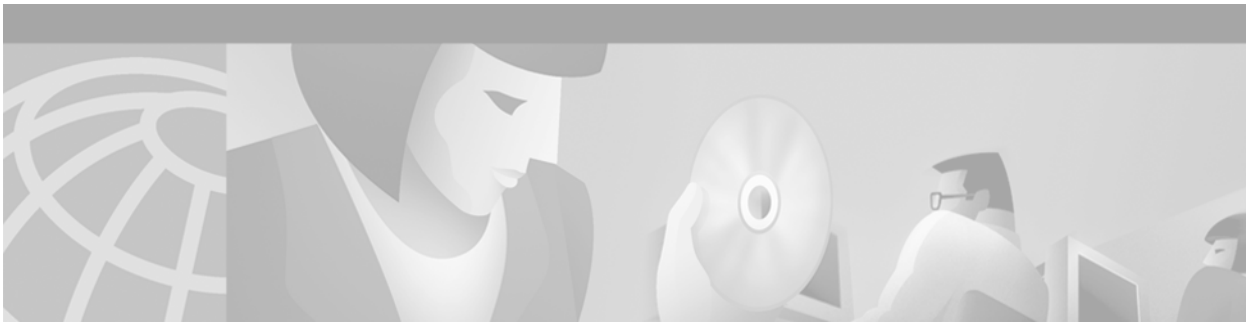


Configuring Voice over ATM

This chapter shows you how to configure Voice over ATM (VoATM). This chapter contains the following sections:

- Prerequisite Tasks
- VoATM Configuration Task List
- Configuring Voice Ports on the Cisco MC3810
- Configuring ATM PVCs for VoATM
- Configuring ATM SVCs for VoATM
- Preparing to Configure Voice Dial Peers
- Configuring Dial Peers
- Configuring Cisco-Trunk Permanent Calls on a Cisco MC3810
- VoATM Configuration Examples

VoATM enables a router to carry voice traffic (for example, telephone calls and faxes) over an ATM network. For a description of the commands used to configure VoATM, refer to the *Cisco IOS Multiservice Applications Command Reference publication.*

Note VoATM is supported only on the Cisco MC3810 in Cisco IOS Release 12.1. The Cisco MC3810 supports compressed VoATM on ATM port 0 only. When VoATM is enabled on the Cisco MC3810, the channel group, time-division multiplexing (TDM) group, and channel-associated signalling (CAS) functionality are not available on the Multiflex Trunk (MFT) because ATM uses all T1/E1 time slots.

Prerequisite Tasks

Before you can configure your router to use VoATM, you must first perform the following tasks:

- Establish a working ATM network. For more information about configuring ATM, refer to the *Cisco IOS Wide-Area Networking Configuration Guide.*

  Note ATM defaults to Integrated Local Management Interface (ILMI). If your carrier is using Local Management Interface (LMI), make sure to configure LMI support on the Cisco MC3810.

- Configure the clock source for the Cisco MC3810 interfaces. For more information, refer to the "Configuring Synchronous Clocking on the Cisco MC3810" appendix.
- Complete your company dial plan.
- Establish a working telephony network based on your company dial plan:
  - Integrate your dial plan and telephony network into your existing ATM network topology. Make routing and/or dialing transparent to the user—for example, avoid secondary dial tones from secondary switches, where possible.
  - Contact your PBX vendor for instructions about how to reconfigure the appropriate PBX interfaces.

After you have analyzed your dial plan and decided how to integrate it into your existing ATM network, you are ready to configure your network devices to support VoATM.

VoATM Configuration Task List

VoATM has a set of core tasks. To configure VoATM, perform the the tasks in the following sections:

- Configuring Voice Ports on the Cisco MC3810
- Configuring ATM PVCs for VoATM
- Configuring ATM SVCs for VoATM
- Preparing to Configure Voice Dial Peers
- Configuring Dial Peers
- Configuring Cisco-Trunk Permanent Calls on a Cisco MC3810

Configuring Voice Ports on the Cisco MC3810

The Cisco MC3810 hardware features different hardware configuration options for voice ports:

- Six analog voice interfaces—The Cisco MC3810 version with the analog voice module (AVM) supports up to six analog voice personality modules (APMs) with each voice module supporting a single signalling type. Each voice personality module maps to a single analog voice port. The FXO, FXS, and E&M voice modules can be installed in any combination.
- One digital voice module (DVM)—The Cisco MC3810 version with the DVM provides support for up to 24 voice channels, one for each voice port. Depending on whether the controller is T1 or E1, different DS0 voice channels are used. The DVM supports CAS for the following types: FXO, FXS, and E&M. For E&M signalling, the DVM also supports E1 MELCAS, a standard used primarily in the United Kingdom.
- With the optional BRI voice module (BVM) installed, the Cisco MC3810 multiservice access concentrator provides four ISDN BRI ports for connection to ISDN PBXs (private integrated services network exchanges, or PINXs). The BVM has four ISDN BRI ports for voice traffic. Each BRI port supports two voice channels (ISDN B channels) and one signalling channel (ISDN D channel).

Note If the BVM is installed, the speed (clock rate) of serial port 1 of the Cisco MC3810 is limited to a maximum of 192 kbps. This restriction assumes that the Multiflex Trunk (MFT) is installed in slot 3 on the Cisco MC3810. If the MFT is not installed, serial port 1 will not operate.

Voice ports provide support for three basic voice signalling formats:

- Foreign Exchange Office (FXO) interface. This interface allows a connection to be directed to the PSTN central office. The FXO interface also allows a connection to be directed to a standard PBX interface if the local telecommunications authority permits. This interface is of value for Off-Premise Extension (OPX) applications.
- Foreign Exchange Station (FXS) interface. This interface allows connection for basic telephone equipment and keysets, and supplies ring, voltage, and dial tone.
- "Ear and Mouth" (or "RecEive and TransMit") (E&M) interface allows connection for PBX trunk lines (tie lines). E&M is a signalling technique for 2-wire and 4-wire telephone and trunk interfaces. In addition, the Cisco MC3810 supports Mercury Exchange Limited channel associated signalling (MELCAS) for E&M voice ports.

In general, voice-port commands define the characteristics associated with a particular voice-port signalling type. Under most circumstances, the default voice-port command values are adequate to configure FXO and FXS ports to transport voice data using the Cisco MC3810. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice-port values configured, depending on the specifications of the devices in your telephony network.

Table 28 lists the valid slot and port numbers for the different voice interfaces.

Table 28 Voice Interface Slot and Port Number

Interface Type	Slot	Valid Port Numbers
Analog voice module (AVM)	1	1–6
Digital voice module (DVM)	1	Digital T1: 1–24 Digital E1: 1–15 and 17–31
Multiflex Trunk (MFT)	0	Digital T1: 1–24 Digital E1: 1–15 and 17–31

Note The voice-port number designations start with 1. Unlike serial port interfaces and interfaces on other Cisco products, there is no port 0 for voice ports.

Configuring FXO or FXS Voice Ports on the Cisco MC3810

Under most circumstances the default voice-port values are adequate for both FXO and FXS voice ports. To configure FXO and FXS voice ports on the Cisco MC3810, use the following commands beginning in global configuration mode:

	Command	Purpose

Step 1	For Cisco MC3810 series analog voice ports:
	`router(config)# voice-port slot/port`
	For Cisco MC3810 series digital voice ports:
	`router(config)# voice-port slot:ds0-group`
		Identifies the voice port you want to configure and enters voice-port configuration mode.

Step 2	`Router(config-voiceport)# dial-type {pulse | dtmf}`	(For FXO ports only) Selects the appropriate dial type for out-dialing. The default is dtmf.
Step 3	`Router(config-voiceport)# signal {loop-start | ground-start}`	Configures the signalling type for analog FXO and FXS voice ports. The default is loop-start.
Step 4	`Router(config-voiceport)# compand-type {u-law | a-law}`	(Digital voice ports only) Configures the companding standard used to convert between analog and digital signals in pulse code modulation (PCM) systems.
Step 5	`Router(config-voiceport)# cptone country`	Configures the appropriate call progress tone for the local region. The default for this command is **us**. For a list of supported countries, refer to the *Cisco IOS Multiservice Applications Command Reference* publication.
Step 6	`Router(config-voiceport)# ring number number`	(For FXO ports only) Configures the number of rings detected before a connection is closed on the FXO port.
Step 7	`Router(config-voiceport)# ring frequency number`	(For FXS ports only) Specifies the local ring frequency for the FXS voice port. The *number* value should be set to either 20 or 30.

	Command	Purpose
Step 8	`Router(config-voiceport)# connection {plar | tie-line | plar-opx} string`	Configures the voice-port connection mode type and the destination telephone number. The **plar** value is used for private line auto ringdown (PLAR) connections. The **tie-line** value on the Cisco MC3810 is used for a tie-line connection to a PBX. The **plar-opx** value on the Cisco MC3810 is used for PLAR OPX, to allow the local voice port to provide a local response before the remote voice port receives an answer. If you select the **plar-opx** value, you must also configure the **voice confirmation-tone** command. **Note** If you will be configuring Cisco switched trunk connections, do not configure this command now. See the "Configuring Cisco-Trunk Permanent Calls on a Cisco MC3810" section later in this chapter.
Step 9	`Router(config-voiceport)# description string`	(Optional) Enters a string description for the voice port. The string describes the voice port in displays. You can use the description command to note the voice port location or use.
Step 10	`Router(config-voiceport)# codec {g729r8 | g729ar8 | g726r32 | g711alaw | g711ulaw}`	((Optional for Cisco MC3810 only) Configures the voice-port compression mode. The g729ar8 value is the default and is recommended. The g729ar8 compression mode can support a maximum of 24 simultaneously active on-net voice calls and the g729r8 value can only support a maximum of 12. The g729 compression modes have a nominal data rate of 8 kbps. The Cisco MC3810 also supports configuring the **codec** command on the dial peer. Because the Cisco 2600 and 3600 series routers support the **codec** command on the dial peer only, it is recommended you configure this command on the dial peer.
Step 11	`Router(config-voiceport)# vad`	(Optional) Enables voice activity detection (VAD). **Note** The Cisco MC3810 also supports configuring the **vad** command on the dial peer. Because the Cisco 2600 and 3600 series routers support the **vad** command on the dial peer only, it is recommended you configure this command on the dial peer.

	Command	Purpose
Step 12	`Router(config-voiceport)# comfort-noise`	(Optional) Specifies that background noise will be generated if you have VAD activated.
Step 13	`Router(config-voiceport)# voice confirmation-tone`	(Optional) If the voice port is configured for **connection plar-opx** for OPX, disables the two-beep confirmation tone that a caller hears when picking up the handset.

Verifying the FXO and FXS Voice Port Configuration

You can verify your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- To verify that the data configured is correct, use the **show voice port** command.
- To verify the current status of all DSP voice channels, use the **show voice dsp** command.
- To verify the call status for all voice ports, use the **show voice call** summary command.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Cisco IOS IP and IP Routing Configuration Guide*.
- To make sure that the port is enabled, use the **show voice port** command. If the port is offline, use the **no shutdown** command.
- Check the dial-peer configuration.
- Check the Frame Relay, ATM, or HDLC configuration.
- Check whether the voice network module (VNM) has been correctly installed. For more information, refer to the *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide*.

Fine-Tuning FXO and FXS Voice Ports on the Cisco MC3810

Depending on your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation for FXO or FXS voice ports. Collectively, these commands are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

To fine-tune FXO or FXS voice ports on the Cisco MC3810, use the following commands beginning in global configuration mode:

	Command	Purpose

Step 1	For Cisco MC3810 series analog voice ports:
	`router(config)# voice-port slot/port`
	For Cisco MC3810 series digital voice ports:
	`router(config)# voice-port slot:ds0-group`
		Identifies the voice port you want to configure and enters voice-port configuration mode.

Step 2	`Router(config-voiceport)# input gain value`	Specifies (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from –6 to 14.
Step 3	`Router(config-voiceport)# output attenuation value`	Specifies (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14.
Step 4	`Router(config-voiceport)# echo-cancel enable`	Enables echo cancellation of voice that is sent out the interface and received back on the same interface.
Step 5	`Router(config-voiceport)# echo-cancel coverage value`	Adjusts the size (in milliseconds) of the echo cancel. Acceptable values are 16, 24, and 32.
Step 6	`Router(config-voiceport)# timeouts initial seconds`	Configures the initial timeout value. The initial timeout value specifies the number of seconds the system waits for the caller to input the first digit of the dialed digits. The default is 10 seconds.
Step 7	`Router(config-voiceport)# timeouts interdigit seconds`	Configures the interdigit timeout value. The timeouts interdigit value specifies the number of seconds the system waits (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The default is 10 seconds.
Step 8	`Router(config-voiceport)# timing digit milliseconds`	If the voice-port dial type is DTMF, configures the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100 milliseconds. The default is 100.
Step 9	`Router(config-voiceport)# timing inter-digit milliseconds`	If the voice-port dial type is DTMF, configures the DTMF interdigit signal duration. The range of the DTMF interdigit signal duration is from 50 to 500 milliseconds. The default is 100.
Step 10	`Router(config-voiceport)# timing pulse-digit milliseconds`	If the voice-port dial type is pulse, configures the pulse digit signal duration. The range of the pulse digit signal duration is from10 to 20 milliseconds. The default is 20.
Step 11	`Router(config-voiceport)# timing pulse-inter-digit milliseconds`	If the voice-port dial type is pulse, configures the pulse interdigit signal duration. The range of the pulse interdigit signal duration is from 100 to 1000 milliseconds. The default is 500.
Step 12	`Router(config-voiceport)# impedance {600r | 600c | 900r | 900c}`	(For FXO ports only) Configures the impedance. The default is 600 ohms real.

	Command	Purpose
Step 13	`Router(config-voiceport)# loss-plan {plan1 | plan2 | plan3 | plan4 | plan7 | plan8 | plan9}`	(For analog ports only) Specifies the loss plan for this voice port according to the signal level requirements for the DSP and the PBX. the default is plan1, which provides the following gain offset levels: • FXO—A-D gain = 0 dB, D-A gain = 0 dB • FXS—A-D gain = –3 dB, D-A gain = –3 dB
Step 14	`Router(config-voiceport)# ring cadence [on1 | off1] [on2 | off2] [on3 | off3] [on4 | off4] [on5 | off5] [on6 | off6]`	(For FXS ports only) Specifies the local ring cadence for the FXS voice port. Using this command, specify the on and off pulses for the ring. The ring cadence differs depending on the local region. The units are in 100-millisecond units.

Note After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

Configuring E&M Voice Ports on the Cisco MC3810

Unlike FXO and FXS voice ports, the default E&M voice-port parameters most likely will not be sufficient to enable voice data transmission over your network.

Note E&M voice-port values must match those of the PBX to which it is connected. Refer to the documentation that came with your specific PBX for the appropriate E&M voice-port configuration command values.

To configure E&M voice ports on the Cisco MC3810, use the following commands beginning in global configuration mode:

	Command	Purpose

Step 1	For Cisco MC3810 series analog voice ports:
	`router(config)# voice-port slot/port`
	For Cisco MC3810 series digital voice ports:
	`router(config)# voice-port slot:ds0-group`
		Identifies the voice port you want to configure and enters voice-port configuration mode.

Step 2	`Router(config-voiceport)# connection {plar | tie-line | plar-opx} string`	Configures the voice-port connection mode type and the destination telephone number. The **plar** value is used for private line auto ringdown (PLAR) connections. The **tie-line** value on the Cisco MC3810 is used for a tie-line connection to a PBX. The **plar-opx** value on the Cisco MC3810 is used for PLAR OPX, to allow the local voice port to provide a local response before the remote voice port receives an answer. If you select the **plar-opx** value, you must also configure the **voice confirmation-tone** command. **Note** If you will be configuring Cisco switched trunk connections, do not configure this command now. See the "Configuring Cisco-Trunk Permanent Calls on a Cisco MC3810" section later in this chapter.
Step 3	`Router(config-voiceport)# dial-type dtmf`	Selects the appropriate dial type for out-dialing. For E&M voice ports, the only available choice is DTMF.
Step 4	`Router(config-voiceport)# operation {2-wire | 4-wire}`	Selects the appropriate cabling scheme for this voice port.

	Command	Purpose
Step 5	`Router(config-voiceport)# type {1 | 2 | 3 | 5}`	Selects the appropriate E&M interface type. Type 1 indicates the following lead configuration: E—output, relay to ground M—input, referenced to ground Type 2 indicates the following lead configuration: E—output, relay to SG M—input, referenced to ground SB—feed for M, connected to –48V SG—return for E, galvanically isolated from ground Type 3 indicates the following lead configuration: E—output, relay to ground M—input, referenced to ground SB—connected to –48V SG—connected to ground Type 5 indicates the following lead configuration: E—output, relay to ground M—input, referenced to –48V
Step 6	`Router(config-voiceport)# signal {wink-start | immediate | delay-dial}`	Configures the signalling type for E&M voice ports. The default is wink start.

	Command	Purpose
Step 7	`Router(config-voiceport)# codec {g729r8 | g729ar8}` or	Configures the voice-port compression mode. The g729ar8 value is the default and is recommended. The g729ar8 compression mode can support a maximum of 24 simultaneously active on-net voice calls and the g729r8 value can only support a maximum of 12. Both compression modes have a nominal data rate of 8 kbps.
	`Router(config-voiceport) codec {g726r32 | g711alaw | g711ulaw}` or configure the **fax rate** dial-peer command. See the "Configuring VoATM Dial Peers" section later in this chapter.	(Fax only, optional) Configures the voice-port compression mode. If you are configuring a dial peer for faxes over ATM switched virtual circuits (SVCs), the default codec setting for the voice port does not function properly with the typical fax rate of 9600 bps. Therefore, you can either change the codec or change the fax rate. The codec must support a minimum of 22 kbps. To change the codec from the default g729ar8 codec (8 kbps), enter the voice-port **codec** command to specify one of the following compression modes: • **g726r32**—Specifies G.726 32K ADCPM compression. • **g711alaw**—Specifies G.711 64K PCM A-Law compression. • **g711ulaw**—Specifies G.711 64K PCM U-Law compression. **Note** The Cisco MC3810 also supports configuring the **codec** command on the dial peer as well. Because the Cisco 2600 and 3600 series routers support the **codec** command on the dial peer only, it is recommended you configure the **codec** command on the dial peer.
Step 8	`Router(config-voiceport)# compand-type {u-law | a-law}`	(Digital voice ports only) Configures the companding standard used to convert between analog and digital signals in PCM systems.

	Command	Purpose
Step 9	`Router(config-voiceport)# cptone country`	Configures the appropriate call progress tone for the local region. The default for this command is **us**. For a list of supported countries, refer to the *Cisco IOS Multiservice Applications Command Reference* publication.
Step 10	`Router(config-voiceport)# description string`	(Optional) Attaches descriptive text about this voice port connection.
Step 11	`Router(config-voiceport)# vad`	(Optional) Enables VAD on the voice port. **Note** The Cisco MC3810 also supports configuring the **vad** command on the dial peer. Because the Cisco 2600 and 3600 series routers support the **vad** command on the dial peer only, it is recommended you configure this command on the dial peer.
Step 12	`Router(config-voiceport)# voice confirmation-tone`	(Optional) If the voice port is configured for **connection plar-opx** for OPX, disables the two-beep confirmation tone that a caller hears when picking up the handset.

Verifying the E&M Voice Port Configuration

You can verify your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- To verify that the data configured is correct, use the **show voice port** command.
- To verify the current status of all DSP voice channels, use the **show voice dsp** command.
- To verify the call status for all voice ports, use the **show voice call** summary command.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Cisco IOS IP and IP Routing Configuration Guide*.
- To make sure that the port is enabled, use the **show voice port** command. If the port is offline, use the **no shutdown** command.
- Check the dial-peer configuration.
- Check the Frame Relay, ATM, or HDLC configuration.

- Check whether the voice network module (VNM) has been correctly installed. For more information, refer to the *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide*.

Fine-Tuning E&M Voice Ports on the Cisco MC3810

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain and output attenuation for E&M voice ports. Collectively, these commands are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

To fine-tune E&M voice ports on the Cisco MC3810, use the following commands beginning in global configuration mode:

	Command	Purpose

Step 1	For Cisco MC3810 series analog voice ports:
	`router(config)# voice-port slot/port`
	For Cisco MC3810 series digital voice ports:
	`router(config)# voice-port slot:ds0-group`
		Identifies the voice port you want to configure and enters voice-port configuration mode.

Step 2	`Router(config-voiceport)# input gain value`	Specifies (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from –6 to 14.
Step 3	`Router(config-voiceport)# output attenuation value`	Specifies (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14.
Step 4	`Router(config-voiceport)# echo-cancel enable`	Enables echo cancellation of voice that is sent out the interface and received back on the same interface.
Step 5	`Router(config-voiceport)# echo-cancel coverage value`	Adjusts the size (in milliseconds) of the echo cancel. Acceptable values are 16, 24, and 32.
Step 6	`Router(config-voiceport)# non-linear`	Enables nonlinear processing, which shuts off any signal if no near-end speech is detected. (Nonlinear processing is used with echo cancellation.)
Step 7	`Router(config-voiceport)# timeouts initial seconds`	Configures the initial timeout value. The initial timeout value specifies the number of seconds the system waits for the caller to input the first digit of the dialed digits. The default is 10 seconds.
Step 8	`Router(config-voiceport)# timeouts interdigit seconds`	Configures the interdigit timeout value. The timeouts interdigit value specifies the number of seconds the system waits (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The default is 10 seconds.

	Command	Purpose
Step 9	`Router(config-voiceport)# timeouts wait-release {value | infinity}`	Configures the timeout value for releasing voice ports. This command limits the duration that a voice port stays in the call failure state while the Cisco MC3810 sends a busy tone, reorder tone or out-of-service tone to the port.
Step 10	`Router(config-voiceport)# timing digit milliseconds`	If the voice-port dial type is DTMF, configures the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100 milliseconds. The default is 100.
Step 11	`Router(config-voiceport)# timing inter-digit milliseconds`	If the voice-port dial type is DTMF, configures the DTMF interdigit signal duration. The range of the DTMF interdigit signal duration is from 50 to 500 milliseconds. The default is 100.
Step 12	`Router(config-voiceport)# timing pulse-digit milliseconds`	If the voice-port dial type is pulse, configures the pulse digit signal duration. The range of the pulse digit signal duration is from 10 to 20 milliseconds. The default is 20.
Step 13	`Router(config-voiceport)# timing pulse-inter-digit milliseconds`	If the voice-port dial type is pulse, configures the pulse interdigit signal duration. The range of the pulse interdigit signal duration is from 100 to 1000 milliseconds. The default is 500.
Step 14	`Router(config-voiceport)# timing wink-duration milliseconds`	Configures the timing wink-duration value. This value sets the wink signal duration for a wink-start signal. This value applies only if the **signal** command is set to "wink-start." The range is from 100 to 400 milliseconds and the default is 200.
Step 15	`Router(config-voiceport)# timing wink-wait milliseconds`	Configures the timing wink-wait value. This value sets the wink wait duration for a wink-start signal. This value applies only if the **signal** command is set to "wink-start." The range is from 100 to 5000 milliseconds and the default is 200.
Step 16	`Router(config-voiceport)# timing clear-wait milliseconds`	Configures the timing clear-wait value. This value sets the amount of time between the inactive seizure signal and the call being cleared. The range is from 100 to 2000 milliseconds and the default is 400.
Step 17	`Router(config-voiceport)# timing delay-duration milliseconds`	Configures the timing delay-duration value. This value sets the delay signal duration for delay dial signalling. This value applies only if the **signal** command is set to "delay-dial." The range is from 100 to 5000 milliseconds and the default is 140.

	Command	Purpose
Step 18	`Router(config-voiceport)# timing delay-start` *milliseconds*	Configures the timing delay-start value. This value sets the delay interval between the generation of the delay-start signal from incoming detection seizure. This value applies only if the **signal** command is set to "delay-dial." The range is from 100 to 290 milliseconds and the default is 150.
Step 19	`Router(config-voiceport)# timing percentbreak` *percent*	Configures the timing percent-break value. This value sets the percentage of the break period for a dialing pulse. The default is 50 percent.

Note After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

Activating the Voice Port

After you have configured the voice port, you need to activate the voice port to bring it online. In fact it is a good idea to cycle the port—meaning to shut the port down and then bring it online again.

To activate a voice port, use the following command in voice-port configuration mode:

Command	Purpose
`Router(config-voiceport)# no shutdown`	Activates the voice port.

To cycle a voice port, use the following commands in voice-port configuration mode:

:

	Command	Purpose
Step 1	`Router(config-voiceport)# shutdown`	Deactivates the voice port.
Step 2	`Router(config-voiceport)# no shutdown`	Activates the voice port.

Note If you will not use a voice port, shut it down.

Configuring ATM PVCs for VoATM

This section describes the preliminary ATM configuration tasks necessary to support VoATM PVCs. The commands and procedures in this section are specific to the Cisco MC3810.

Note If any CAS groups, channel groups, or clear channels are configured on T1/E1 controller 0 on the Cisco MC3810, you must remove them before configuring VoATM. Because ATM requires all DS0s, if any DS0s on controller 0 are used by other applications, the ATM configuration cannot take place.

To configure the Cisco MC3810 to support VoATM PVCs on the T1/E1 trunk, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# controller {t1 | e1} 0`	Selects T1/E1 controller 0. ATM is supported only on controller 0 on the Cisco MC3810.
Step 2	`Router(config-controller)# mode atm`	Specifies that the controller will support ATM encapsulation, and to create virtual ATM interface 0, which you will use to create the ATM PVCs. When the controller is set to ATM mode, the following takes place: • The controller framing is automatically set to Extended Superframe (ESF) on T1 and to CRC4 on E1. The line code is automatically set to B8ZS on T1 and to HDB3 on E1. • Channel groups, CAS groups, common channel signalling (CCS) groups, or clear channels are not allowed on the trunk because ATM traffic occupies all the DS0s.
Step 3	`Router(config-controller)# no shutdown`	Activates the controller.
Step 4	`Router(config-controller)# exit`	Exits controller configuration mode.
Step 5	`Router(config)# interface atm0 {multipoint | point-to-point}`	Enters interface configuration mode to configure ATM interface 0. If the VoATM connection will be over a point-to-point network, specify the **point-to-point** option. The default option, **multipoint**, assumes you have a fully meshed network.
Step 6	`Router(config-if)# ip address ip-address mask`	Assigns the IP address and subnet mask to the interface.
Step 7	`Router(config-if)# pvc [name] vpi/vci`	Creates an ATM PVC for voice traffic and enters virtual circuit configuration mode.
Step 8	`Router(config-if-atm-pvc)# encapsulation aal5mux voice`	Sets the encapsulation of the PVC to support voice traffic. **Note** To configure a PVC to support data traffic, use aal5snap encapsulation.
Step 9	`Router(config-if-atm-pvc)# vbr-rt peak-rate average-rate [burst]`	Configures the peak rate, average rate, and the burst cell size to perform traffic shaping between voice and data PVCs. The **vbr-rt** command configures the VBR for real-time networks such as for voice networks. For more information on how to calculate the **vbr-rt** command values, see the "Calculating the VBR Real-Time Options for Traffic Shaping" section later in this chapter.

	Command	Purpose
Step 10	`Router(config-if-atm-pvc)# exit`	Exits ATM virtual circuit configuration mode. The only commands in ATM virtual circuit configuration mode used for ATM voice PVCs are **encapsulation aal5mux voice**, **vbr-rt**, and **ilmi**. Repeat Steps 7 through 10 for each ATM voice PVC you want to configure. When you have completed configuring all the ATM voice PVCs, continue with Steps 11 through 14.
Step 11	`Router(config-if)# pvc [name] vpi/vci`	Creates an ATM PVC for data traffic and enters PVC configuration mode.
Step 12	`Router(config-if-atm-pvc)# encapsulation aal5snap`	Sets the encapsulation of the PVC to support ATM data traffic. In ATM PVC configuration mode, configure either the **ubr**, **ubr+**, or the **vbr-nrt** traffic shaping commands for the data PVC as appropriate.
Step 13	`Router(config-if-atm-pvc)# exit`	Exits ATM virtual circuit configuration mode. Repeat Steps 11 and 12 for each data PVC configured.
Step 14	`Router(config-if)# exit`	Exits interface configuration mode.

The VoATM configuration must be performed on the routers on both sides of the voice connection.

Note When verifying your ATM PVC connectivity, note that you cannot issue the **ping** command over a voice PVC because the command applies to data only. If you have data and voice PVCs set to the same destination, you can issue the **ping** command over the data PVC.

Calculating the VBR Real-Time Options for Traffic Shaping

The **vbr-rt** command configures the VBR for real-time networks such as for voice networks. This command performs traffic shaping between voice and data PVCs. Traffic shaping is necessary so that the carrier does not discard the incoming calls from the Cisco MC3810. To configure voice and data traffic shaping, you must configure the peak, average, and burst options for voice traffic. Configure the burst value if the PVC will be carrying bursty traffic. The peak, average, and burst values are needed so the PVC can effectively handle the bandwidth for the number of voice calls. To calculate the *minimum* peak, average, and burst values for the number of voice calls, use the following calculations:

- Peak value: (2 x the maximum number of calls) x 16 Kb

  The peak value equals the Peak Information Rate (PIR).

- Average value: (1 x the maximum number of calls) x 16 Kb

  The average value equals the Average Information Rate (AIR). This value correlates to the carrier's sustainable cell rate (SCR).

- Burst value: (4 x the maximum number of calls)

  The burst value is the burst size in cells.

Note When you configure data PVCs that will be traffic shaped with voice PVCs, use the **aal5snap** encapsulation and calculate the overhead as 1.13 times the voice rate.

Configuring ATM SVCs for VoATM

In this section, the ATM interface is set up, including PVCs to carry signalling for SVCs. In addition, you specify a network service access point (NSAP) address for the ATM SVC. The *Cisco IOS Wide-Area Networking Configuration Guide* and *Cisco IOS Wide-Area Networking Command Reference* publications provide additional information about this and other aspects of ATM configuration.

To configure ATM SVCs to support VoATM, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# interface atm 0 [.subinterface-number {multipoint | point-to-point}]`	Configures the ATM interface. Optionally, you can create and configure a subinterface. This is useful when you wish to configure an extra parameter on the ATM interface. For example, you can specify one IP address on the main interface, as shown in Step 2; then you can configure a second IP address on a subinterface. The *subinterface-number* argument is a value in the range from 1 to 4294967293. Enter the **multipoint** keyword when your network is fully meshed and you want to communicate with multiple routers. The **point-to-point** keyword configures the subinterface for communication with one router, as in a hard-wired connection.
Step 2	`Router(config-if)# ip address ip-address mask`	Assigns the IP address and subnet mask to the interface.
Step 3	`Router(config-if)# atm voice aesa {default | esi-address}`	Sets the unique ATM end-station address (AESA) for an ATM interface that is using SVC mode for voice. The **default** keyword automatically creates an NSAP address for the interface, based on a prefix from the ATM switch (26 hexadecimal characters), the MAC address (12 hexadecimal characters) as the end station identifier (ESI), and a selector byte (2 hexadecimal characters). The *esi-address* option requires that you enter 12 hexadecimal characters as the ESI. The ATM switch provides the prefix and the voice selector byte provides the remaining characters. You can view the assigned address using the **show atm video-voice address** command.

	Command	Purpose
Step 4	`Router(config-if)# pvc [name] vpi/vci ilmi`	Creates an ATM permanent virtual circuit (PVC) for ILMI management purposes and enters PVC configuration mode. The optional *name* argument is a unique label that can be up to 16 characters long. The *name* argument identifies to the processor the virtual path identifier-virtual channel identifier (VPI-VCI) pair to use for a particular packet. The ATM network VPI of this PVC is an 8-bit field in the header of the ATM cell. The *vpi* value is unique only on a single link, not throughout the ATM network, because it has local significance only. The *vpi* value must match that of the switch. Valid values are from 0 to 255, but the value is usually 0 for ILMI communications. If not specified, the *vpi* value is set to 0. You cannot set both *vpi* and *vci* to 0; if one is 0, the other cannot be 0. For ILMI communications the VCI value is typically 16. The VCI is a 16-bit field in the header of the ATM cell. The VCI value is unique only on a single link—not throughout the ATM network—because it has local significance only. To set up communication with the ILMI, enter a value of **ilmi** for ATM adaptation layer (AAL) encapsulation; the associated *vpi* and *vci* values are typically 0 and 16, respectively. **Note** Typically, the low values 0 to 31 are reserved for specific traffic (for example, F4 OAM, SVC signalling, ILMI, and so on) and you should not use them for other PVCs.
Step 5	`Router(config-if-atm-pvc)# pvc [name] vpi/vci qsaal`	See the explanations in Step 4 for the *name*, *vpi*, and *vci* values. To enable the signalling for setup and tear-down of SVCs, specify the Q.SAAL (signalling ATM adaptation layer) encapsulation; the associated *vpi* and *vci* values are typically 0 and 5, respectively. You cannot create this PVC on a subinterface.
Step 6	`Router(config-if-atm-pvc)# exit`	Exits PVC interface configuration mode.

Verifying the ATM Interface Configuration

To verify the ATM interface configuration, perform the following steps:

Step 1 To see how SVCs and PVCs are set up, enter the privileged EXEC **show atm vc** command, as in the following example:

```
Router# show atm vc
VCD /                                        Peak  Avg/Min Burst
Interface   Name       VPI   VCI  Type   Encaps   SC   Kbps   Kbps   Cells  Sts
0           1            0     5  PVC    SAAL     UBR       0                UP
0           2            0    16  PVC    ILMI     UBR       0                UP
0           379          0    60  SVC    SNAP     UBR       0                UP
0           986          0    84  SVC    SNAP     UBR       0                UP
0           14           0   133  SVC    VOICE    VBR      64     16    10    UP
0           15           0   134  SVC    VOICE    VBR      64     16    10    UP
0           16           0   135  SVC    VOICE    VBR      64     16    10    UP
0           17           0   136  SVC    VOICE    VBR      64     16    10    UP
0           18           0   137  SVC    VOICE    VBR      64     16    10    UP
0           19           0   138  SVC    VOICE    VBR      64     16    10    UP
0           20           0   139  SVC    VOICE    VBR      64     16    10    UP
0           21           0   140  SVC    VOICE    VBR      64     16    10    UP
0           22           0   141  SVC    VOICE    VBR      64     16    10    UP
0           23           0   142  SVC    VOICE    VBR      64     16    10    UP
0           24           0   143  SVC    VOICE    VBR      64     16    10    UP
0           25           0   144  SVC    VOICE    VBR      64     16    10    UP
0           26           0   145  SVC    VOICE    VBR      64     16    10    UP
0           27           0   146  SVC    VOICE    VBR      64     16    10    UP
0           28           0   147  SVC    VOICE    VBR      64     16    10    UP
```

Step 2 Enter the **show atm svc** command with or without the VPI/VCI specified. The following example shows information for a specific SVC:

```
Router# show atm svc 0/134

ATM0: VCD: 5, VPI: 0, VCI: 134
VBR, PeakRate: 64000
AAL5, etype: 0x0, Flags 0x440, VCmode: 0xE000
OAM frequency: 0 second(s), OAM retry frequency: 1 second(s)
OAM up retry count: 3, OAM down retry count: 5
OAM Loopback status: OAM Disabled
OAM VC state: Not Managed
ILMI VC state: Not Managed
InARP DISABLED
InPkts: 4, OutPkts: 4, InBytes: 432, OutBytes: 432
InPRoc: 4, OutPRoc: 4, Broadcasts: 0
InFast: 0, OutFast: 0, InAS: 0, OutAS: 0
OAM cells received: 0
F5 InEndloop: 0, F5 InSegloop: 0, F5 InAIS: 0, F5 InRDI:0
F4 InEndloop: 0, F4 InSegloop: 0, F4 InAIS: 0, F4 InRDI:0
OAM cells sent: 0
F5 OutEndloop: 0, F5 OutSegloop: 0, F5 OutRDI: 0
OAM cell drops: 0
Status: UP
TTL: 3
interface = ATM0, call locally initiated, call reference = 5558610
vcnum = 5, vpi = 0, vci = 134, state = Active(U10), point-to-point call
Retry count: Current = 0
timer currently inactive, timer value = 00:00:00
Remote Atm Nsap address:47.00918100000000400B0A2501.0060837B4743.00, VCowner:Static Map
```

Step 3 To learn which PVCs are set up for ILMI management and Q.SAAL signalling, enter the **show atm pvc** command with the VPI/VCI specified, as in the following example:

```
Router# show atm pvc 0/5
ATM0: VCD: 2, VPI: 0, VCI: 5, Connection Name: SAAL
UBR, PeakRate: 56
AAL5-SAAL, etype:0x4, Flags: 0x26, VCmode: 0x0
OAM frequency: 0 second(s), OAM retry frequency: 1 second(s), OAM retry frequenc
y: 1 second(s)
OAM up retry count: 3, OAM down retry count: 5
OAM Loopback status: OAM Disabled
OAM VC state: Not Managed
ILMI VC state: Not Managed
InARP DISABLED
InPkts: 2044, OutPkts: 2064, InBytes: 20412, OutBytes: 20580
InPRoc: 2044, OutPRoc: 2064, Broadcasts: 0
InFast: 0, OutFast: 0, InAS: 0, OutAS: 0
OAM cells received: 0
F5 InEndloop: 0, F5 InSegloop: 0, F5 InAIS: 0, F5 InRDI: 0
F4 InEndloop: 0, F4 InSegloop: 0, F4 InAIS: 0, F4 InRDI: 0
OAM cells sent: 0
F5 OutEndloop: 0, F5 OutSegloop: 0, F5 OutRDI: 0
F4 OutEndloop: 0, F4 OutSegloop: 0, F4 OutRDI: 0
OAM cell drops: 0
Compress: Disabled
Status: INACTIVE, State: NOT_IN_SERVICE
!
Router# show atm pvc 0/16
ATM0: VCD: 1, VPI: 0, VCI: 16, Connection Name: ILMI
UBR, PeakRate: 56
AAL5-ILMI, etype:0x0, Flags: 0x27, VCmode: 0x0
OAM frequency: 0 second(s), OAM retry frequency: 1 second(s), OAM retry frequenc
y: 1 second(s)
OAM up retry count: 3, OAM down retry count: 5
OAM Loopback status: OAM Disabled
OAM VC state: Not Managed
ILMI VC state: Not Managed
InARP DISABLED
InPkts: 398, OutPkts: 421, InBytes: 30493, OutBytes: 27227
InPRoc: 398, OutPRoc: 421, Broadcasts: 0
InFast: 0, OutFast: 0, InAS: 0, OutAS: 0
OAM cells received: 0
F5 InEndloop: 0, F5 InSegloop: 0, F5 InAIS: 0, F5 InRDI: 0
F4 InEndloop: 0, F4 InSegloop: 0, F4 InAIS: 0, F4 InRDI: 0
OAM cells sent: 0
F5 OutEndloop: 0, F5 OutSegloop: 0, F5 OutRDI: 0
F4 OutEndloop: 0, F4 OutSegloop: 0, F4 OutRDI: 0
OAM cell drops: 0
Compress: Disabled
Status: INACTIVE, State: NOT_IN_SERVICE
```

Step 4 To view information about the ATM interface, enter the privileged EXEC **show atm interface** command and specify ATM 0, as in the following example:

```
Router# show interface atm 0
ATM0 is up, line protocol is up
  Hardware is PQUICC Atom1
  Internet address is 9.1.1.6/8
  MTU 1500 bytes, sub MTU 1500, BW 1536 Kbit, DLY 20000 usec,
     reliability 255/255, txload 22/255, rxload 11/255
  NSAP address: 47.009181000000000002F26D4901.000011116666.06
  Encapsulation ATM
  292553397 packets input, -386762809 bytes
  164906758 packets output, 1937663833 bytes
  0 OAM cells input, 0 OAM cells output, loopback not set
  Keepalive not supported
  Encapsulation(s):, PVC mode
  1024 maximum active VCs, 28 current VCCs
  VC idle disconnect time: 300 seconds
  Signalling vc = 1, vpi = 0, vci = 5
  UNI Version = 4.0, Link Side = user
  Last input 00:00:00, output 2d05h, output hang never
  Last clearing of "show interface" counters never
  Input queue: -1902/75/0 (size/max/drops); Total output drops: 205
  Queueing strategy: weighted fair
  Output queue: 0/1000/64/0 (size/max total/threshold/drops)
     Conversations  0/0/256 (active/max active/max total)
     Reserved Conversations 0/0 (allocated/max allocated)
  5 minute input rate 67000 bits/sec, 273 packets/sec
  5 minute output rate 136000 bits/sec, 548 packets/sec
     76766014 packets input, 936995443 bytes, 0 no buffer
     Received 0 broadcasts, 0 runts, 0 giants, 0 throttles
     0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
     367264676 packets output, 3261882795 bytes, 0 underruns
     0 output errors, 0 collisions, 2 interface resets
     0 output buffer failures, 0 output buffers swapped out
```

Step 5 Enter the **show atm video-voice address** privileged EXEC command to see information about the ATM interface address, which is particularly helpful because the address is assigned automatically with the **atm voice aesa** command. The following display also confirms that the ILMI status is confirmed—the ILMI PVC is set up to allow SVC management:

```
Router# show atm video-voice address
nsap address                                       type         ilmi status
47.009181000000000002F26D4901.00107B4832E1.FE   VOICE_AAL5   Confirmed
47.009181000000000002F26D4901.00107B4832E1.C8   VIDEO_AAL1   Confirmed
```

Troubleshooting VoATM SVCs

When problems occur with voice over ATM SVCs, perform the following steps to look first for simpler problems before progressing to more complex possible issues:

Step 1 Make sure that the ATM interface, serial ports, and controllers are set to **no shutdown**.

Step 2 On both Cisco MC3810 multiservice access concentrators, make sure that ILMI and Q.SAAL PVCs are set up in order to allow SVC communications. The **show atm pvc** command displays information about configured PVCs, including the ILMI and Q.SAAL PVCs.

```
Router# show atm pvc
 VCD /                                          Peak  Avg/Min Burst
Interface   Name       VPI   VCI  Type   Encaps   SC   Kbps   Kbps   Cells  Sts
0           1           0     5   PVC    SAAL     UBR        56                UP
0           2           0    16   PVC    ILMI     UBR        56                UP
```

Step 3 Ensure that NSAP addresses are set up and confirmed as operational under the ATM interfaces of the Cisco MC3810 multiservice access concentrators on both sides of the communication. Enter the **show atm video-voice address** or **show atm ilmi-status** privileged EXEC commands, which are shown in the following example. The **show atm ilmi-status** command provides more details about the ILMI PVC than does **show atm video-voice address** command:

```
router# show atm video-voice address
nsap address                                      type         ilmi status
47.0091810000000002F26D4901.00107B4832E1.FE   VOICE_AAL5   Confirmed

router# show atm ilmi-status

Interface : ATM0 Interface Type : Private UNI (User-side)
ILMI VCC : (0, 16) ILMI Keepalive : Enabled (5 Sec 4 Retries)
ILMI State:       UpAndNormal
Peer IP Addr:     10.1.1.11       Peer IF Name:     ATM1/0/0
Peer MaxVPIbits:  8               Peer MaxVCIbits:  14
Active Prefix(s) :
47.0091.8100.0000.0002.f26d.4901
End-System Registered Address(s) :
47.0091.8100.0000.0002.f26d.4901.0000.1111.5555.05(Confirmed)
47.0091.8100.0000.0002.f26d.4901.0010.7b48.32e1.fe(Confirmed)
47.0091.8100.0000.0002.f26d.4901.0010.7b48.32e1.c8(Confirmed)
```

Step 4 Check the voice ports for busyout status by entering the **show voice busyout** command. If the **busyout-monitor interface** or **busyout forced** command has been issued, one or more voice ports may be busied out, either due to a serial interface failure or because the voice port has been forced into a busyout state.

```
router# show voice busyout
If following network interfaces are down, voice port will be put into busyout state
Serial0
The following voice ports are in busyout state 1/10
```

Enter the **show interfaces serial** command to check the specified serial interface, or enter the **show voice port** command to check the voice port status. The **no** form of the **busyout forced** command restores the voice port.

Step 5 Check for clocking problems. Enter the **show controllers t1** command or the **show controllers e1** command to check for slip errors, as shown in the following excerpt from the command output:

```
 .
 .
 .
Data in current interval (819 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
 Data in Interval 1:
     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
 Data in Interval 2:
     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
 Data in Interval 3:
     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
  .
```

A few slip errors may not indicate a problem with clocking. However, if there are numerous errors, especially incrementing numbers of errors, you should check the following possibilities:

- The network clocks are not set to the same clock rate. Enter the **show network-clocks** command on the devices to ensure that these clock rates match.
- The Cisco MC3810 multiservice access concentrators may not be using the same clock source. For example, if there are two back-to-back Cisco MC3810 multiservice access concentrators and one is using an internal clock source, the other must use the line clock source in order to obtain clocking from the same device. Enter the **show network-clocks** and **show controllers t1** commands or the **show controllers e1** commands to learn the clock source settings. For additional guidance, see the "Configuring Synchronized Clocking" appendix.

Step 6 Check the functionality of the Service-Specific Connection-Oriented Protocol (SSCOP). Enter the privileged EXEC **show sscop** command. See the following excerpt from the command output:

```
router# show sscop
SSCOP details for interface ATM0
   Current State = Data Transfer Ready
```

Interpretation of the command output requires familiarity with SSCOP, so unless you understand the protocol, just use the command to ensure that the protocol is in a state of readiness, as shown in the example. If you need to make changes, see the *Cisco IOS Wide-Area Networking Configuration Guide* and the *Cisco IOS Wide-Area Networking Command Reference publications*.

Note If you plan to adjust SSCOP parameters, you may wish to complete the rest of the troubleshooting steps before taking this route.

Step 7 To verify that each dial peer has been configured properly to communicate with the other, enter the **show dial-peer voice** command on the local and remote concentrators, as shown in the following example:

```
Router1# show dial-peer voice
VoiceEncapPeer33
        tag = 1, destination-pattern = '5558810',preference = 0,
        Admin state is up, Operation state is up
        type = pots, prefix = '',       fwd-digits = 0,
        session-target = '', voice-port = 1/1
VoiceOverATMPeer333
        tag = 2, destination-pattern = '559...',preference = 0,
        Admin state is up, Operation state is up
        type = voatm, session-target = 'ATM0'
     nsap '47.0091810000000002F26D4901.567856785678.56',

Router2# show dial-peer voice
VoiceEncapPeer44
        tag = 20, destination-pattern = '5559810',preference = 0,
        Admin state is up, Operation state is up
        type = pots, prefix = '',       fwd-digits = 0,
        session-target = '', voice-port = 1/1
VoiceOverATMPeer444
        tag = 10, destination-pattern = '5558...',preference = 0,
        Admin state is up, Operation state is up
        type = voatm, session-target = 'ATM0'
    nsap '47.0091810000000002F26D4901.100110011001.01',
```

Step 8 To see information about current and recent voice calls, allowing analysis of possible problems, enter the **show call history voice record** privileged EXEC command:

```
router# show call history voice record
ConnectionId=[0x9CE20881 0x224855C1 0x0 0x1C9B84C7]
Media=TELE, TxDuration= 301962 ms
CallingNumber=6668808
SetupTime=47995411 x 10ms
ConnectTime=47995671 x 10ms
DisconectTime=48025867 x 10ms
DisconnectText=local onhook

ConnectionId=[0x9CE20881 0x224855C1 0x0 0x1C9B84C7]
Media=ATM, LowerIfName=ATM0, VPI=0, VCI=299
CalledNumber=5559808
SetupTime=47995483 x 10ms
ConnectTime=47995671 x 10ms
DisconectTime=48025867 x 10ms
DisconnectText=remote onhook

ConnectionId=[0x9CE20881 0x224855C2 0x0 0x1C9B84CB]
Media=TELE, TxDuration= 301950 ms
CallingNumber=5558803
SetupTime=47995412 x 10ms
ConnectTime=47995682 x 10ms
DisconectTime=48025877 x 10ms
DisconnectText=local onhook
```

Preparing to Configure Voice Dial Peers

After you have analyzed your dial plan and decided how to integrate it into your existing network, you are ready to configure your network devices to support VoATM. The actual configuration procedure depends on the topology of your voice network, but in general you need to perform the tasks in the following sections:

- Organizing Voice Network Information
- Creating a Peer Configuration Table

Timesaver You might want to configure the ATM dial peers in a back-to-back configuration before separating them across the ATM network. Using a back-to-back configuration, you can test your VoATM and dial-peer configuration to learn if you can successfully make a voice connection. Then, when you place both peers on the network, if you cannot make a voice connection you can isolate the cause as a network problem. For an example of a back-to-back voice over ATM configuration, see the "VoATM Configuration Examples" section later in this chapter.

Organizing Voice Network Information

After you have merged your telephony and WAN networks, there are tasks you can perform to simplify configuring VoATM. One task is to collect all of the information directly related to each dial peer by creating a peer configuration table.

Creating a Peer Configuration Table

Specific information relative to each dial peer needs to be identified before you can configure VoATM. One way to do this is to create a peer configuration table.

Figure 85 shows a diagram of a small voice network in which Router 1, with ATM virtual circuit 20, connects a small sales branch office to the main office through Router 2. Only two devices in the sales branch office need to be established as dial peers: a basic telephone and a fax machine. Router 2, with an ATM virtual circuit of 40, is the primary gateway to the main office; as such, it needs to be connected to the company PBX. Three devices need to be established as dial peers in the main office, all of which are basic telephones connected to the PBX.

Table 29 shows the peer configuration table for the example illustrated in Figure 85.

Figure 85 Sample VoATM Network

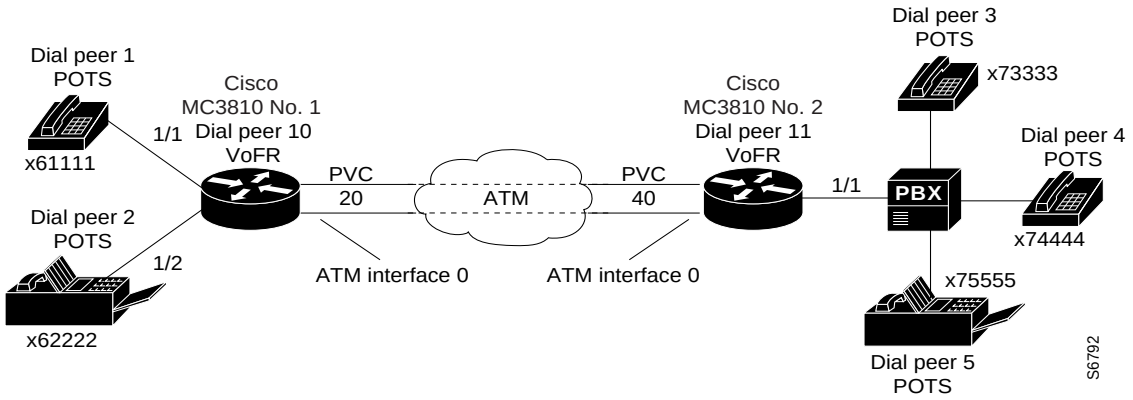


Table 29 Peer Configuration Table for Sample VoATM Network

Dial Peer	Extension	Prefix	Destination-Pattern	Type	Voice Port	Session Target
Router 1						
1	61111	—	+13101161111	POTS	1/1	—
2	62222	—	+13101162222	POTS	1/2	—
10	—	—	+1310117....	VoATM	—	S2 20
Router 2						
11	—	—	+1310116....	VoATM	—	S2 40
3	73333	7	+1310117....	POTS	1/1	—
4	74444	7	+1310117....	POTS	1/1	—
5	75555	7	+1310117....	POTS	1/1	—

The dial plan shown in Table 29 lists a simple dial-peer configuration table, with no special configuration for how you forward or play out excess digits. For more information on other options for designing your dial plan and configuring your dial peers to connect with PBXs, see the “Dial Planning and Dial-Peer Digit Manipulation Options” section in the “Configuring Voice over Frame Relay” chapter. The concepts described in that chapter also apply to VoATM.

Configuring Dial Peers

Dial peers describe the entities to and from which a call is established. Dial-peer configuration tasks define the address or set of addresses serviced by that dial peer and the call parameters required to establish a call to and from that dial peer.

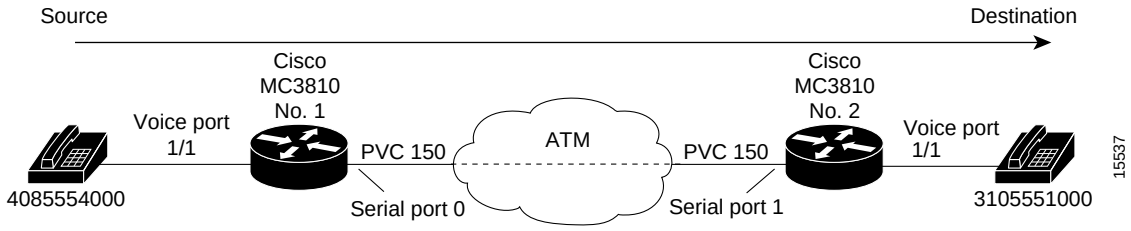
The following two different kinds of dial peers pertain to VoATM:

- POTS—Dial peer connected via a traditional telephony network. POTS dial peers point to a particular voice port on a voice-network device.
- VoATM—Dial peer connected via an ATM WAN backbone. VoATM dial peers point to specific voice-network devices.

POTS dial peers associate a telephone number with a particular voice port so that incoming calls for that telephone number can be received. VoATM dial peers point to specific voice-network devices (by associating destination telephone numbers with a specific ATM virtual circuit) so that outgoing calls can be placed. Both POTS and VoATM dial peers are required if you want to both send and receive calls using VoATM.

Establishing two-way communication using VoATM requires establishing a specific voice connection between two defined endpoints. As shown in Figure 86, for outgoing calls (from the perspective of the POTS dial peer 1), the POTS dial peer establishes the source (the originating telephone number and voice port) of the call. The VoATM dial peer establishes the destination by associating the destination phone number with a specific ATM virtual circuit.

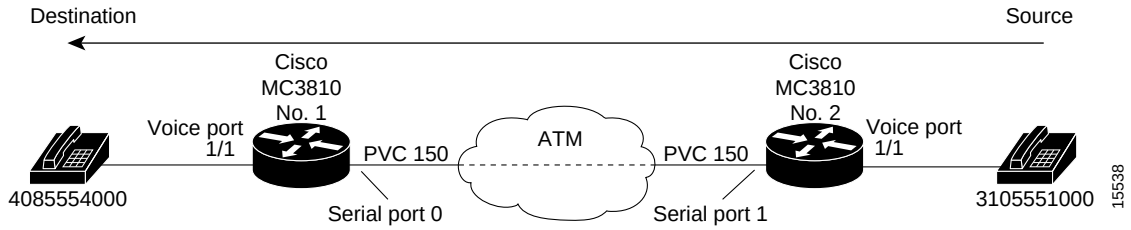
Figure 86 Calls from the Perspective of Cisco MC3810 No. 1



In the example, the destination pattern 14085554000 string maps to a U.S. phone number 555-4000, with the digit 1 plus the area code (408) preceding the number. When you configure the destination pattern, set the dial string to match the local dial conventions.

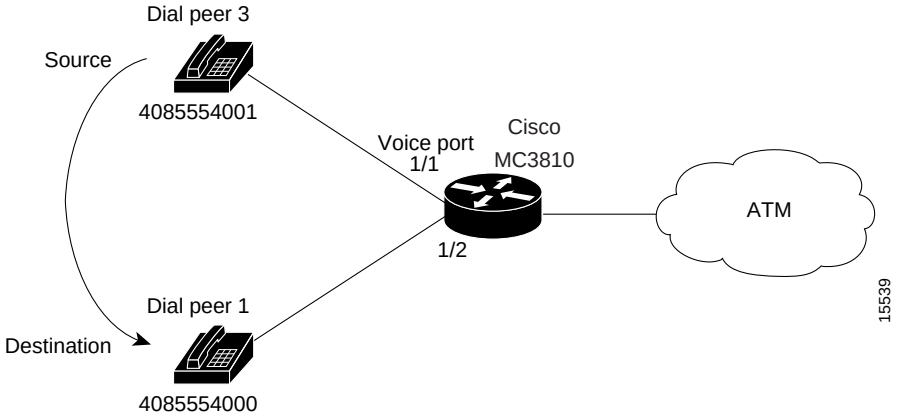
To complete the two-way communications loop, you need to configure VoATM dial peer 2 as shown in Figure 87.

Figure 87 Calls from the Perspective of Cisco MC3810 No. 2



The only exception to this dial peer concept is when both POTS dial peers are connected to the same concentrator, as shown in Figure 88. In this circumstance, because both dial peers share the same destination IP address, you need not configure a VoATM dial peer.

Figure 88 Communication Between Dial Peers Sharing the Same Concentrator



When configuring dial peers, you need to understand the relationship between the destination pattern and the session target. The destination pattern represents the pattern for the device at the voice connection endpoint, such as a telephone or a PBX. The session target represents the serial port on the peer Cisco MC3810 at the other end of the ATM connection. Figure 89 and Figure 90 show the relationship between the destination pattern and the session target, as seen from the perspective of both Cisco MC3810 concentrators in a VoATM configuration.

Figure 89 Relationship Between the Destination Pattern and Session Target from the Perspective of Cisco MC3810 No.1

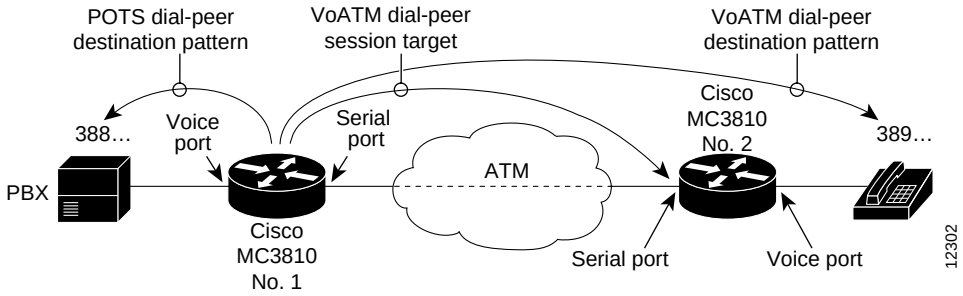
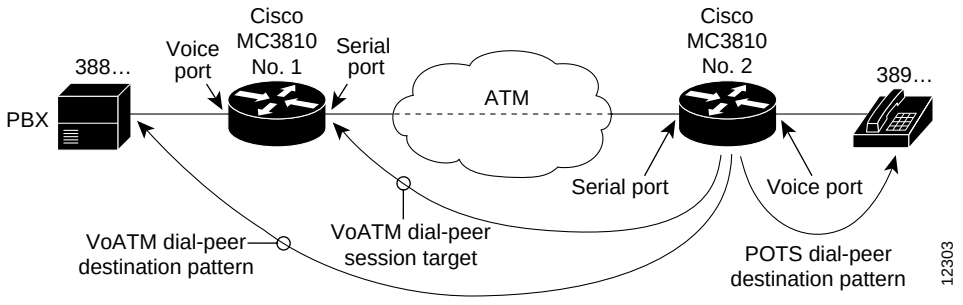


Figure 90 Relationship Between the Destination Pattern and Session Target from the Perspective of Cisco MC3810 No.2



Configuring POTS Dial Peers

To configure a POTS dial peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its telephone number, and associate it with a voice port through which calls will be established. Under most circumstances, the default values for the remaining dial-peer configuration commands will be sufficient to establish connections.

To configure POTS peers, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer voice` *number* `pots`	Defines a POTS dial peer and enters dial-peer configuration mode. All subsequent commands that you enter in dial-peer configuration mode before you exit will apply to this dial peer. The *number* tag identifies the dial peer and must be unique on the router. Do not duplicate a specific *number* tag.
Step 2	`Router(config-dial-peer)# destination-pattern` *string*	Configures the dial peer destination pattern. The *string* argument is a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0–9 and the letters A–D. The following special characters can be entered in the string: • The star character (*) and the pound sign (#) that appear on standard touch-tone dial pads can be used in any dial string. However, these characters cannot be used as leading characters in a string (for example, *650). • The period (.) can be entered any time, and is used as a wildcard character. • The comma (,) can be used only in prefixes, and is used to insert a 1-second pause or a delay. • The timer (T) character can be used to configure variable-length dial plans. For more information, see the “Dial Planning and Dial-Peer Digit Manipulation Options” section in the “Configuring Voice over Frame Relay” chapter.
Step 3	`Router(config-dial-peer)# port` *slot*`/`*port*	Associates this POTS dial peer with a specific logical dial interface. Enter the slot/port number of the voice port connected to the POTS dial peer.

	Command	Purpose
Step 4	`Router(config-dial-peer)# preference value`	(Optional) Configures a preference for the POTS dial peer. The value is a number from 0–10 where the lower the number, the higher the preference. If POTS and voice-network (VoATM) peers are mixed in the same hunt group, POTS dial peers will be searched first, even if a voice-network peer has a higher preference number. For more information about hunt groups and preferences, see the "Dial Planning and Dial-Peer Digit Manipulation Options" section in the "Configuring Voice over Frame Relay" chapter.
Step 5	`Router(config-dial-peer)# forward-digits {num-digit | all}`	(Optional) If you will use the digit-forwarding feature, configures the digit-forwarding method that will be used on the dial peer.
Step 6	`Router(config-dial-peer)# prefix string`	(Optional) If the forward digits feature was not configured in the last step, assigns the dialed digits prefix for the dial peer.

To configure additional POTS dial peers, exit dial-peer configuration mode by entering the **exit** command, and repeat the previous steps.

Configuring VoATM Dial Peers

To configure a VoATM dial peer, you need to uniquely identify the peer (by assigning it a unique tag number), and to define the outgoing serial port number and the virtual circuit number.

To configure VoATM dial peers, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer voice tag voatm`	Defines a VoATM dial peer for VoATM and enters dial-peer configuration mode. The *tag* value identifies the dial peer and must be unique on the router. Do not duplicate a specific *tag* number.
Step 2	`Router(config-dial-peer)# destination-pattern string`	Configures the dial peer destination pattern. The same restrictions for the string listed in the POTS dial-peer configuration also apply to the VoATM destination pattern.

	Command	Purpose
Step 3	`Router(config-dial-peer)# session target ATM0 {svc nsap address | pvc [vpi/vci | name]}`	Configures the ATM session target for the dial peer. Specify ATM 0 as the interface. When you use SVCs, the system reconciles dialed digits with the remote ATM interface voice NSAP address. **Note** If you are using PVCs for voice, you can specify a PVC defined on the ATM interface as a session target, by using a name or a VPI/VCI combination.
Step 4	`Router(config-dial-peer)# codec type [bytes payload_size]`	Specifies the voice coder rate of speech and payload size for the dial peer. The default dial peer codec is g729r8. Note that the Cisco MC3810 is limited to a maximum of 12 calls when using g729r8; to support up to 24 calls on the Cisco MC3810, use g729ar8. Specifying the payload size by entering the **bytes** value is optional. Each codec type defaults to a different payload size if you do not specify a value. To obtain a list of the default payload sizes, enter the **codec** command and the **bytes** option followed by a question mark (?). **Note** If configuring regular switched voice calls on the Cisco MC3810, you must configure the codec type on the voice port.
Step 5	`Router(config-dial-peer)# fax rate 4800`	(Fax only, optional) Configures the fax rate on the dial peer. If you are configuring a dial peer for faxes over ATM SVCs, the default codec setting for the voice port does not function properly unless the fax rate is slowed from the typical speed of 9600 bps. Therefore, you can either change the codec (see Step 4) or change the fax rate to 4800 bps or 2400 bps.
Step 6	`Router(config-dial-peer)# no vad`	(Optional) Disables VAD on the dial peer. This command is enabled by default.
Step 7	`Router(config-dial-peer)# preference value`	(Optional) Configures a preference for the VoATM dial peer. The value is a number from 0–10 where the lower the number, the higher the preference.
Step 8	`Router(config-dial-peer)# alt-dial string`	(Optional) Configures the alternate dial-out string when configuring on-net-to-off-net alternative dialing.

Note The Cisco MC3810 supports ATM traffic over T1/E1 controller 0 only.

To configure additional VoATM dial peers, exit dial-peer configuration mode by entering the **exit** command, and repeat the previous steps.

Depending on your configuration, you may need to consider how to configure your voice networks with excess digit playout, forward digits and default voice routes, or to use hunt groups with dial-peer preferences. For more information on these topics, see the "Dial Planning and Dial-Peer Digit Manipulation" section in the "Configuring Voice over Frame Relay" chapter. Although the descriptions for these topics use VoFR examples, these topics also apply to VoATM.

Configuring VoATM Dial Peers for Cisco-Trunk (Private Line) Calls

If you will be sending Cisco-trunk (private line) calls over ATM, you must configure the VoATM dial peers to specifically support Cisco-trunk (private line) calls. Cisco-trunk (private line) calls are permanent calls.

Note You can also create a voice class to configure trunk conditioning values for the idle and out-of-service (OOS) states, and then assign the voice class to the VoATM dial peer. For more information, see the "Configuring Voice-Related Support Features" chapter.

To configure a VoATM dial peer to support Cisco-trunk permanent (private line) calls, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer voice` *number* `voatm`	Defines a VoATM dial peer and enters dial-peer configuration mode. The *number* tag identifies the dial peer and must be unique on the router. Do not duplicate a specific *number* tag.
Step 2	`Router(config-dial-peer)# destination-pattern` *string*	Configures the dial peer destination pattern.
Step 3	`Router(config-dial-peer)# session target` *interface*	Configures the VoATM session target for the dial peer.
Step 4	`Router(config-dial-peer)# session protocol cisco-switched`	Configures the session protocol to support Cisco-trunk calls. The cisco-switched option is the default setting, and entering this command is not required.

	Command	Purpose
Step 5	`Router(config-dial-peer)# codec` *type* [**bytes** *bytes*]	Specifies the voice coder rate of speech and payload size for the dial peer. The default dial peer codec is g729r8. Note that the Cisco MC3810 is limited to a maximum of 12 calls when using g729r8; to support up to 24 calls on the Cisco MC3810, use g729ar8. Specifying the payload size by entering the **bytes** value is optional. Each codec type defaults to a different payload size if you do not specify a value. To obtain a list of the default payload sizes, enter the **codec** command and the **bytes** option followed by a question mark (?). **Note** On the Cisco MC3810, you can also assign codec values to the voice port. When you configure the codec type for regular switched voice calls, you must set the codec type on the Cisco MC3810 voice port. When you configure the codec for permanent calls (the **cisco-trunk** keyword), you must configure the codec type on the dial peer. You cannot specify the payload size on the voice port.
Step 6	`Router(config-dial-peer)# dtmf-relay`	(Optional) If the codec type is a low bit-rate codec such as g729 or g723, specifies support for DTMF relay to improve end-to-end transport of DTMF tones. DTMF tones do not always propagate reliably with low bit-rate codecs. DTMF relay is disabled by default.
Step 7	`Router(config-dial-peer)# signal-type {cas | cept | ext-signal | transparent}`	Defines the flavor of the ABCD signalling packets that are generated by the voice port and sent to the data network. The signal type must be configured to the same setting at both ends of the permanent voice call. Enter **cas** to support CAS. Enter **cept** to support the European CEPT standard (related to MEL CAS). Enter **ext-signal** to indicate that ABCD signalling packets should not be sent, for configurations where the line signalling information is carried externally to the voice port. Enter **transparent** (for digital T1/E1 interfaces on the Cisco MC3810 only) to read the ABCD signalling bits directly from the T1/E1 interface without interpretation, and to pass them transparently to the data network (this is also known as transparent FRF.11 signalling).

	Command	Purpose
Step 8	`Router(config-dial-peer)# no vad`	(Optional) Disables VAD on the dial peer. This command is enabled by default.
Step 9	`Router(config-dial-peer)# sequence-numbers`	(Optional) Enables the voice sequence number if required for your configuration. This command is disabled by default.
Step 10	`Router(config-dial-peer)# preference value`	(Optional) Configures a preference for the VoATM dial peer. The value is a number from 0–10 where the lower the number, the higher the preference in hunt groups.
Step 11	`Router(config-dial-peer)# fax rate {2400 | 4800 | 7200 | 9600 | 14400 | disable | voice}`	(Optional) Configures the transmission speed (in bps) at which a fax will be sent to the dial peer. The default is **voice**, which specifies the highest possible transmission speed allowed by the voice rate.

To configure another VoATM dial peer for Cisco-trunk (private line) calls, exit dial-peer configuration mode and repeat Steps 1 through 11.

To configure Cisco-trunk permanent calls on a Cisco MC3810, see the next section, “Configuring Cisco-Trunk Permanent Calls on a Cisco MC3810.”

Configuring Cisco-Trunk Permanent Calls on a Cisco MC3810

A Cisco-trunk (private line) call is basically a normal dynamic switched call of indefinite duration that uses a fixed destination telephone number and includes optional transparent end-to-end signalling. The telephone number of the destination endpoint is permanently configured into the router so that it always selects a fixed destination. After the call is established, either at boot-up or when configured, the call stays up until one of the voice ports or network ports is shut down, or until a network disruption occurs.

The **connection trunk** voice-port configuration command is used to establish a Cisco-trunk call; the dial peer is configured using the **session protocol cisco-switched** command, which invokes the Cisco proprietary session protocol.

The Cisco-trunk (private line) trunk call functionality provides the following benefits:

- Provides true permanent (private line) connections.
- Provides comprehensive busyout support for trunk connections (for more information, see the “Configuring Voice-Related Support Features” chapter).
- Provides transparent CAS protocol transport to allow the trunk to carry arbitrary ABCD signalling protocols.
- Provides conversion from North American signalling protocols to CEPT (Conférence Européenne des Postes et des Télécommunications) signalling protocols used for European voice networks.
- Provides remote analog to digital channel-bank operation for converting from DVM to AVM configurations on the Cisco MC3810.

To configure Cisco-trunk permanent calls on a Cisco MC3810 for VoATM, use the following commands beginning in interface configuration mode:

	Command	Purpose

Step 1	For Cisco MC3810 series analog voice ports:
	`router(config)# voice-port slot/port`
	For Cisco MC3810 series digital voice ports:
	`router(config)# voice-port slot:ds0-group`
		Enters voice-port configuration mode.

Step 2	`Router(config-voiceport)# connection trunk destination-string [answer-mode]`	Configures the trunk connection, specifying the telephone number in the *destination-string* argument. When you configure Cisco-trunk permanent calls, one side must be the call initiator (master) and the other side is normally the call answerer (slave). By default, the voice port operates in master mode. Enter the **answer-mode** keyword to specify that the voice port should operate in slave mode.
Step 3	`Router(config-voiceport)# shutdown`	Shuts down the voice port.
Step 4	`Router(config-voiceport)# no shutdown`	Reactivates the voice port to enable the trunk connection to take effect.

Note Every time you enter the **connection trunk** or **no connection trunk** command, you must toggle the voice port (by entering **shutdown**, then **no shutdown**) for the changes to take effect.

Disabling Dial-Peer Hunting on a Dial Peer

To disable dial-peer hunting on a dial peer, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer voice tag {pots | voatm}`	Enters dial-peer configuration mode for the specified dial peer.
Step 2	`Router(config-dial-peer)# huntstop`	Disables dial-peer hunting on the dial peer. Once you enter this command, no further hunting will be allowed if a call fails on the specified dial peer.

To reenable dial-peer hunting on a dial peer, enter the fsollowing commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer voice tag {pots | voatm}`	Enters dial-peer configuration mode for the specified dial peer.
Step 2	`Router(config-dial-peer)# no huntstop`	Reenables dial-peer hunting on the dial peer. Once you enter this command, no further hunting will be allowed if a call fails on the specified dial peer.

Verifying the VoATM Connection

Verify that the voice connection is working by doing the following:

- Pick up the handset on a telephone connected to the configuration and verify that you can get a dial tone.
- Make a call from the local telephone to a configured dial peer and verify that the call attempt is successful.

You can check the validity of your dial-peer and voice-port configuration by performing the following tasks:

- If you have relatively few dial peers configured, you can use the **show dial-peer voice** command to verify that the data configured is correct.
- To show the status of the voice ports, use the **show voice port** command.
- To show the call status for all voice ports, use the **show voice call** command.
- To show the current status of all DSP voice channels, use the **show voice dsp** command.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with the dial-peer configuration, you can try to resolve the problem by performing the following tasks:

- To verify that the data is configured correctly on both, use the **show dial-peer voice** command on the local and remote concentrators.
- To verify that ATM interface 0 is up, use the **show interface** command.
- Make sure the voice port, serial port and/or controller T1 0 is set to **no shutdown**.

VoATM Configuration Examples

Configuration examples for VoATM are shown in the following sections:

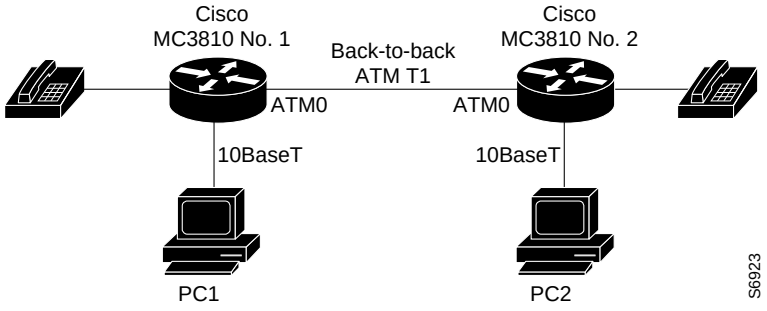
- Back-to-Back VoATM PVCs Example
- Voice and Data Traffic over ATM PVCs Example
- VoATM SVCs Example

Back-to-Back VoATM PVCs Example

Figure 91 shows a configuration example for two Cisco MC3810 concentrators configured back-to-back, with VoATM configured for both concentrators. This setup is useful to test your VoATM configuration locally to ensure voice connections can be made locally before you configure VoATM across a larger network. Following the figure are the commands required to configure the Cisco MC3810 concentrators in this example.

Note There may be special considerations for configuring back-to-back ATM compared to configuring for an ATM network.

Figure 91 Back-to-Back VoATM PVCs Configuration

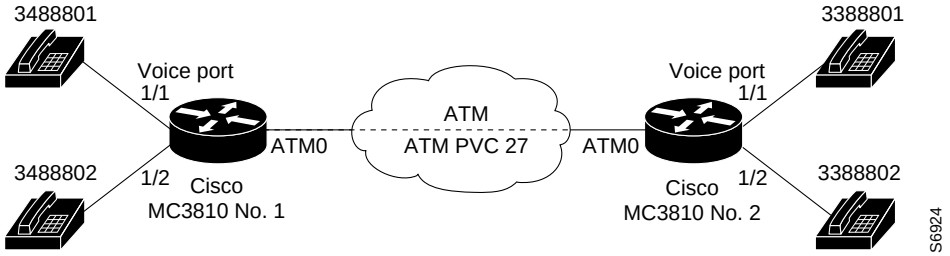


Cisco MC3810 No. 1	Cisco MC3810 No. 2
hostname location1 no ip domain-lookup ! interface Ethernet0 ip address 10.1.10.1 255.255.255.0 no ip mroute-cache no ip route-cache !	hostname location2 no ip domain-lookup ! interface Ethernet0 ip address 10.1.20.1 255.255.255.0 no ip mroute-cache no ip route-cache !
controller T1 0 clock source internal mode atm ! interface atm0 point-to-point ip address 10.1.1.1 255.255.255.0 no ip mroute-cache !	controller T1 0 clock source line mode atm ! interface atm0 point-to-point ip address 10.1.1.2 255.255.255.0 no ip mroute-cache !
pvc 1 1 100 encapsulation aal5mux voice vbr-rt 384 192 48 ! pvc 2 1 200 encapsulation aal5snap map-group atm1 ! router rip redistribute connected network 10.0.0.0 ! no ip classless !	pvc 1 1 100 encapsulation aal5mux voice vbr-rt 384 192 48 ! pvc 2 1 200 encapsulation aal5snap map-group atm1 ! router rip redistribute connected network 10.0.0.0 ! no ip classless !
map-list atm1 ip 10.1.1.2 atm pvc 2 broadcast ! dial-peer voice 1 pots destination-pattern 10 port 1/1 ! dial-peer voice 202 voatm destination-pattern 2. session target ATM0 1 ! end	map-list atm1 ip 10.1.1.1 atm pvc 2 broadcast ! dial-peer voice 1 pots destination-pattern 20 port 1/1 ! dial-peer voice 202 voatm destination-pattern 1. session target ATM0 1 ! end

Voice and Data Traffic over ATM PVCs Example

Figure 92 shows an example for both voice and data traffic over ATM between two Cisco MC3810 concentrators, including configuration for voice ports and dial peers. Following the figure are the commands required to configure the Cisco MC3810 concentrators in this example.

Figure 92 Voice and Data Traffic over ATM PVCs Configuration Example



Cisco MC3810 No. 1	Cisco MC3810 No. 2
interface Ethernet0 ip address 172.22.124.239 255.255.0.0 ! controller T1 0 mode ATM ! interface atm0 point-to-point ip address 223.223.224.229 255.255.255.0 no ip mroute-cache no ip route-cache map-group atm1 !	interface Ethernet0 ip address 172.22.124.247 255.255.0.0 ! controller T1 0 mode ATM ! interface atm0 point-to-point ip address 223.223.224.228 255.255.255.0 no ip mroute-cache no ip route-cache map-group atm1 !
pvc 26 26 200 encapsulation aal5snap ! pvc 27 27 270 encapsulation aal5mux voice vbr-rt 384 192 48 ! no ip classless !	pvc 26 26 200 encapsulation aal5snap ! pvc 27 27 270 encapsulation aal5mux voice vbr-rt 384 192 48 ! no ip classless !
map-list atm1 ip 223.223.224.228 atm pvc 26 broadcast ! voice-port 1/1 ! voice-port 1/2 ! voice-port 1/3 ! voice-port 1/4 !	map-list atm1 ip 223.223.224.229 atm pvc 26 broadcast ! login line vty 1 4 login ! voice-port 1/1 ! voice-port 1/2 ! voice-port 1/3 ! voice-port 1/4 !
dial-peer voice 1 pots destination-pattern 3488801 port 1/1 ! dial-peer voice 2 pots destination-pattern 3488802 port 1/2 ! end	dial-peer voice 1 pots destination-pattern 3388801 port 1/1 ! dial-peer voice 2 pots destination-pattern 3388802 port 1/2 ! dial-peer voice 1001 voatm destination-pattern 348.... session target ATM0 27 ! end

VoATM SVCs Example

Figure 93 shows the configurations of two Cisco MC3810 multiservice access concentrators that each have voice dial peers connecting over ATM SVCs.

Figure 93 Sample Configuration: Two Cisco MC3810s Using ATM SVCs for Voice

Cisco MC3810 No. 1	Cisco MC3810 No. 2
hostname MC3810A ! network-clock base-rate 64k ip subnet-zero ip wccp version 2 ip host keyer-ultra 223.255.254.254 ! appletalk routing ipx routing 1111.0045.0005	hostname MC3810B ! network-clock base-rate 64k ip subnet-zero ip wccp version 2 ip host keyer-ultra 223.255.254.254 ! appletalk routing ipx routing 1111.0045.0002
controller T1 0 framing esf linecode b8zs mode atm	controller T1 0 framing esf linecode b8zs mode atm
controller T1 1 framing esf clock source internal linecode b8zs mode cas voice-group 1 timeslots 1-24 type e&m-immediate-start	controller T1 1 framing esf clock source internal linecode b8zs mode cas voice-group 1 timeslots 1-24 type e&m-immediate-start
interface atm0 ip address 9.1.1.5 255.0.0.0 no ip directed-broadcast no ip route-cache atm pvc 1 0 5 qsaal atm pvc 2 0 16 ilmi atm ilmi-keepalive atm voice aesa default	interface atm0 ip address 9.1.1.6 255.0.0.0 no ip directed-broadcast no ip route-cache atm pvc 1 0 5 qsaal atm pvc 2 0 16 ilmi atm ilmi-keepalive atm voice aesa default

Cisco MC3810 No. 1	Cisco MC3810 No. 2
voice-port 1/1 ! voice-port 1/2 ! voice-port 1/3 timeouts call-disconnect 0 ! . . voice-port 1/24	voice-port 1/1 ! voice-port 1/2 ! voice-port 1/3 ! . . . voice-port 1/24
. . . dial-peer voice 2 pots destination-pattern 5558802 port 1/2 ! dial-peer voice 3 pots destination-pattern 5558803 port 1/3 ! dial-peer voice 4 pots destination-pattern 5558804 port 1/4 ! dial-peer voice 5 pots destination-pattern 5558805 port 1/5 . . . dial-peer voice 102 voatm destination-pattern 5559... session target ATM0 svc nsap 47.0091810000000002F26D4901.00107B09C645.FE	. . . dial-peer voice 2 pots destination-pattern 5559802 port 1/2 ! dial-peer voice 3 pots destination-pattern 5559803 port 1/3 ! dial-peer voice 4 pots destination-pattern 5559804 port 1/4 ! dial-peer voice 5 pots destination-pattern 5559805 port 1/5 . . . dial-peer voice 202 voatm destination-pattern 5558... session target ATM0 svc nsap 47.0091810000000002F26D4901.00107B4832E1.FE