15
channel-group 0 timeslots 17-31
!
controller E1 1/1
framing crc4
linecoding hdb3
clock source line
tdm-group 2 timeslots 1-15
!
interface serial 1/0:0
encapsulation frame-relay
!
interface serial 1/0:1.1
ip address 209.165.200.252 255.255.255.224
frame-relay interface-dlci 100 br
!
interface ethernet 0
ip address 209.165.200.250 255.255.255.224
!
router eigrp 1
network 209.165.200.224
!
connect tdm1 T1 1/0 1 T1 1/1 2
```

```
hostname RTR-B
!
controller E1 1/0
framing crc4
linecoding hdb3
clock source line primary
tdm-group 1 timeslots 1-15
channel-group 0 timeslots 17-31
!
controller E1 1/1
framing crc4
linecoding hdb3
clock source line
tdm-group 2 timeslots 1-15
!
interface serial 1/0:0
encapsulation frame-relay
!
interface serial 1/0:1.1
ip address 209.165.200.253 255.255.255.224
frame-relay interface-dlci 100 br
!
interface ethernet 0
ip address 209.165.201.1 255.255.255.224
!
router eigrp 1
network 209.165.200.224
network 209.165.201.0
!
connect tdm1 T1 1/0 1 T1 1/1 2
```

Drop and Insert with PSTN, Data, and VoIP Services

Figure 4 Sample Configuration:Drop and Insert with PSTN, Data, and VoIP Services



This configuration shows how to use some T1 channels for passing voice from the PSTN to the PBX, and some channels for data services that also pass VoIP traffic. This setup requires both a Digital T1 Packet Voice Trunk Network Module with a Multiflex VWIC installed and a separate Multiflex VWIC.

Clock Sources

The primary clock source is T1 1/0, and its clock is a reference for T1 1/1. If T1 1/0 fails, the clock source to drive T1 1/1 is generated internally.

```
hostname RTR-A
!
controller T1 1/0
description - NM-HDV connected to PBX
framing esf
linecoding b8zs
clock source internal
tdm-group 1 timeslots 1-12
ds0-group 2 timeslots 13-24 type e&m-wink
!
controller T1 1/1
description - xconnect to VWIC T1
framing esf
linecoding b8zs
clock source line
tdm-group 2 timeslots 1-12
!
controller T1 2/0
description - connected to TELCO WAN
framing esf
linecoding b8zs
channel-group 0 timeslots 13-24
tdm-group 3 timeslots 1-12
clock source line
!
controller T1 2/1
description - xconnect to NM-HDV
framing esf
linecoding b8zs
clock source internal
tdm-group 4 timeslots 1-12
!
voice-port 1/0:2
!
interface serial 2/0:0
encapsulation frame-relay
!
interface serial 1/0:0.1
ip address 209.165.200.252 255.255.255.224
frame-relay interface-dlci 100 br
!
interface ethernet 0
ip address 209.165.200.250 255.255.255.224
!
router eigrp 1
network 209.165.200.224
!
dial-peer voice 1 voip
destination-pattern 5...
session target ipv4:209.165.200.253
!
dial-peer voice 2 pots
destination-pattern 4...
prefix 4
prefix 5
port 1/0:2
port 1/0:2
!
connect tdm1 T1 1/0 1 T1 1/1 2
connect tdm2 T1 2/0 3 T1 2/1 4
```

```
hostname RTR-B
!
controller T1 1/0
description -  NM-HDV connected to PBX
framing esf
linecoding b8zs
clock source internal
tdm-group 1 timeslots 1-12
!
controller T1 1/1
description - xconnect to VWIC T1
framing esf
linecoding b8zs
clock source line
tdm-group 2 timeslots 1-12
!
!
controller T1 2/0
description - connected to TELCO WAN
framing esf
linecoding b8zs
channel-group 0 timeslots 13-24
tdm-group 3 timeslots 1-12
clock source line
!
controller T1 2/1
description - xconnect NM-HDV
framing esf
linecoding b8zs
clock source internal
tdm-group 4 timeslots 1-12
!
voice-port 1/0:2
!
interface serial 2/0:0
encapsulation frame-relay
!
interface serial 1/0:0.1
ip address 209.165.200.253 255.255.255.0
frame-relay interface-dlci 100 br
!
interface ethernet 0
ip address 209.165.201.1 255.255.255.224
!
router eigrp 1
network 209.165.200.224
network 209.165.201.0
!
dial-peer voice 1 voip
destination-pattern 4...
session target ipv4:209.165.200.252
!
dial-peer voice 2 pots
destination-pattern 5...
!
connect tdm1 T1 1/0 1 T1 1/1 2
connect tdm2 T1 2/0 3 T1 2/1 4
```

Additional Considerations

The following connections are made by using channels 1 through 12 from the service provider:

- The channels are brought into the Multiflex VWIC that is not installed in the Digital T1 Packet Voice Trunk Network Module.
- These 12 channels cross-connect to the other Multiflex VWIC port.
- From there, an external T1 crossover cable cross-connects the channels to the first T1 port on the Digital T1 Packet Voice Trunk Network Module.
- The 12 channels cross-connect to the other T1 port on the Digital T1 Packet Voice Trunk Network Module and out to the connected PBX.

Channels 13 through 24 pass Frame-Relay LMI from the service provider for data services, and the channels terminate on the Multiflex VWIC channel group. This serial interface is used for data traffic from the Ethernet, as well as VoIP traffic that originates on channels 13 through 24 from the PBX connected to the Digital T1 Packet Voice Trunk Network Module.

Command Reference

New or modified commands are included in the Command Reference section of *Configuring Digital T1 Packet Voice Trunk Network Modules on Cisco 2600 and 3600 Series Routers.* All other commands used with this feature are documented in the Cisco IOS Release 12.0 command references.

Glossary

AMI—alternate mark inversion. Line-code type used on T1 and E1 circuits. In AMI, zeros are represented by 01 during each bit cell, and ones are represented by 11 or 00, alternately, during each bit cell. AMI requires that the sending device maintain ones density, which is not maintained independent of the data stream. Sometimes called *binary coded alternate mark inversion*.

ATM—Asynchronous Transfer Mode. International standard for cell relay where multiple service types (such as voice, video, or data) are conveyed in fixed-length (53-byte) cells that allow cell processing to occur in hardware; thereby transit delays are reduced. ATM is designed to take advantage of high-speed transmission media such as E3, SONET, and T3.

B8ZS—binary 8-zero substitution. Line-code type, used on T1 and E1 circuits, in which a special code is substituted whenever 8 consecutive zeros are sent over the link. This code is then interpreted at the remote end of the connection. This technique guarantees that ones density is independent of the data stream.

CAS—channel-associated signaling. Trunk signaling (for example, in a T1 line) in which control signals, such as those for synchronizing and bounding frames, are carried in the same channel along with voice and data signals.

CCS—common channel signaling. Trunk signaling (for example, using Primary Rate Interface) in which a control channel carries signaling for separate voice and data channels.

CO—central office. Local telephone company office where all local loops in a given area connect and circuit switching of subscriber lines occurs.

codec—Coder-decoder. Device that typically uses pulse code modulation to transform analog signals into a digital bit stream and digital signals back into analog.

DTMF—Dual-tone multifrequency. Use of two simultaneous voice-band tones for dialing (such as touch tone).

Drop and Insert—(also called *TDM Cross-Connect*) Allows DSO channels from one T1 or E1 facility to be digitally cross-connected to DS0 channels on another T1 or E1. By using this method, channel traffic is sent between a PBX and CO PSTN switch or other telephony device, so that some PBX channels are directed for long-distance service through the PSTN while the router compresses others for interoffice VoIP calls. In addition, Drop and Insert can cross-connect a telephony switch (from the CO or PSTN) to a channel bank for external analog connectivity.

DSP—digital signal processor.

E1—European digital carrier facility used for transmitting data through the telephone hierarchy. The transmission rate for E1 is 2.048 megabits per second (Mbps).

E&M—rEceive and transMit, or Ear and Mouth. Type of signaling originally developed for analog two-state voltage telephony using the ear and mouth leads; in digital telephony, uses two bits.

ESF—Extended Superframe. Framing type used on T1 circuits that consists of 24 frames of 192 bits each with the 193rd bit providing timing and other functions. ESF is an enhanced version of SF format.

FDL—Facility Data Link. A 4-Kbps channel, provided by the Extended SuperFrame (ESF) T1 framing format. The FDL performs outside the payload capacity and allows a service provider to check error statistics on terminating equipment, without intrusion.

FXO—Foreign Exchange Office. A voice interface emulating a PBX trunk line to a switch or telephone equipment to a PBX extension interface.

FXS—Foreign Exchange Station. A voice interface for connecting telephone equipment; it emulates the extension interface of a PBX or the subscriber interface for a switch.

HBD3—High-Density Bipolar 3. Line code type used on E1 circuits.

IETF—Internet Engineering Task Force.

ISDN—Integrated Services Digital Network. Communication protocol offered by telephone companies. ISDN permits telephone networks to carry data, voice, and other source traffic.

packet—Logical grouping of information that includes a header containing control information and (usually) user data. Packets are most often used to refer to network layer units of data.

POTS—plain old telephone service.

PDVM—packet data voice module.

PSTN—Public Switched Telephone Network. General term referring to the variety of telephone networks and services in place worldwide.

QoS—quality of service. Measure of performance for a transmission system that reflects its transmission quality and service availability.

SF—Super Frame. Common framing type used on T1 circuits. SF consists of 12 frames of 192 bits each with the 193rd bit providing error checking and other functions. SF is superseded by ESF, but is still widely used. Also called *D4 framing*.

SNMP—Simple Network Management Protocol. Network management protocol used almost exclusively in TCP/IP networks. SNMP provides a means to monitor and control network devices, and to manage configurations, statistics collection, performance, and security.

T1—Digital WAN carrier facility. T1 transmits DS 1-formatted data at 1.544 Mbps through the telephone switching network by using alternate mark inversion or B8ZS coding.

T1 trunk—Digital WAN carrier facility. See *T1*.

TDM—time-division multiplexing.

Trunk—Physical and logical connection between two switches across which network traffic travels. A backbone is composed of a number of trunks.

UNI—User-Network Interface. ATM Forum specification that defines an interoperability standard for the interface between ATM-based products (a router or an ATM switch) located in a private network, and the ATM switches located within the public carrier networks. Also used to describe similar connections in Frame Relay networks.

VAD—voice activity detection.