



MIX-Enabled Multichannel T1/E1 Port Adapter

The Multiservice Interchange (MIX) port adapter (PA-MCX) adds time-division multiplexing (TDM) connection capabilities so that you can combine types of traffic traveling through Cisco 7200 VXR series routers. On a Cisco 7200 VXR series router, MIX enables TDM connections between two ports on the same MIX-enabled port adapter.

MIX functions permit connection of TDM streams to support applications that are sensitive to time delay, such as voice and video, and provide customers the flexibility to manage this traffic through the router as traditional TDM connections or in a packet-based format. This document includes the following sections:

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Feature Overview

The PA-MCX port adapters are multichannel port adapters that provide T1/E1 connectivity for data and voice traffic. The PA-MCX ports can be configured either as DS1/PRI ports for data traffic, or as packet voice ports that, when used in conjunction with a digital voice port adapter, allow Cisco 7200 VXR routers to become dedicated packet voice hubs or packet voice gateways that connect to both private branch exchanges (PBXs) and the Public Switched Telephone Network (PSTN). This allows packet voice and packet fax calls to be placed over the WAN and sent through the gateway into the traditional circuit-switched voice infrastructure.

The following port adapters are currently MIX-enabled:

- PA-MCX-2TE1
- PA-MCX-4TE1
- PA-MCX-8TE1
- PA-VXC-2TE1

MIX functionality is used to support the extended drop-and-insert application, and the DSP Farm feature which are described below:

- Drop-and-Insert Feature
- DSP Farm Feature

Drop-and-Insert Feature

Drop-and-insert capability allows digital signal level 0 (DS0s) from several T1/E1 interfaces (such as voice, data, or video) to be aggregated on one channelized interface and routed across the network or handed off to a Public Switched Telephone Network (PSTN). Previously, separate T1/E1s were required for data and voice streams, but with MIX functionality both types of traffic can be handled in separate channels on the same T1/E1 carrier. For most operations, this means fewer T1/E1s are needed, thereby reducing operating costs.

Another benefit of drop-and-insert functionality is an increase in the flexibility and choice that administrators can exercise over different types of traffic in their networks. With the availability of both TDM and packet streams on the Cisco 7200 VXR series routers, all voice, video, and data can be routed over the same platform, thus keeping infrastructure costs down.

Time-sensitive traffic such as video and voice can be managed as TDM streams, with traditional quality of service (QoS) considerations. All the slots participating in the drop-and-insert function are synchronized across the TDM backplane to the same clock to prevent clock slips that can cause frame loss. To synchronize the data flow of these delay-sensitive data types, clocking can be obtained from internal or external sources. Gradually, voice and video traffic can be migrated to packet-based formats without having to invest in new equipment.

Drop-and-insert technology is one way to integrate old PBX technologies with packetized voice. With drop and insert, you can take 64 Kb DS0 channels from one T1/E1 and digitally cross connect them to 64 Kb DS0 channels on another T1 or E1. Drop-and-insert functionality is sometimes called TDM cross connect.

With drop-and-insert functionality, individual 64 Kb DS0 channels can be transparently passed and uncompressed between T1 or E1 ports without passing through a DSP. Using this method, the channel traffic is sent between a PBX and central office 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 central office [CO] or PSTN) to a channel bank to provide external analog connectivity.

DSP Farm Feature

The DSP Farm feature enables a router to become an integrated voice and data solution providing port adapter support to more voice terminations. The DSP-farm capabilities of the digital voice port adapters provide a high-density voice and fax gateway solution to connect PBXs, the PSTN, and the WAN to create a multiservice network.

With MIX functionality, DSP resources are pooled into a farm so that they can be used by TDM streams regardless of the slot or port from which the streams originate. TDM streams carrying voice are routed to the appropriate DSP resources and packetized voice is sent out over the data network.

Membership in the DSP farm is optional. You can include a DSP resource in a DSP farm or reserve it to be used only on traffic from its own interfaces.

Benefits

The MIX-enabled 2/4/8 Port Multichannel T1/E1 port adapter provides the following benefits:

- Large scale voice and data termination—A maximum of 48 T1/E1 data interfaces can be supported on a Cisco 7206 VXR router.
- Drop-and-Insert functionality—DS0 drop-and-insert functionality between any interface within the port adapter provides efficient traffic aggregation and reduction in WAN links.
- DSP Farm—PA-VXC-2TE1+ port adapters in Cisco 7200 VXR series routers can share DSP resources providing low-cost, high-density voice termination. Twenty T1 or sixteen E1 voice interfaces can be terminated on a single Cisco 7206 VXR router.

Related Features and Technologies

- PA-MC-T1—in data mode
- PA-MC-E1—in data mode
- PA-VXC-TE1+

Related Documents

- *MIX Enabled Multichannel T1/E1 Port Adapter Installation and Configuration*
- *Configuring T1/E1 High Capacity Digital Voice Port Adapters*

Supported Platforms

- Cisco 7200 VXR series routers

Supported Standards, MIBs, and RFCs

Standards

The following standards are supported by the PA-MCX when used together with a PA-VXC-2TE1 for voice applications only:

- FRF.11 Voice over Frame Relay
- H.323 Version 2 Voice over IP including RAS, Fast Connect, codec negotiation
- ITU standard compression algorithms (G.703, G.704, G.729, G.723.1, G.729a/b, G.711, G.728, G.726)

MIBs

- CISCO-VOICE-DIAL-CONTROL-MIB

To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB web site on Cisco Connection Online (CCO) at <http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml>.

RFCs

- RFC1406, *Definitions of Managed Objects for the DS1 and E1 Interface Types*

Prerequisites

- Multiservice Interchange (MIX) card installed on a Cisco 7200 VXR series router
- Cisco IOS Release 12.1(5)T installed

Configuration Tasks

See the following sections for configuration tasks for the MIX port adapter feature. Each task in the list is identified as either optional or required.


- Configuring Card Type and T1 Controller Settings (Required)
- Configuring Card Type and E1 Controller Settings (Required)
- Configuring the Service Type (Required)
- Configuring a Voice Port (Required only for voice applications)
- Configuring Voice Dial Peers (Required for voice applications only)
- Configuring Network Clock Use (Required only for voice applications)

Configuring Card Type and T1 Controller Settings

The following steps specify codec settings for card types and set up T1 controllers for clocking and other T1 parameters, as well as for DS0 groups that define the channels for compressed voice and TDM groups for drop-and-insert capability.

	Command	Purpose
Step 1	Router# configure terminal	Enter global configuration mode.
Step 2	Router(config)# card type t1 slot	Enter T1 card type and specify the slot location by using a value from 0 to 5, depending upon your router.
Step 3	Router(config)# controller T1 slot/port	Enter controller configuration mode for the T1 controller at the specified slot/port location.
Step 4	Router(config-controller)# clock source {line internal}	Configure controller T1 1/0 to 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.
Step 5	Router(config-controller)# framing {sf esf}	Set the framing according to your service provider's instructions. Choose Extended Superframe (ESF) format or Super Frame (SF) format.
Step 6	Router(config-controller)# linecode {b8zs ami}	Set the line encoding according to your service provider's instructions. <ul style="list-style-type: none"> Bipolar-8 zero substitution (B8ZS) encodes a sequence of eight zeros in a unique binary sequence to detect line coding violations. Alternate mark inversion (AMI) 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.

Command	Purpose
<p>Step 7</p> <pre>Router(config-controller)# cablelength long {-15db -22.5db -7.5db 0db} or Router(config-controller)# cablelength short {110ft 220ft 330ft 440ft 550ft 600ft}</pre>	<p>(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.</p> <p>To set a cable length longer than 600 feet for a T1 link, use the cablelength long command. The keywords are as follows:</p> <ul style="list-style-type: none"> • -15db—specifies the decibel pulse level at -15 dB. • -22.5db—specifies the decibel pulse level at -22.5 dB. • -7.5db—specifies the decibel pulse level at -7.5 dB. • 0db—specifies the decibel pulse level at 0 dB. This is the default pulse rate. <p>To set a cable length 600 feet or less for a T1 link, use the cablelength short command. There is no default for cablelength short. The keywords are as follows:</p> <ul style="list-style-type: none"> • 110ft—specifies a cable length from 0 to 110 feet. • 220ft—specifies a cable length from 111 to 220 feet. • 330ft—specifies a cable length from 221 to 330 feet. • 440ft—specifies a cable length from 331 to 440 feet. • 550ft—specifies a cable length from 441 to 550 feet. • 600ft—specifies a cable length from 551 to 600 feet. <p>If you do not set the cable length, the system defaults to a setting of cablelength long 0db.</p>

	Command	Purpose
Step 8	Router(config-controller)# tdm-group <i>tdm-group-no</i> timeslots <i>timeslot-list</i> type	(Optional) Use this command only when you need TDM channel groups for the drop-and-insert (also called TDM cross-connect) function with a two-port T1 trunk multiplex interface card. <ul style="list-style-type: none"> <i>tdm-group-no</i>—a value from 1 to 31 that identifies the channel group. <i>timeslot-list</i>—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.  <p>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.</p>
Step 9	Router(config-controller)# no shutdown	Activates the controller.
Step 10	Router(config-controller)# exit	Exit controller configuration mode. Skip the next step if you are not setting up the drop-and-insert capability.
Step 11	Router(config)# connect <i>id</i> T1 <i>slot/port</i> <i>tdm-group-no-1</i> T1 <i>slot/port</i> <i>tdm-group-no-2</i>	(Optional) This global configuration command sets up the connection between two T1 TDM groups of timeslots on the trunk interfaces for the drop-and-insert capability. <i>id</i> —a name for the connection. Identify each T1/E1 controller by its <i>slot/port</i> location. Valid values for slot and port are 0 and 1. <i>tdm-group-no-1</i> and <i>tdm-group-no-2</i> —identify the TDM group numbers (from 1 to 31) on the specified controller. The groups were set up in Step 9.

Configuring Card Type and E1 Controller Settings


The following steps specify codec settings for card types and set up E1 controllers for clocking and other E1 parameters, as well as for DS0 groups that define the channels for compressed voice and TDM groups for drop-and-insert capability.

	Command	Purpose
Step 1	Router# configure terminal	Enter global configuration mode.
Step 2	Router(config)# card type e1 <i>slot</i>	Enter E1 card type and specify the slot location by using a value from 0 to 5, depending upon your router.

	Command	Purpose
Step 3	Router(config)# controller E1 slot/port	Enter controller configuration mode for the E1 controller at the specified <i>slot/port</i> location. Valid values for <i>slot</i> and <i>port</i> are 0 and 1.
Step 4	Router(config-controller)# clock source {line internal}	Configure controller E1 1/0 to 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.
Step 5	Router(config-controller)# framing {crc4 no crc4}	Set the framing according to your service provider's instructions. Choose CRC4 format or NO CRC4 format.
Step 6	Router(config-controller)# linecode {hdb3 ami}	Set the line encoding according to your service provider's instructions.
Step 7	Router(config-controller)# tdm-group tdm-group-no timeslots timeslot-list	<p>(Optional) Use this command only when you need TDM channel groups for the Drop-and-Insert (also called TDM Cross-Connect) function with a two-port T1/E1 multiflex trunk interface card.</p> <p><i>tdm-group-no</i> is a value from 1 to 31 that identifies the channel group.</p> <p><i>timeslot-list</i> 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 or E1, allowable values are from 1 to 24.</p> <p>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.</p>
Step 8	Router(config-controller)# no shutdown	Activate the controller.


Configuring the Service Type


Because the PA-MXC can be used for either voice or data connections, you must configure the T1/E1 controller for either data or voice using the **service type** controller configuration command:

	Command	Purpose
Step 1	Router(config)# controller [e1 t1] slot/port	Enter controller configuration mode for the T1 or E1 controller at the specified <i>slot/port</i> location.
Step 2	Router(config-controller)# service-type {data cas-voice ccs-voice}	<p>Configures the port as a data-only port or a TDM cross-connect to a digital voice port adapter.</p> <ul style="list-style-type: none"> • data—data port only • cas-voice—voice port with channel associated signaling (CAS) <p> Note When a port is set to CAS voice mode, 64Kbs data channel-groups can also be configured on that same T1 port.</p> <ul style="list-style-type: none"> • ccs-voice—voice port with common channel signalling (CCS)

Configuring a Voice Port



Follow these steps to set up voice ports to support the local and remote stations. Not all possible commands are shown here. To learn more, see *Voice, Video, and Home Applications Configuration Guide* and *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.1.

	Command	Purpose
Step 1	Router# configure terminal	Enter global configuration mode.
Step 2	Router(config)# voice-port <i>slot/port:ds0-group-no</i>	<p>Enter voice-port configuration mode.</p> <ul style="list-style-type: none"> • <i>slot</i>—the router location where the voice port adapter is installed. Valid entries are from 0 to 3. • <i>port</i>—the voice interface card location. Valid entries are 0 or 1. <p>Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.</p> <p> Note This voice-port command syntax does not apply to analog voice port adapters and voice interface cards. The latter are specified using <i>slot/subunit/port</i>, designating the router slot for the voice port adapter, the location of the voice interface card in the port adapter, and the port on the voice interface card.</p>
Step 3	Router(config-voice-port)# busyout monitor interface <i>interface number</i>	<p>(Optional) This command allows you to specify a LAN or WAN interface that will be monitored, and, when it is down, trigger a busyout (off-hook) state on the voice port. This allows rerouting of calls. For example, if you specify <i>serial 1/0</i> 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.</p> <p>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.</p>
Step 4	Router(config-voice-port)# comfort-noise	<p>(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 resulting silence can fool the caller into thinking the call is disconnected instead of being merely idle.</p>

	Command	Purpose
Step 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.
Step 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.
Step 7	Router(config-voice-port)# connection {plar trunk} <i>string</i>	(Optional) This command sets up a connection mode for the voice port. <ul style="list-style-type: none"> • plar—specifies a private line, automatic Ringdown (PLAR) connection, which rings a remote telephone when the dial peer goes off hook. • trunk—specifies a straight tie-line connection to a PBX. • <i>string</i>—specifies the remote telephone number or significant start digits of the number.
Step 8	Router(config-voice-port)# timeouts interdigit <i>seconds</i>	(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. <p> 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 12.1 <i>Voice, Video, and Home Applications Configuration Guide</i>.</p>
Step 9	Router(config-voice-port)# exit	Exit voice-port configuration mode. Repeat Steps 2 through 9 for each DS0 group you create.

Configuring Voice Dial Peers

Follow these steps to set up voice dial peers to support the local and remote stations. Not all possible commands are shown here. To learn more, see *Voice, Video, and Home Applications Configuration Guide and Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.1.

	Command	Purpose
Step 1	Router# configure terminal	Enter global configuration mode.
Step 2	Router(config)# dial-peer voice <i>number</i> pots	Enter dial-peer configuration mode and define a local dial peer that will connect to the plain old telephone service (POTS) network. <ul style="list-style-type: none"> • <i>number</i>—one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647. • pots—a peer using basic telephone service.
Step 3	Router(config-dialpeer)# destination-pattern <i>string</i> [T]	Configure the dial peer's destination pattern so that the system can reconcile dialed digits with a telephone number. <ul style="list-style-type: none"> • <i>string</i>—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. The following special characters can be entered: <ul style="list-style-type: none"> – 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. <p>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 #).</p> <p> _____</p> <p>Note The timer character must be a capital T.</p>
Step 4	Router(config-dialpeer)# prefix <i>string</i>	(Optional) Include a dial-out prefix that the system enters automatically instead of people dialing it. <ul style="list-style-type: none"> • <i>string</i>—a value from 0 to 9, and you can use a comma (,) to indicate a pause. <p> _____</p> <p>Note There are other digit manipulation commands available to handle such situations as prefixes for special services, ignoring some digits, and dialing into remote PBXs as though they are local.</p>

Command	Purpose
Step 5 Router(config-dialpeer)# port <i>slot/port:ds0-group-no</i>	This command associates the dial peer with a specific logical interface. <ul style="list-style-type: none"> • <i>slot</i>—the router location where the voice port adapter is installed. Valid entries are from 0 to 3. • <i>port</i>—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/E1 card.
Step 6 Router(config)# dial-peer voice <i>number</i> voip	Enter dial-peer configuration mode and define a remote VoIP dial peer. <ul style="list-style-type: none"> • <i>number</i>—one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647. • voip—a VoIP peer using voice encapsulation on the IP network.

Command	Purpose
Step 7 Router(config-dialpeer)# codec { g711alaw g711ulaw g723ar53 g723ar63 g723r53 g723r63 g726r16 g726r24 g726r32 g728 g729r8 [pre-ietf] g729br8 } [<i>bytes</i>]	<p>The voice-card configuration codec command sets the codec options that are available when you execute this command. If you do not set codec complexity, g729r8 with IETF bit-ordering is used.</p> <p>If you set codec complexity to high, the following options are available:</p> <ul style="list-style-type: none"> • 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 <p>If you set codec complexity to medium, the following options are valid:</p> <ul style="list-style-type: none"> • 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 <p>The optional <i>bytes</i> 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).</p> <p>If you specify g729r8, then the IETF (Internet Engineering Task Force) bit-ordering is used. For interoperability with a Cisco 7200 series router running a Cisco IOS release prior to Release 12.0(5)T or 12.0(4)XH, you <i>must</i> specify the additional key word pre-ietf after g729r8.</p>

	Command	Purpose
Step 8	Router(config-dialpeer)# vad	(Optional) This setting is enabled by default. It activates voice activity detection (VAD). VAD allows the system to reduce unnecessary voice transmissions caused by unfiltered background noise.
Step 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 may 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).
Step 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.
Step 11	Router(config-dialpeer)# destination-pattern <i>string</i> [T]	See Step 3 in this procedure.
Step 12	Router(config-dialpeer)# session target { ipv4:destination-address dns:[\$\$\$. \$d\$. \$e\$. \$u\$.] <i>host-name</i> }	Configure the IP session target for the dial peer. <ul style="list-style-type: none"> • ipv4:destination-address—IP address of the dial peer. • dns:host-name—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. <p>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 <i>Voice, Video, and Home Applications Command Reference</i> for Cisco IOS Release 12.1.</p>

Configuring Network Clock Use

Time delay-sensitive traffic such as voice and video requires cross-connected port adapters to synchronize their data flow. When you configure network clocking, you can select which modules participate in clocking that is synchronized across the TDM backplane and what the source of that network clock will be. You can specify several levels of priority, so if your highest priority is unavailable, your second choice takes over automatically. Potential clock sources include internal generation from a port adapter phase-locked loop (PLL) or external retrieval from PSTN or PBX clocks over trunk lines.

	Command	Purpose
Step 1	Router(config)# frame-clock-select <i>slot number</i>	Allows the specified port adapter to participate in network clocking.
Step 2	Router(config-if)# frame-clock-select <i>priority controller</i>	Establishes the source and priority of the clocking on the TDM backplane and, in turn, all port adapters participating in network clocking.

Troubleshooting Tips

The table below lists **debug** and **show** commands to help analyze problems with your configuration.

Command	Purpose
show diag [<i>slot-number</i>] [<i>details</i>] [<i>summary</i>]	In privileged EXEC mode, displays hardware information on the line cards, including DRAM and static RAM (SRAM). See the following example.
debug conn	In privileged EXEC mode, displays information from the connection manager, the TDM client, and the digital signal processor (DSP) client.

The following example shows output for the **show diag** command for slot 2:

```
Router# show diag 2
Slot 2:
  PA-MCX-8TE1 Port adapter, 8 ports
  Port adapter is analyzed
  Port adapter insertion time 6d00h ago
  EEPROM contents at hardware discovery:
  Hardware Revision       : 1.0
  PCB Serial Number      : MIC04291ZS4
  Part Number            : 73-4118-03
  Board Revision         : 01
  RMA Test History       : 00
  RMA Number             : 0-0-0-0
  RMA History            : 00
  Deviation Number       : 0-0
  Product Number        : PA-MCX-8TE1
  Top Assy. Part Number  : 800-05358-03
  EEPROM format version 4
  EEPROM contents (hex):
  0x00: 04 FF 40 01 7A 41 01 00 C1 8B 4D 49 43 30 34 32
  0x10: 39 31 5A 53 34 82 49 10 16 03 42 30 31 03 00 81
  0x20: 00 00 00 00 04 00 80 00 00 00 00 CB 94 50 41 2D
  0x30: 4D 43 58 2D 38 54 45 31 20 20 20 20 20 20 20 20
  0x40: 20 C0 46 03 20 00 14 EE 03 FF FF FF FF FF FF FF
  0x50: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
  0x60: 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
  0x70: 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
```

Monitoring and Maintaining MIX Connections

To monitor MIX settings and connections, use the following commands.

Command	Purpose
<code>show connections all</code>	Shows detailed information about the connections created by the connect command.
<code>show controllers</code>	Displays all network modules and their interfaces.
<code>show network clocks</code>	Displays the list of all sources of network clocks according to their priority, as well as the currently chosen clock.

Configuration Examples

This section provides a configuration example of drop-and-insert of voice or data.

Drop-and-Insert of Voice or Data

This example shows drop-and-insert of time slots 13–24 of T1 1/0 to time slots 1–12 of T1 1/1. The TDM groups are defined by the **tdm-group** command. The drop-and-insert connection is made by the **connect tdm** command. The time slots can carry either voice or data. When they carry voice in CAS applications, the **service-type cas-voice** controller configuration command must be issued first.

- The **tdm-group 2 timeslots 13-24** command defines drop-and-insert capability by setting up the time slots from each T1/E1 that are used in the digital cross-connect.
- The **connect tdm1** command activates the drop-and-insert digital cross connect between the T1s. The **tdm1** portion of the command is 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 connect** command.

```
hostname RTR-A
!
card type t1 1
!

controller T1 1/0
 service-type cas-voice
 clock source line
 framing esf
 linecoding b8zs
 ds0-group 1 timeslots 1-12 type e&m-wink
 tdm-group 2 timeslots 13-24
!
controller T1 1/1
 service-type cas-voice
 clock source line
 framing esf
 linecoding b8zs
 tdm-group 3 timeslots 1-12
!
!
connect tdm1 T1 1/0 2 T1 1/1 3
```

Command Reference

This section documents new commands. All other commands used with this feature are documented in the Cisco IOS Release 12.1 command reference publications.

- **reserve**
- **service-type**

reserve

To specify how many DSP resources on a DSP farm-type interface are locally reserved, use the **reserve** DSP interface configuration command. To disable reservation, use the **no** form of this command.

reserve *0-30*

no reserve

Syntax Description	<i>0-30</i>	Number of DSP resources to be reserved.
---------------------------	-------------	---

Defaults No DSP farm-type interfaces are reserved.

Command Modes DSP interface configuration

Command History	Release	Modification
	12.1(5)T	This command was introduced.

Usage Guidelines Using the **reserve** command to reserve DSP resources on a DSP farm-type interface prevents these resources from being used by external ports. It is not generally necessary to do this.

Examples In the following example, four DSP resources are locally reserved on a DSP farm-type interface:

```
Router (dsp-interface)# reserve 4
```

service-type

To specify whether the port is to be configured as a data-only port or for a TDM cross-connect to a digital voice port adapter, use the **service-type** configuration controller command. To disable configuration, use the **no** form of this command.

service-type { **data** | **cas-voice** | **ccs-voice** }

no service-type { **data** | **cas-voice** | **ccs-voice** }

Syntax Description

data	Data-only port.
cas-voice	Voice port with CAS signalling.
ccs-voice	Voice port with CCS signalling.

Defaults

ccs-voice

Command Modes

Configuration controller

Command History

Release	Modification
12.1(5)T	This command was introduced.

Usage Guidelines

Use the **service type data** command to retain the maximum compatibility with traditional multichannel port adapters such as the PA-MC-T1 and PA-MC-E1. In this mode, no voice traffic is supported. You can configure the clock source individually for each port.

Use the **service type ccs-voice** command to connect to telephone equipment that supports common channel signalling such as ISDN PRI or QSIG. All ports configured in this mode must use a common clock source to avoid framing slips. Use the **frame-clock select** global configuration command to configure a port as the primary clock source. Use the **clock source internal** controller configuration command to configure the other ports to derive an internal clock source.

Use the **service type cas-voice** command to connect to connect to telephone equipment that uses CAS signalling such as T1 CAS or E1 R2. When a port is set to CAS voice mode, 64Kbs data channel-groups can also be configured on that same T1 port.

CAS-Voice mode preserves ABCD signalling across channels for drop and insert applications. All ports configured in this mode must use a common clock source to avoid framing slips. Use the **frame-clock select** global configuration command to configure a port as the primary clock source. Use the **clock source internal** controller configuration command to configure the other ports to derive an internal clock source.

Examples

The following example configures the second port in slot 1 as a voice port with CAS signalling:

```
Router(config)# controller t1 1/1
Router(config-controller)# service-type cas-voice
```

Related Commands

clock source {line | internal}Configures the T1/E1 controller to specify the clock source.

Glossary

AAL—ATM Adaptation Layer. Service-dependent sublayer of the data link layer. The AAL accepts data from different applications and presents it to the ATM layer in the form of 48-byte ATM payload segments. AALs consist of two sublayers: convergence sublayer (CS) and segmentation and reassembly (SAR). AALs differ on the basis of the source-destination timing used, whether they use constant bit rate (CBR) or variable bit rate (VBR), and whether they are used for connection-oriented or connectionless mode data transfer. At present, the four types of AAL recommended by the ITU-T are AAL1, AAL2, AAL3/4, and AAL5.

AAL1—ATM adaptation layer 1. One of four AALs recommended by the ITU-T. AAL1 is used for connection-oriented, delay-sensitive services requiring constant bit rates, such as uncompressed video and other isochronous traffic.

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. Ones density is not maintained independently of the data stream. Sometimes called *binary coded alternate mark inversion*.

ATM—Asynchronous Transfer Mode. International standard for cell relay in which multiple service types (such as voice, video, or data) are conveyed in fixed-length (53-byte) cells. Fixed-length cells allow cell processing to occur in hardware, thereby reducing transit delays. 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 ones density 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.

CBR—constant bit rate. QoS class defined by the ATM Forum for ATM networks. CBR is used for connections that depend on precise clocking to ensure undistorted delivery.

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 to which all local loops in a given area connect and in which 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 signals.

Drop-and-Insert—(also called TDM cross connect) Allows DS0 channels from one T1 or E1 facility to be digitally cross-connected to DS0 channels on another T1 or E1. 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.

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

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.

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; emulates the extension interface of a PBX or the subscriber interface for a switch.

IETF—Internet Engineering Task Force

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

IVR—interactive voice response. Term used to describe systems that provide information in the form of recorded messages over telephone lines in response to user input in the form of spoken words or more commonly DTMF signaling. Examples include banks that allow you to check your balance from any telephone and automated stock quote systems.

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

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, using alternate mark inversion (AMI) 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

