



ATM Software Segmentation and Reassembly (SAR)

Feature History

Release	Modification
12.2(2)XB	Cisco 2600 Series T1/E1 ATM and Cisco 3660 T1 Inverse Multiplexing over ATM (IMA) ATM Adaption Layer 2 (AAL2) Support was introduced.
12.2(8)T	This feature was integrated into the Cisco IOS Release 12.2(8)T.

This document describes the ATM Software SAR feature and includes the following sections:

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Feature Overview

The ATM Software Segmentation and Reassembly (SAR) feature allows the Cisco 2600 series to carry voice and data traffic over ATM networks using AAL2 and AAL5 and allows the Cisco 3660 to support AAL2 voice traffic.

For the Cisco 2600 series, this feature works in conjunction with the T1/E1 multiflex voice/WAN interface card (VWIC), which is plugged into a WIC slot to provide one ATM WAN interface at a T1/E1 rate supporting up to 24/30 channels of voice.

T1/E1 ATM support is a time-to-market feature that helps service providers take advantage of the inherent quality of service (QoS) features of ATM multiservice applications. FR-ATM (FRF.5 and FRF.8) internetworking is supported on the Cisco 2600 series.

On the Cisco 3660 a T1 IMA network module is used as the IMA interface providing a maximum of one ATM IMA interface that supports up to 48/60 voice channels. Up to eight T1/E1s and multiple IMA groups are permitted, but only the first IMA group supports voice over AAL2 for up to 48/60 voice channels. NM-IMA already supports AAL5 on both the Cisco 2600 series and Cisco 3600 series (not just 3660).

The Cisco 2600 Series T1/E1 ATM portion of this feature provides a shared implementation of the ATM features currently available on the Cisco MC3810 with the Cisco 2600 series.

Figure 1 illustrates the ATM AAL2 nonswitched trunking feature connecting two private branch exchanges (PBXs) together without the call agent (CA).

Figure 1 ATM AAL2 Non-Switched Trunking Solution

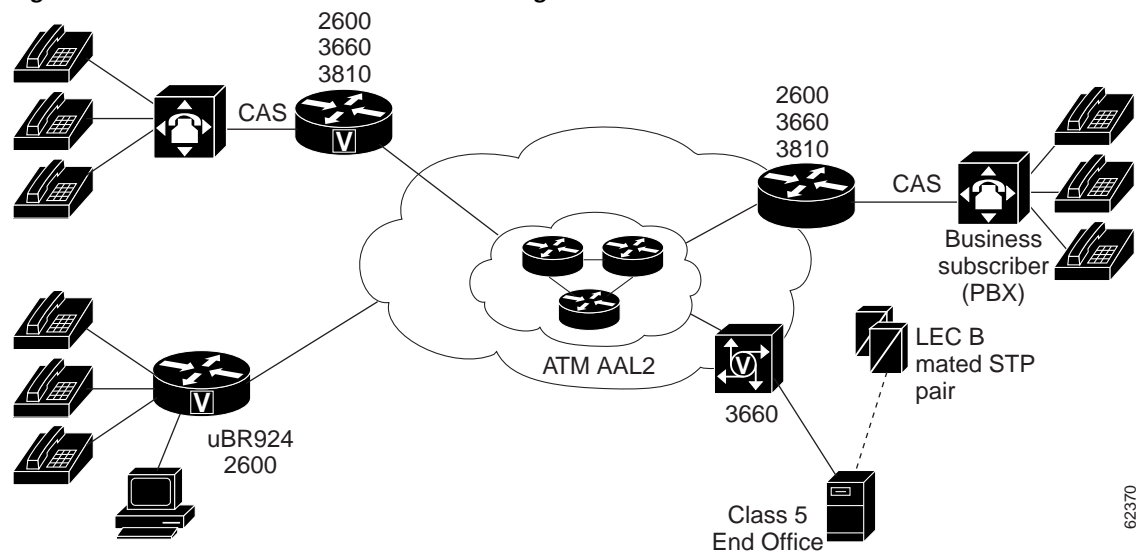
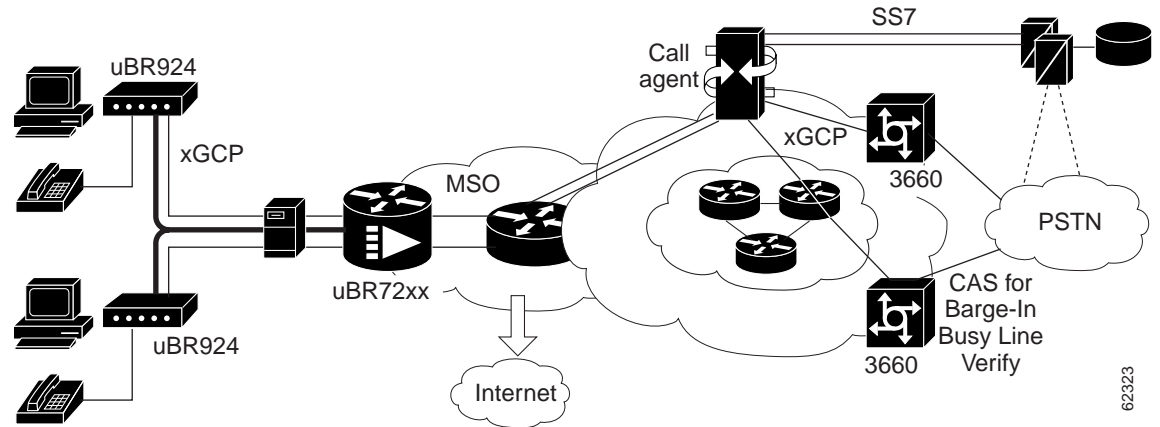


Figure 2 and Figure 3 illustrate CA solutions. In these solutions, a CA provides business voice services traditionally offered by a circuit-based PBX.

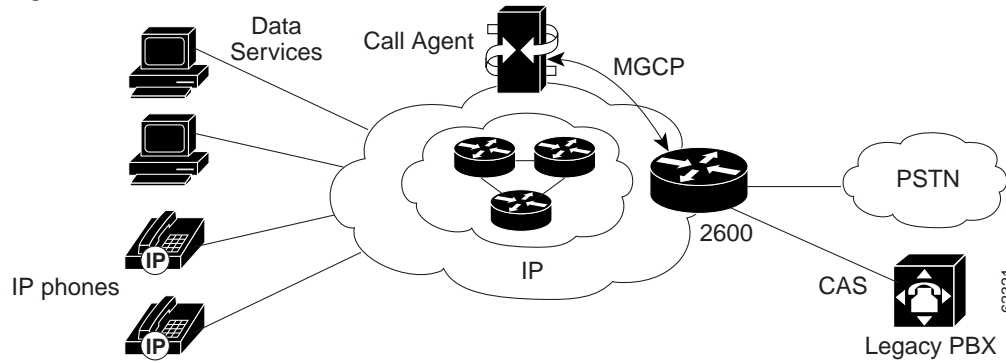
In Figure 2, the trunking gateway (the Cisco 3660 platform) requires support of incoming and outgoing multi-frequency signaling for the barge-in and busy-line verify features. The residential gateway (the Cisco uBR924 cable access router) must support the CLASS features and 911 capability.

Figure 2 Residential Cable Access Solution



In [Figure 3](#), the gateway (the Cisco 2600 platform) requires PBX connectivity to interface with the legacy PBX.

Figure 3 IP Centrex Solution



Benefits

AAL2 and AAL5 Functionality

Adds AAL2 and AAL5 functionality to the Cisco 2600 series and AAL2 to the Cisco 3660 on an IMA network module. AAL2 and AAL5 are the most bandwidth-efficient, standards-based trunking methods for transporting compressed voice, voice-band data, and frame-mode data over ATM infrastructures.

Economical ATM SAR Option

Provides robust, low-cost addition of ATM software SAR functionality to the Cisco 2600 series.

Lower Overhead and Better QoS

Enhances continued use of existing ATM infrastructure, providing traditionally high ATM QoS.

Restrictions

Cisco 2600 Series and Cisco 3660

- Analog voice modules are not supported for AAL2 feature. (IP over ATM AAL5 is supported.)

Cisco 2600 Series

- SS7 and bisync protocol cannot be used when this feature is active.
- Only one T1/E1 multiflex VWIC is supported, setting the number of allowable T1/E1 ATM interfaces to one.
- Only the Cisco 2650 and Cisco 2651 support end-to-end, Network Traceable Reference (NTR) clocking. For the NTR clock to work correctly, the T1/E1 multiflex VWIC must be placed in slot zero of the Cisco 2650 and Cisco 2651. In the case where a two-port T1/E1 multiflex VWIC is installed in slot zero, either of the two ports can be configured for support, but only one can be supported.
- The T1/E1 ATM feature requires that the T1/E1 multiflex VWIC be placed in slot zero.

Cisco 3660

- Only one IMA group can support AAL2 voice. If there are multiple IMA groups, then only the first IMA group supports AAL2 voice.
- Two T1/E1s are supported for ATM and 48/60 voice ports for PBX.
- Only the T1/E1 IMA network module supports AAL2 voice. This feature does not support OC3/T3/E3 network modules.
- The T1/E1 IMA network module does not support an NTR clock.

Cisco 2620XM

- When the traffic is sent with rate 100pps (256 bytes size), some cells are lost on the router where VWIC-1MFT-E1 is configured as ATM port. There is no workaround to this limitation. For a detailed description, see *Traffic Shaping on Cisco 3810 Routers* at the following URL:

http://www.cisco.com/warp/public/121/traff_shape3810.pdf

Related Features and Technologies

- Media Gateway Control Protocol (MGCP) channel associated signaling (CAS) PBX and AAL2 permanent virtual circuit (PVC) Software
- PRI/Q.931 Signaling Backhaul
- Voice over ATM with AAL2 Trunking

Related Documents

- MGCP CAS PBX and AAL2 PVC
 - <http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122t/122t2/ftmgcpk.htm>
- ATM forum documents for AAL2
 - ATM Trunking Using AAL2 for Narrowband Services (AF-VTOA-0113.000)

<http://www.atmforum.com/atmforum/specs/approved.html>

- ITU I-series documents located at <http://www.itu.int/home/index.html>
 - I.363.2, *B-ISDN ATM Adaptation Layer Specification: Type 2 AAL*
 - I.366.1, *Segmentation and Reassembly Service Specific Convergence Sublayer for the AAL Type 2*
 - I.366.2, *AAL Type 2 Service Specific Convergence Sublayer for Trunking*
- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2
- *Cisco IOS Interface Command Reference*, Release 12.2
- *Cisco IOS Interface Configuration Guide*, Release 12.2

Supported Platforms

- Cisco 2600 series
- Cisco 3660

Supported Standards, MIBs, and RFCs

Standards

No new or modified standards are supported by this feature.

MIBs

No new or modified MIBs are supported by this feature.

To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL:

<http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml>.

RFCs

- RFC 1577
- RFC 1483
- RFC 2221
- RFC 3661

Prerequisites

T1/E1 multiflex VWICs on Cisco 2600 series routers must be plugged into slot zero. The Cisco 3660 must be configured with a T1/E1 IMA Network Module. PBX voice requires a Digital T1/E1 Packet Voice Trunk Network Module Interface to be installed in the network module slot in the Cisco 2600 series or Cisco 3660.

You can configure the following four features on the Cisco 2600 series and Cisco 3660 routers:

- AAL2 Trunking with CAS
- AAL2 Trunking with common channel signaling (CCS)
- MGCP CAS
- MGCP Primary Rate Interface (PRI) Backhaul

Configuration Tasks for AAL2 Trunking with CAS

See the following sections for configuration tasks for AAL2 Trunking with CAS on Cisco 2600 series and Cisco 3660:

- [Configuring ATM on Cisco 2600 Series, page 7](#) (required for Cisco 2600 series routers only)
- [Configuring ATM on Cisco 3660, page 11](#) (required for Cisco 3660 routers only)
- [Configuring Voice Band Detection Payout Settings, page 15](#) (optional)
- [Configuring Subcell Multiplexing for AAL2 Voice, page 16](#) (optional)
- [Configuring End-to-End Clocking, page 16](#) (required)
- [Configuring Call Admission Control for AAL2 Voice, page 17](#) (required)
- [Configuring Dial Peers for AAL2 Voice, page 18](#) (required)
- [Configuring MGCP POTS Dial Peer, page 20](#) (required)
- [Configuring DS-0 group for CAS, page 20](#) (required)

Configuring ATM on Cisco 2600 Series

This section describes the ATM configuration tasks necessary to support Voice over ATM using AAL2 on the Cisco 2600 series.



Note

If any DS0 groups (CAS groups), channel groups, or clear channels are configured on T1/E1 controller 0, you must remove them before configuring VoATM. Because ATM uses all of the DS0 time slots on the controller, the ATM configuration cannot take place if any DS0s on controller 0 are used by other applications.

You must perform the VoATM configuration on the Cisco 2600 series or Cisco 3660 concentrators at both ends of the ATM link.

To configure a Cisco 2600 series or Cisco 3660 series concentrator to support VoATM on a T1/E1 trunk, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# controller {t1 e1} 0/0	Selects the T1 or E1 controller 0/0.
Step 2	Router(config-controller)# mode atm	Specifies that the controller will support ATM encapsulation and create ATM interface 0/0.
Step 3	Router(config-controller)# framing framing type	Specifies the T1 framing type as extended superframe or esf . When the controller is set to ATM mode, the Controller framing is automatically set to Extended SuperFrame (esf) on T1 and to cr4 on E1.
Step 4	Router(config-controller)# linecode linecode type	Specifies the T1 line code type as b8zs . The linecode is automatically set to b8zs on T1 and to HDB3 on E1.
Step 5	Router(config-controller)# no shutdown	Ensures that the controller is activated.

	Command	Purpose
Step 6	Router(config-controller)# exit	Exits controller configuration mode.
Step 7	Router(config)# interface atm slot/port [subinterface-number [multipoint point-to-point]]	Enters interface configuration mode to configure ATM interface 0/0 or an ATM subinterface. For subinterfaces, the default is multipoint .
Step 8	Router(config-if)# pvc [word] {vpi/vci vci}	Creates an ATM PVC for voice traffic and enter ATM virtual circuit configuration mode. <i>vpi</i> = 0 to 255 <i>vci</i> = 1 to 1023 <i>word</i> = optional PVC identifier (letters only); if you assign a PVC identifier, you can use it to specify this PVC when configuring network dial peers Note AAL2 encapsulation is not supported for interim local management interface (ilmi) and Q signaling ATM adaptation layer (qsaal) PVCs.
Step 9	Router(config-if-atm-vc)# encapsulation aal2	Sets the encapsulation of the PVC to support AAL2 voice traffic. This automatically creates channel identifiers (CIDs) 1 through 255.

Command	Purpose
<p>Step 10 Router(config-if-atm-vc)# vbr-rt <i>peak-rate</i> <i>average-rate</i> [<i>burst</i>]</p>	<p>Configures the PVC for variable-bit-rate real-time (VBRrt) (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • Peak rate—If it does not exceed your carrier’s allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • Average rate—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size—<i>max calls</i> x 85 G.726 with 40 or 80 byte sample size—<i>max calls</i> x 43 G.729 with 20 byte sample size—<i>max calls</i> x 22 G.729 with 30 byte sample size—<i>max calls</i> x 15 <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <ul style="list-style-type: none"> • Burst size—Set the burst size as large as possible, and never less than the minimum burst size. <p>Guidelines are as follows:</p> <p>The minimum burst size is 4 x the number of voice calls.</p> <p>The maximum burst size is the maximum allowed by the carrier.</p> <p>You can calculate the value using the calculator at the following URL:</p> <p>http://www.vnt/SPUniv/DSP/Codec_Calc1.asp</p>
<p>Step 11 Router(config-if-atm-vc)# oam-pvc [manage] [<i>frequency</i>]</p>	<p>(Optional) Configures transmission of end-to-end F5 OAM loopback cells on a PVC, optionally specifies the number of seconds between loopback cells, and optionally enable operation, administration, and maintenance (OAM) management of the connection.</p> <p>The range for <i>frequency</i> is 0 to 600. The default is 10.</p>

	Command	Purpose
Step 12	Router(config-if-atm-vc)# oam retry <i>up-count</i> <i>down-count</i> <i>retry-frequency</i>	<p>(Optional) Specifies OAM management parameters for verifying connectivity of a PVC connection. This command is supported only if OAM management is enabled.</p> <ul style="list-style-type: none"> • The value of <i>up-count</i> is the number of OAM loopback cell responses received to change the PVC connection to up. The range is 1 to 600; the default is 3. • The value of <i>down-count</i> is the number of OAM loopback cell responses not received to change the PVC connection to down. The range is 1 to 600; the default is 5. • The value of <i>retry-frequency</i> is the number of seconds between loopback cells sent to verify the down state of a PVC. The range is 1 to 1000; the default is 1. <p>Note Enter the oam retry command only once with all the arguments in the order shown. The first number always specifies <i>up-count</i>; the second <i>down-count</i>, and the third <i>retry-frequency</i>.</p>
Step 13	Router(config-if-atm-vc)# end	Exits configuration mode.
Step 14	Router# show atm vc	Verifies the ATM PVC configuration.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring ATM on Cisco 3660

Step 1	<pre>Router(config)# interface atm <slot>/ima<grp#> [subinterface-number [multipoint point-to-point]]</pre>	<p>Enters interface configuration mode to configure ATM interface 0/0 or an ATM subinterface.</p> <p>Note To configure an IMA group on each ATM interface, enter the IMA group and group number.</p> <p>The default for subinterfaces is multipoint.</p> <p><i>For all Scenarios:</i> Set up three subinterfaces for point-to-point.</p>
Step 2	<pre>Router(config-if)# pvc [name] vpi/vci</pre>	<p>Creates an ATM PVC for voice traffic and enters ATM virtual circuit configuration mode.</p> <p>Note AAL2 encap is not supported for ilmi and qsaal PVCs.</p>
Step 3	<pre>Router(config-if-atm-vc)# encapsulation aal2</pre>	<p>Sets the encapsulation of the PVC to support AAL2 voice traffic. This automatically creates channel identifiers (CIDs) 1 through 255.</p>

Step 4	<pre>Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]</pre>	<p>Configures the PVC for VBR-rt (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>Peak rate</i>—If it does not exceed your carrier's allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>Average rate</i>—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size—$max\ calls \times 85$ G.726 with 40 or 80 byte sample size—$max\ calls \times 43$ G.729 with 30 byte sample size—$max\ calls \times 15$ G.729 with 20 byte sample size—$max\ calls \times 22$ G.729 with 10 byte sample size—$max\ calls \times 43$ <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> • <i>Burst size</i>—Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: <ul style="list-style-type: none"> The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: <ul style="list-style-type: none"> http://www-vnt/SPUniv/DSP/Codec_Calc1.asp
Step 5	<pre>Router(config-if-atm-vc)# oam-pvc [manage] [frequency]</pre>	<p>(Optional) Configures transmission of end-to-end F5 OAM loopback cells on a PVC, optionally specifies the number of seconds between loopback cells, and optionally enables OAM management of the connection.</p> <p>The range for <i>frequency</i> is 0 to 600. The default is 10.</p>

Step 6	Router(config-if-atm-vc)# oam retry <i>up-count</i> <i>down-count</i> <i>retry-frequency</i>	<p>(Optional) Specifies OAM management parameters for verifying connectivity of a PVC connection. This command is supported only if OAM management is enabled.</p> <ul style="list-style-type: none"> • The value of <i>up-count</i> is the number of OAM loopback cell responses received to change the PVC connection to up. The range is 1 to 600; the default is 3. • The value of <i>down-count</i> is the number of OAM loopback cell responses not received to change the PVC connection to down. The range is 1 to 600; the default is 5. • The value of <i>retry-frequency</i> is the number of seconds between loopback cells sent to verify the down state of a PVC. The range is 1 to 1000; the default is 1. <p>Note Enter the oam retry command only once with all the arguments in the order shown. The first number always specifies <i>up-count</i>; the second <i>down-count</i>, and the third <i>retry-frequency</i>.</p>
Step 7	Router(config-if-atm-vc)# end	Exits configuration mode.
Step 8	Router# show atm vc	Verifies the ATM PVC configuration.

Step 9	Router(config-if-atm-vc)# vbr-rt <i>peak-rate</i> <i>average-rate</i> [<i>burst</i>]	<p>Configure the PVC for the variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>Peak rate</i>—If it does not exceed your carrier’s allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>Average rate</i>—Calculate according to the maximum number of calls (<i>max calls</i>) the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: for VoIP: G.711 with 40 or 80 byte sample size: <i>max calls</i> x 128K G.726 with 40 byte sample size: <i>max calls</i> x 85K G.729a with 10 byte sample size: <i>max calls</i> x 85K for VoAAL2: G.711 with 40 byte sample size: <i>max calls</i> x 85K G.726 with 40 byte sample size: <i>max calls</i> x 43K G.729a with 10 byte sample size: <i>max calls</i> x 43K If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less. • <i>Burst size</i>— Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: http://www-vnt/SPUniv/DSP/Codec_Calc1.asp
Step 10	Router(config-if-atm-vc)# vcci <i>pvc-identifier</i>	Assigns a unique identifier to the PVC.
Step 11	Router(config-if-atm-vc)# exit	Exits ATM virtual circuit configuration mode.
Step 12	Router(config-if)# exit	Exits interface configuration mode.
Step 13	Router(config)# dial-peer <i>voice number pots</i>	Enters dial peer configuration mode for the plain old telephone service (POTS) dial peer.
Step 14	Router(config-dial-peer)# application MGCPAPP	Initiates the MGCP protocol for the voice ports.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring Voice Band Detection Playout Settings

To configure voice band detection playout buffer delay on Cisco 2600 series and Cisco 3600 series routers, use the following commands beginning in the voice service configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# vbd-playout-delay maximum time	Specifies the maximum AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers in milliseconds. The time is set in milliseconds. The range is from 40-1700 milliseconds. The default is set to 200 milliseconds.
Step 4	Router(config-voice-service-session)# vbd-playout-delay minimum time	Specifies the minimum AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers. The time is set in milliseconds. The range is from 4-1700 milliseconds. The default is set to 4 milliseconds.
Step 5	Router(config-voice-service-session)# vbd-playout-delay mode {fixed passthrough}	Configures voice band detection playout delay adaptation mode on a Cisco router. When the vbd-delay-playout mode is set to fixed, jitter buffer is set at a constant delay in milliseconds. When the vbd-delay-playout mode is set to passthrough, jitter buffer is set to DRAIN_FILL for clock compensation. There is no default.
Step 6	Router(config-voice-service-session)# vbd-playout-delay nominal time	Specifies the nominal AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers. The time is set in milliseconds. The range is from 0-1500 milliseconds. The default is 100 milliseconds.
Step 7	Router(config-voice-service-session)# end	Exits voice-service-session configuration mode.

Configuring Subcell Multiplexing for AAL2 Voice

This section describes the configuration tasks necessary to enable AAL2 common part sublayer (CPS) subcell multiplexing when the Cisco 2600 series router or a Cisco 3660 interoperates with a voice interface service module (VISM) in an MGX switch.

To configure the Cisco 2600 series router or the Cisco 3660 to perform subcell multiplexing, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# subcell-mux <number>	Enables subcell multiplexing. The number is time in milliseconds. By default, subcell multiplexing is not enabled.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

Configuring End-to-End Clocking



Note

The following commands can be used to configure the Cisco 3660 only when there is a time-division multiplexing (TDM) switch module on board. For the Cisco 2600 series, these commands are automatically allowed.

	Command	Purpose
Step 1	Router(config)# network-clock-participate { nm wic } <i>slot</i>	Enables the Cisco 2600 series router or the Cisco 3660 to receive clock signals from the VWIC by entering the keyword wic and the slot number 0 on the router.
Step 2	For Cisco 2600 series: Router(config)# network-clock-participate { nm wic } <i>slot</i> For Cisco 3660: Router(config)# network-clock-participate { nm } <i>slot</i>	Enables the Cisco 2600 series router or the Cisco 3660 to receive clock signals from the network module by entering the keyword nm and the slot number 1 on the router.
Step 3	Router(config) network-clock-select <i>priority t1 slot/port</i>	Names a source to provide timing for the network clock and to specify the selection <i>priority</i> for this clock source. The priority selection is 1 or 2. Use the no form of this command to cancel the selection.
Step 4	Router(config) network-clock-select <i>priority t1 slot/port</i>	Assigns <i>priority 1</i> to ATM interface 0/0 and <i>priority 2</i> to controller 1/0.
Step 5	Router(config) voice-card <i>slot</i>	Enters voice-card configuration mode and sets codec complexity. For <i>slot</i> , use a value from 0 to 3 that describes the card location in the module.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring Call Admission Control for AAL2 Voice

This section describes the configuration tasks necessary to configure call admission control (CAC) for AAL2 voice. The commands and procedures in this section are common to the Cisco 2600 series and the Cisco 3660 routers.

You can configure a Cisco 2600 series router or a Cisco 3660 as either a CAC master or a CAC slave. By default, this is a CAC slave. You typically configure a CAC master at one end of an ATM trunk and a CAC slave at the opposite end. A Cisco 2600 series router or a Cisco 3660 configured as a master always performs CAC during fax/modem upspeed. A Cisco 2600 series router or a Cisco 3660 configured as a slave sends a request for CAC to the CAC master.

To configure a Cisco 2600 series router or a Cisco 3660 as a CAC master, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# cac master	Configures this Cisco 2600 series router or a Cisco 3660 as a CAC master.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

To return a Cisco 2600 series router or a Cisco 3660 to its default operation as a CAC slave, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# no cac master	Configures this Cisco 2600 series router or a Cisco 3660 as a CAC slave.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

Configuring Dial Peers for AAL2 Voice

For more information on dial peers and dial-peer configuration, see the “Configuring Voice over ATM” chapter in the *Cisco IOS Multiservice Applications Configuration Guide*, Release 12.1.

Configuring Network Dial Peers to Support AAL2

To configure a network dial peer for Voice over ATM (VoATM), specify a unique tag number, an ATM, a virtual circuit number, and a channel identifier (CID).

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 <i>tag</i> identifies the dial peer. Each <i>tag</i> on any one router must be unique.
Step 2	Router(config-dial-peer)# destination-pattern string	Configures the dial peer's destination pattern. The <i>string</i> is a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9 and the letters A through D. The following special characters can be entered in the string: <ul style="list-style-type: none"> • The star (*) and the pound sign (#) can be used in a dial string, but not as leading characters (for example *650 is not permitted). • The period (.) can be entered as a wildcard digit. Network dial peers typically use wildcards to represent a range of destination telephone numbers (for example, 1408555.... for all numbers in area code 408 with the 555 prefix). • The comma (,) can be used only in prefixes, and is used to insert a 1-second pause. • The timer (T) character can be used to configure variable-length dial plans.
Step 3	Router(config-dial-peer)# session protocol aal2-trunk	Configures the session protocol to support AAL2-trunk permanent (private line) trunk calls.

	Command	Purpose
Step 4	<pre>Router(config-dial-peer)# session target atm 0/0 pvc {word vpi/vci vci} cid (for Cisco 2600 series) For Cisco 3660: router(config-dial-peer)# session target atm <slot>/ima <group#></pre>	<p>Configures the ATM session target for the dial peer. Be sure to specify atm 0/0 as the interface for the PVC.</p> <p>Use <i>word</i> to identify the PVC if a word name was assigned when the PVC was created in the “Configuring ATM on Cisco 2600 Series” section on page 7.</p> <p>Use <i>word</i> to identify the PVC if a word name was assigned when the PVC was created in the “Configuring ATM on Cisco 3660” section on page 11.</p>
Step 5	<pre>Router(config-dial-peer)# codec aal2 profile {itut custom} profile-number codec</pre>	<p>Specifies a codec profile for the DSP.</p> <p>Profile options are itut 1, itut 2, itut 7, custom 100, and custom 110.</p> <p>The default is itut 1 with codec G.711 u-law.</p> <p>See the “Command Reference” section for the codec options available for each AAL2 profile.</p> <p>Note Use this command instead of the codec (dial-peer) command for AAL2 trunk applications.</p>
Step 6	<pre>Router(config-dial-peer)# dtmf-relay</pre>	<p>(Optional) If the codec type is a low bit-rate codec such as g729 or g723, specify support for dual tone multifrequency (DTMF) relay to improve end-to-end transport of DTMF tones. DTMF tones do not always propagate reliably with low bit-rate codecs.</p> <p>DTMF relay is disabled by default.</p>
Step 7	<pre>Router(config-dial-peer)# signal-type {ext-signal transparent}</pre>	<p>(Optional) Defines the type of ABCD signaling packets that are generated by the voice port and sent over the ATM network. The signal type must be configured to the same setting at both ends of the PVC.</p> <p>Enter ext-signal for common channel signaling (CCS). ABCD signaling packets are not sent.</p> <p>Enter transparent for nonswitched trunks using channel associated signaling (CAS). ABCD signaling bits are passed transparently to the ATM network.</p>
Step 8	<pre>Router(config-dial-peer)# no vad</pre>	<p>(Optional) Disables voice activity detection (VAD) on the dial peer. VAD is enabled by default.</p>
Step 9	<pre>Router(config-dial-peer)# exit</pre>	<p>Exits from the dial-peer configuration mode.</p>
Step 10	Repeat Step 1 through Step 9	Configures additional VoATM dial peers.

Configuring MGCP POTS Dial Peer

To configure MGCP POTS dial peer on the Cisco 2600 series and Cisco 3660, complete the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	<code>Router(config)# dial-peer voice number pots</code>	Enters the dial-peer configuration mode for the POTS dial peer.
Step 2	<code>Router(config-dial-peer)# application MGCPAPP</code>	Initiates the MGCP protocol for the voice ports.
Step 3	<code>Router(config-dial-peer)# port slot/port:ds0-group</code>	This command associates the dial peer with a specific logical interface. The value of <i>slot</i> is the router location where the voice port adapter is installed. Valid entries are from 0 to 3. The value of <i>port</i> indicates the voice interface card location. Valid entries are 0 or 1. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
Step 4	<code>Router(config-dial-peer)# exit</code>	Exits from the dial-peer configuration mode.

Configuring DS-0 group for CAS

To configure ds0 group for CAS on the Cisco 2600 series and Cisco 3660, complete the following steps:

	Command	Purpose
Step 1	<code>Router(config)# controller {t1 e1} slot/port</code>	<i>For the CAS PBX Scenarios only:</i> Selects the T1/E1 controller 1/0.
Step 2	<code>Router(config-controller)# mode cas</code>	<i>For the CAS PBX Scenarios only:</i> Specifies that the controller will support CAS.
Step 3	<code>Router(config-controller)# ds0-group channel-number timeslots range type signaling-type</code>	<i>For the CAS PBX Scenarios only:</i> Configures the T1 time slots for CAS calls. The scenarios use the following three DS0 definitions: <ul style="list-style-type: none"> ds0-group 1 time slots 1-8 type e&m-immediate-start ds0-group 2 time slots 9-16 type e&m-wink-start ds0-group 3 time slots 17-24 type fxs-ground-start
Step 4	<code>Router(config-controller)# exit</code>	<i>For the CAS PBX Scenarios only:</i> Exits controller configuration mode.

Configuration Tasks for AAL2 Trunking with CCS

See the following sections for configuration tasks for AAL2 Trunking with CCS on Cisco 2600 series and Cisco 3660:

- [Configuring ATM on the Cisco 2600 Series, page 21](#) (required only for Cisco 2600 series)
- [Configuring ATM on the Cisco 3660, page 24](#) (required only for Cisco 2600 series)
- [Configuring Voice Band Detection Payout Settings, page 28](#) (optional)
- [Configuring Subcell Multiplexing for AAL2 Voice, page 29](#) (optional)
- [Configuring End-to-End Clocking, page 29](#) (required)
- [Configuring Call Admission Control for AAL2 Voice, page 30](#) (required)
- [Configuring Dial Peers for AAL2 Voice, page 31](#) (required)
- [Configuring MGCP POTS Dial Peer, page 33](#) (required)
- [Configuring DS-0 Group and Channel Group for CCS, page 33](#) (required)
- [Configuring T-CCS Frame Forwarding, page 34](#) (required)

Configuring ATM on the Cisco 2600 Series

This section describes the ATM configuration tasks necessary to support Voice over ATM using AAL2 on Cisco 2600 series.



Note

If any DS0 groups (CAS groups), channel groups, or clear channels are configured on T1/E1 controller 0, you must remove them before configuring VoATM. Because ATM uses all of the DS0 time slots on the controller, the ATM configuration cannot take place if any DS0s on controller 0 are used by other applications.

You must perform the VoATM configuration on the Cisco 2600 series or Cisco 3660 concentrators at both ends of the ATM link.

To configure a Cisco 2600 series or Cisco 3660 series concentrator to support VoATM on a T1/E1 trunk, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# controller {t1 e1} 0/0	Selects the T1 or E1 controller 0/0.
Step 2	Router(config-controller)# mode atm	Specifies that the controller will support ATM encapsulation and create ATM interface 0/0.
Step 3	Router(config-controller)# framing <i>framing type</i>	Specifies the T1 framing type as extended super frame or esf . When the controller is set to ATM mode, the Controller framing is automatically set to Extended SuperFrame (esf) on T1 and to crc4 on E1.
Step 4	Router(config-controller)# linecode <i>linecode type</i>	Specifies the T1 linecode type as b8zs . The linecode is automatically set to b8zs on T1 and to hdb3 on E1.
Step 5	Router(config-controller)# no shutdown	Ensures that the controller is activated.
Step 6	Router(config-controller)# exit	Exits controller configuration mode.

	Command	Purpose
Step 7	Router(config)# interface <i>atmslot/port</i> [<i>subinterface-number</i> [multipoint point-to-point]]	Enters interface configuration mode to configure ATM interface 0/0 or an ATM subinterface. For subinterfaces, the default is multipoint .
Step 8	Router(config-if)# pvc [<i>word</i>] { <i>vpi/vci</i> <i>vci</i> }	Creates an ATM PVC for voice traffic and enters ATM virtual circuit configuration mode. <i>vpi</i> = 0 to 255 <i>vci</i> = 1 to 1023 <i>word</i> = optional PVC identifier (letters only); if you assign a PVC identifier, you can use it to specify this PVC when configuring network dial peers Note AAL2 encap is not supported for ilmi and qsaal PVCs.
Step 9	Router(config-if-atm-vc)# encapsulation aal2	Sets the encapsulation of the PVC to support AAL2 voice traffic. This automatically creates channel identifiers (CIDs) 1 through 255.

Command	Purpose
<p>Step 10 Router(config-if-atm-vc)# vbr-rt <i>peak-rate</i> <i>average-rate</i> [<i>burst</i>]</p>	<p>Configures the PVC for variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>peak-rate</i>—If it does not exceed your carrier’s allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>average-rate</i>—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size—<i>max calls</i> x 85 G.726 with 40 or 80 byte sample size—<i>max calls</i> x 43 G.729 with 30 byte sample size—<i>max calls</i> x 15 G.729 with 20 byte sample size—<i>max calls</i> x 22 G.729 with 30 byte sample size—<i>max calls</i> x 15 <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <ul style="list-style-type: none"> • <i>burst</i>—Set the burst size as large as possible, and never less than the minimum burst size. <p>Guidelines are as follows:</p> <p>The minimum burst size is 4 x the number of voice calls.</p> <p>The maximum burst size is the maximum allowed by the carrier.</p> <p>You can calculate the value using the calculator at the following URL:</p> <p>http://www.vnt/SPUniv/DSP/Codec_Calc1.asp</p>
<p>Step 11 Router(config-if-atm-vc)# oam-pvc [manage] [<i>frequency</i>]</p>	<p>(Optional) Configures transmission of end-to-end F5 OAM loopback cells on a PVC, optionally specifies the number of seconds between loopback cells, and optionally enables OAM management of the connection.</p> <p>The range for <i>frequency</i> is 0 to 600. The default is 10.</p>

	Command	Purpose
Step 12	Router(config-if-atm-vc)# oam retry <i>up-count</i> <i>down-count</i> <i>retry-frequency</i>	<p>(Optional) Specifies OAM management parameters for verifying connectivity of a PVC connection. This command is supported only if OAM management is enabled.</p> <ul style="list-style-type: none"> The value of <i>up-count</i> is the number of OAM loopback cell responses received to change the PVC connection to up. The range is 1 to 600; the default is 3. The value of <i>down-count</i> is the number of OAM loopback cell responses not received to change the PVC connection to down. The range is 1 to 600; the default is 5. The value of <i>retry-frequency</i> is the number of seconds between loopback cells sent to verify the down state of a PVC. The range is 1 to 1000; the default is 1. <p>Note Enter the oam retry command only once with all the arguments in the order shown. The first number always specifies <i>up-count</i>; the second <i>down-count</i>, and the third <i>retry-frequency</i>.</p>
Step 13	Router(config-if-atm-vc)# end	Exits configuration mode.
Step 14	Router# show atm vc	Verifies the ATM PVC configuration.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring ATM on the Cisco 3660

	Command	Purpose
Step 1	Router(config)# interface atm <slot>/ ima <grp#> [<i>subinterface-number</i> [multipoint point-to-point]]	<p>Enters interface configuration mode to configure ATM interface 0/0 or an ATM subinterface.</p> <p>Note To configure an IMA group on each ATM interface, enter the IMA group and group number.</p> <p>The default for subinterfaces is multipoint.</p> <p><i>For all Scenarios:</i> Set up three subinterfaces for point-to-point.</p>

Step 2	Router(config-if)# pvc [name] vpi/vci	<p>Creates an ATM PVC for voice traffic and enters ATM virtual circuit configuration mode.</p> <p>Note AAL2 encaps is not supported for ilmi and qsaal PVCs.</p>
Step 3	Router(config-if-atm-vc)# encapsulation aal2	<p>Sets the encapsulation of the PVC to support AAL2 voice traffic. This automatically creates channel identifiers (CIDs) 1 through 255.</p>
Step 4	Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]	<p>Configures the PVC for variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>peak rate</i>—If it does not exceed your carrier’s allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>average rate</i>—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size—<i>max calls</i> x 85 G.726 with 40 or 80 byte sample size—<i>max calls</i> x 43 G.729 with 30 byte sample size—<i>max calls</i> x 15 G.729 with 20 byte sample size—<i>max calls</i> x 22 G.729 with 10 byte sample size—<i>max calls</i> x 43 <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <ul style="list-style-type: none"> • <i>burst size</i>—Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: <ul style="list-style-type: none"> The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. <p>You can calculate the value using the calculator at the following URL:</p> <p>http://www-vnt/SPUniv/DSP/Codec_Calc1.asp</p>
Step 5	Router(config-if-atm-vc)# oam-pvc [manage] [frequency]	<p>(Optional) Configures transmission of end-to-end F5 OAM loopback cells on a PVC, optionally specify the number of seconds between loopback cells, and optionally enable OAM management of the connection.</p> <p>The range for <i>frequency</i> is 0 to 600. The default is 10.</p>

Step 6	Router(config-if-atm-vc)# oam retry <i>up-count</i> <i>down-count</i> <i>retry-frequency</i>	<p>(Optional) Specifies OAM management parameters for verifying connectivity of a PVC connection. This command is supported only if OAM management is enabled.</p> <ul style="list-style-type: none"> • The value of <i>up-count</i> is the number of OAM loopback cell responses received to change the PVC connection to up. The range is 1 to 600; the default is 3. • The value of <i>down-count</i> is the number of OAM loopback cell responses not received to change the PVC connection to down. The range is 1 to 600; the default is 5. • The value of <i>retry-frequency</i> is the number of seconds between loopback cells sent to verify the down state of a PVC. The range is 1 to 1000; the default is 1. <p>Note Enter the oam retry command only once with all the arguments in the order shown. The first number always specifies <i>up-count</i>; the second <i>down-count</i>, and the third <i>retry-frequency</i>.</p>
Step 7	Router(config-if-atm-vc)# end	Exits configuration mode.
Step 8	Router# show atm vc	Verifies the ATM PVC configuration.

Step 9	<pre>Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]</pre>	<p>Configures the PVC for the variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> <i>peak rate</i>—If it does not exceed your carrier's allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). <i>average rate</i>—Calculate according to the maximum number of calls (<i>max calls</i>) the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: for VoIP: G.711 with 40 or 80 byte sample size: $max\ calls \times 128K$ G.726 with 40 byte sample size: $max\ calls \times 85K$ G.729a with 10 byte sample size: $max\ calls \times 85K$ for VoAAL2: G.711 with 40 byte sample size: $max\ calls \times 85K$ G.726 with 40 byte sample size: $max\ calls \times 43K$ G.729a with 10 byte sample size: $max\ calls \times 43K$ <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <i>burst</i>—Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: http://www-vnt/SPUniv/DSP/Codec_Calc1.asp
Step 10	<pre>Router(config-if-atm-vc)# vcci pvc-identifier</pre>	Assigns a unique identifier to the PVC.
Step 11	<pre>Router(config-if-atm-vc)# exit</pre>	Exits ATM virtual circuit configuration mode.
Step 12	<pre>Router(config-if)# exit</pre>	Exits interface configuration mode.
Step 13	<pre>Router(config)# dial-peer voice number pots</pre>	Enter dial peer configuration mode for the POTS dial peer.
Step 14	<pre>Router(config-dial-peer)# application MGCPAPP</pre>	Initiates the MGCP protocol for the voice ports.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring Voice Band Detection Playout Settings

To configure voice band detection playout buffer delay on Cisco 2600 series and Cisco 3600 series routers, use the following commands beginning in the voice service configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# vbd-playout-delay maximum time	Specifies the maximum AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers in milliseconds. The time is set in milliseconds. The range is from 40-1700 milliseconds. The default is set to 200 milliseconds.
Step 4	Router(config-voice-service-session)# vbd-playout-delay minimum time	Specifies the minimum AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers. The time is set in milliseconds. The range is from 4-1700 milliseconds. The default is set to 4 milliseconds.
Step 5	Router(config-voice-service-session)# vbd-playout-delay mode {fixed passthrough}	Configures voice band detection playout delay adaptation mode on a Cisco router. When the vbd-delay-playout mode is set to fixed, jitter buffer is set at a constant delay in milliseconds. When the vbd-delay-playout mode is set to passthrough, jitter buffer is set to DRAIN_FILL for clock compensation. There is no default.
Step 6	Router(config-voice-service-session)# vbd-playout-delay nominal time	Specifies the nominal AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers. The time is set in milliseconds. The range is from 0-1500 milliseconds. The default is 100 milliseconds.
Step 7	Router(config-voice-service-session)# end	Exits voice-service-session configuration mode.

Configuring Subcell Multiplexing for AAL2 Voice

This section describes the configuration tasks necessary to enable AAL2 common part sublayer (CPS) subcell multiplexing when the Cisco 2600 series router or a Cisco 3660 interoperates with a voice interface service module (VISM) in an MGX switch.

To configure the Cisco 2600 series router or the Cisco 3660 to perform subcell multiplexing, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# subcell-mux <number>	Enables subcell multiplexing. The number is time in milliseconds. By default, subcell multiplexing is not enabled.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

Configuring End-to-End Clocking



Note

The following commands can be used to configure the Cisco3660 only when there is a TDM switch module on board. For Cisco 2600 series these commands are automatically allowed.

	Command	Purpose
Step 1	Router(config)# network-clock-participate {nm wic} slot	Enables the Cisco 2600 series router or the Cisco 3660 to receive clock signals from the VWIC by entering the keyword wic and the slot number 0 on the router.
Step 2	For Cisco 2600 series: Router(config)# network-clock-participate {nm wic} slot For Cisco 3660: Router(config)# network-clock-participate {nm} slot	Enables the Cisco 2600 series router or the Cisco 3660 to receive clock signals from the network module by entering the keyword nm and the slot number 1 on the router.
Step 3	Router(config) network-clock-select priority t1 slot/port	Names a source to provide timing for the network clock and to specify the selection <i>priority</i> for this clock source. The priority selection is 1 or 2. Use the no form of this command to cancel the selection.
Step 4	Router(config) network-clock-select priority t1 slot/port	Assigns <i>priority 1</i> to ATM interface 0/0 and <i>priority 2</i> to controller 1/0
Step 5	Router(config) voice-card slot	Enters voice-card configuration mode and set codec complexity. For <i>slot</i> use a value from 0 to 3 that describes the card location in the module.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring Call Admission Control for AAL2 Voice

This section describes the configuration tasks necessary to configure call admission control (CAC) for AAL2 voice. The commands and procedures in this section are common to the Cisco 2600 series and the Cisco 3660.

You can configure a Cisco 2600 series router or a Cisco 3660 as either a CAC master or a CAC slave. By default, this is a CAC slave. You typically configure a CAC master at one end of an ATM trunk and a CAC slave at the opposite end. A Cisco 2600 series router or a Cisco 3660 configured as a master always performs CAC during fax/modem upspeed. A Cisco 2600 series router or a Cisco 3660 configured as a slave sends a request for CAC to the CAC master.

To configure a Cisco 2600 series router or a Cisco 3660 as a CAC master, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# cac master	Configures this Cisco 2600 series router or a Cisco 3660 as a CAC master.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

To return a Cisco 2600 series router or a Cisco 3660 to its default operation as a CAC slave, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# no cac master	Configures this Cisco 2600 series router or a Cisco 3660 as a CAC slave.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

Configuring Dial Peers for AAL2 Voice

For more information on dial peers and dial-peer configuration, see the “Configuring Voice over ATM” chapter in the *Cisco IOS Multiservice Applications Configuration Guide*, Release 12.1.

Configuring Network Dial Peers to Support AAL2

To configure a network dial peer for Voice over ATM (VoATM), specify a unique tag number, an atm, a virtual circuit number, and channel identifier (CID).

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 enter dial-peer configuration mode. The <i>tag</i> identifies the dial peer. Each <i>tag</i> on any one router must be unique.
Step 2	Router(config-dial-peer)# destination-pattern string	Configures the dial peer’s destination pattern. The <i>string</i> is a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9 and the letters A through D. The following special characters can be entered in the string: <ul style="list-style-type: none"> • The star (*) and the pound sign (#) can be used in a dial string, but not as leading characters (for example *650 is not permitted). • The period (.) can be entered as a wildcard digit. Network dial peers typically use wildcards to represent a range of destination telephone numbers (for example, 1408555... for all numbers in area code 408 with the 555 prefix). • The comma (,) can be used only in prefixes, and is used to insert a 1-second pause. • The timer (T) character can be used to configure variable-length dial plans.
Step 3	Router(config-dial-peer)# session protocol aal2-trunk	Configures the session protocol to support AAL2-trunk permanent (private line) trunk calls.

	Command	Purpose
Step 4	<pre>Router(config-dial-peer)# session target atm 0/0 pvc {word vpi/vci vci} cid (for Cisco 2600 series) For Cisco 3660: router(config-dial-peer)# session target atm <slot>/ima <group#></pre>	<p>Configures the ATM session target for the dial peer. Be sure to specify atm 0/0 as the interface for the PVC.</p> <p>Use <i>word</i> to identify the PVC if a word name was assigned when the PVC was created in the “Configuring ATM on Cisco 2600 Series” section on page 7.</p> <p>Use <i>word</i> to identify the PVC if a word name was assigned when the PVC was created in the “Configuring ATM on Cisco 3660” section on page 11.</p>
Step 5	<pre>Router(config-dial-peer)# codec aal2 profile {itut custom} profile-number codec</pre>	<p>Specifies a codec profile for the DSP.</p> <p>Profile options are itut 1, itut 2, itut 7, custom 100, and custom 110.</p> <p>The default is itut 1 with codec G.711 u-law.</p> <p>See the “Command Reference” section for the codec options available for each AAL2 profile.</p> <p>Note Use this command instead of the codec (dial-peer) command for AAL2 trunk applications.</p>
Step 6	<pre>Router(config-dial-peer)# dtmf-relay</pre>	<p>(Optional) If the codec type is a low bit-rate codec such as g729 or g723, specify 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.</p> <p>DTMF relay is disabled by default.</p>
Step 7	<pre>Router(config-dial-peer)# signal-type {ext-signal transparent}</pre>	<p>(Optional) Defines the type of ABCD signaling packets that are generated by the voice port and sent over the ATM network. The signal type must be configured to the same setting at both ends of the PVC.</p> <p>Enter ext-signal for common channel signaling (CCS). ABCD signaling packets are not sent.</p> <p>Enter transparent for nonswitched trunks using channel associated signaling (CAS). ABCD signaling bits are passed transparently to the ATM network.</p>
Step 8	<pre>Router(config-dial-peer)# no vad</pre>	<p>(Optional) Disables voice activity detection (VAD) on the dial peer. VAD is enabled by default.</p>
Step 9	<pre>Router(config-dial-peer)# exit</pre>	<p>Exits from the dial-peer configuration mode.</p>
Step 10	<pre>Repeat Step 1 through Step 9</pre>	<p>Configures additional VoATM dial peers.</p>

Configuring MGCP POTS Dial Peer

To configure MGCP POTS dial peer on the Cisco 2600 series and the Cisco 3660, complete the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	<code>Router(config)# dial-peer voice number pots</code>	Enters the dial-peer configuration mode for the POTS dial-peer.
Step 2	<code>Router(config-dial-peer)# application MGCPAPP</code>	Initiates the MGCP protocol for the voice ports.
Step 3	<code>Router(config-dial-peer)# port slot/port:ds0-group</code>	This command associates the dial peer with a specific logical interface. The value of <i>slot</i> is the router location where the voice port adapter is installed. Valid entries are from 0 to 3. The value of <i>port</i> indicates the voice interface card location. Valid entries are 0 or 1. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
Step 4	<code>Router(config-dial-peer)# exit</code>	Exits from the dial-peer configuration mode.

Configuring DS-0 Group and Channel Group for CCS

To configure a DS-0 group and the channel group for CCS on the Cisco 2600 series and the Cisco 3660, complete the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	<code>Router(config)# controller {T1 E1} slot/port</code>	Enters controller configuration mode for the controller at the specified slot/port location. Valid values for slot and port are 0 and 1.
Step 2	<code>Router(config-controller)# mode ccs frame-forwarding</code>	Configures the controller to support CCS transparent signaling.

	Command	Purpose
Step 3	Router(config-controller)# channel-group <i>number</i> timeslots <i>range</i> speed {48 56 64}	Defines the time slots that belong to each T1 or E1 circuit. <ul style="list-style-type: none"> • number—channel-group number. When configuring a T1 data line, channel-group numbers can be values from 0 to 23. When configuring an E1 data line, channel-group numbers can be a values from 0 to 30. • range—time slot or range of time slots belonging to the channel group. The first timeslot is numbered 1. Pick one timeslot from the timeslot range. For a T1 controller, the timeslot range is from 1 to 24. For an E1 controller, the timeslot range is from 1 to 31. • speed {48 56 64}—(Optional) Specifies the line speed (in kilobits per second) of the T1 or E1 link. The default line speed for T1 is 56 kbps. The default line speed for E1 is 64 kbps.
Step 4	Router(config-controller)# ds0-group <i>channel-number</i> timeslots <i>range</i> type <i>signaling-type</i>	<i>For the CCS PBX Scenarios only:</i> Configures the T1 time slots for CCS calls. The signaling type is external signaling.
Step 5	Router(config-controller)# exit	<i>For the CCS PBX Scenarios only:</i> Exits controller configuration mode.

Configuring T-CCS Frame Forwarding

To configure T-CCS frame-forwarding on the Cisco 2600 series and the Cisco 3660, complete the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# interface serial <i>1:channelnumber</i>	Enters interface configuration mode. This procedure maps the D channel from the digital T1/E1 packet voice trunk network module to the specified interface. The channel number argument specifies the channel number. For T1, enter the channel number as 23. For E1, enter 15.
Step 2	Router(config-if)# ccs encap atm	(ATM only) Configures the CCS encapsulation to use the ATM packet format.
Step 3	Router(config-if)# ccs connect { serial atm } <i>slot/number</i> [dlci dlci pvc vci pvc vcd pvc vpi/vci pvc string]	(Frame Relay and ATM) Configures the CCS connection. If the CCS connection is over Frame Relay, specify a serial interface and the DLCI. If the CCS connection is over ATM, specify ATM, slot and interface, and the PVC.
Step 4	Router(config-if)# exit	Exits the interface mode.

Configuration Tasks for MGCP CAS

See the following sections for configuration tasks for MGCP CAS on Cisco 2600 series and Cisco 3660 routers. Each task in this list is identified as either required or optional:

- [Configuring MGCP CAS PBX on the Cisco 2600 Series and Cisco 3660, page 35](#) (required)
- [Configuring ATM on the Cisco 2600 Series, page 38](#) (required only for Cisco 2600 series)
- [Configuring ATM on the Cisco 3660, page 41](#)(required only for Cisco 3660)
- [Configuring Voice Band Detection Playout Settings, page 45](#) (optional)
- [Configuring Subcell Multiplexing for AAL2 Voice, page 46](#) (required)
- [Verifying the MGCP CAS PBX and AAL2 PVC Configurations, page 46](#) (optional)
- [Configuring End-to-End Clocking, page 47](#) (required)
- [Configuring Call Admission Control for AAL2 Voice, page 47](#) (required)
- [Configuring MGCP POTS Dial Peer, page 48](#) (required)

Configuring MGCP CAS PBX on the Cisco 2600 Series and Cisco 3660

Use the following commands for configuring the Media Gateway Control Protocol (MGCP) CAS PBX on Cisco 2600 series and Cisco 3660 routers:

	Command	Purpose
Step 1	Router(config)# mgcp	Starts the MGCP daemon.
Step 2	Router(config)# mgcp call-agent {ipaddr host-name} [port] [service-type type] version version-number	Configures the MGCP call agent and service type.
Step 3	Router(config-if)# mgcp sgcp restart notify	Causes MGCP to send and process simple gateway control protocol (SGCP) restart in progress (RSIP) messages.
Step 4	Router(config-if)# mgcp modem passthrough [cisco ca nse]	Enables the router to process fax or modem messages. VoAAL2 does not support cisco .
Step 5	Router(config)# mgcp tse payload type	Enables the telephony signaling events (TSE) payload for fax and modem messages.
Step 6	Router(config)# no mgcp timer receive-rtcp	Turns off the real-time transport protocol (RTP) RTP control protocol (RTCP) transmission interval at the gateway.
Step 7	Router(config)# mgcp timer rtp-nse timer	Turns on the RTP named signaling events (NSE) timeout interval at the gateway.
Step 8	Router(config)# mgcp quarantine mode process	(Optional) Turns on processing for MGCP quarantine mode.
Step 9	Router(config)# controller {t1 e1} slot/port	<i>For the CAS PBX scenarios only:</i> Selects the T1/E1 controller 1/0.
Step 10	Router(config-controller)# mode cas	<i>For the CAS PBX scenarios only:</i> Specifies that the controller will support CAS.

	Command	Purpose
Step 11	<pre>Router(config-controller)# ds0-group channel-number timeslots range type signaling-type tone type addr info service service-type</pre>	<p><i>For the CAS PBX scenarios only:</i> Configures the T1 time slots for CAS calls. The scenarios use the following three digital signal level 0 (DS-0) definitions:</p> <ul style="list-style-type: none"> • ds0-group 1 time slots 1-8 type e&m-immediate-start • ds0-group 2 time slots 9-16 type e&m-wink-start • ds0-group 3 time slots 17-24 type fxs-ground-start
Step 12	<pre>Router(config-controller)# exit</pre>	<p><i>For the CAS PBX scenarios only:</i> Exits controller configuration mode.</p>

	Command	Purpose
Step 13	<pre>Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]</pre>	<p>Configure the PVC for the variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> <i>peak rate</i>—If it does not exceed your carrier's allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). <i>average rate</i>—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: For VoIP: G.711 with 169 or 80 byte sample size: <i>max calls</i> x 128K G.726 with 40 byte sample size: <i>max calls</i> x 85K G.729a with 10 byte sample size: <i>max calls</i> x 85K For VoAAL2: G.711 with 40 byte sample size: <i>max calls</i> x 85K G.726 with 40 byte sample size: <i>max calls</i> x 43K G.729a with 30 byte sample size: <i>max calls</i> x 15K <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <i>burst</i>— Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: The minimum burst size is 4 times the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: http://www-vnt/SPUniv/DSP/Codec_Calc1.asp
Step 14	<pre>Router(config-if-atm-vc)# vcci pvc-identifier</pre>	Assigns a unique identifier to the PVC.
Step 15	<pre>Router(config-if-atm-vc)# exit</pre>	Exits ATM virtual circuit configuration mode.
Step 16	<pre>Router(config-if)# exit</pre>	Exits interface configuration mode.
Step 17	<pre>Router(config)# dial-peer voice number pots</pre>	Enters dial-peer configuration mode for the POTS dial peer.
Step 18	<pre>Router(config-dial-peer)# application MGCPAPP</pre>	Initiates MGCP for the voice ports. You can enter the MGCPAPP keyword in either uppercase or lowercase.

Configuring ATM on the Cisco 2600 Series

This section describes the ATM configuration tasks necessary to support Voice over ATM using AAL2 on Cisco 2600 series.



Note

If any DS0 groups (CAS groups), channel groups, or clear channels are configured on T1/E1 controller 0, you must remove them before configuring VoATM. Because ATM uses all the DS-0 time slots on the controller, the ATM configuration cannot take place if any DS0s on controller 0 are used by other applications.

You must perform the VoATM configuration on the Cisco 2600 series or Cisco 3660 concentrators at both ends of the ATM link.

To configure a Cisco 2600 series or Cisco 3660 series concentrator to support VoATM on a T1/E1 trunk, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# controller {t1 e1} 0/0	Selects the T1 or E1 controller 0/0.
Step 2	Router(config-controller)# mode atm	Specifies that the controller will support ATM encapsulation and create ATM interface 0/0.
Step 3	Router(config-controller)# framing <i>framing type</i>	Specifies the T1 framing type as extended super frame or esf . When the controller is set to ATM mode, the Controller framing is automatically set to Extended SuperFrame (esf) on T1 and to cr4 on E1.
Step 4	Router(config-controller)# linecode <i>linecode type</i>	Specifies the T1 linecode type as b8zs . The linecode is automatically set to b8zs on T1 and to hdb3 on E1.
Step 5	Router(config-controller)# no shutdown	Ensures that the controller is activated.
Step 6	Router(config-controller)# exit	Exits controller configuration mode.
Step 7	Router(config)# interface <i>atmslot/port</i> [<i>subinterface-number</i> [multipoint point-to-point]]	Enters interface configuration mode to configure ATM interface 0/0 or an ATM subinterface. For subinterfaces, the default is multipoint .
Step 8	Router(config-if)# pvc [<i>word</i>] { <i>vpi/vci</i> <i>vci</i> }	Creates an ATM PVC for voice traffic and enters ATM virtual circuit configuration mode. <i>vpi</i> = 0 to 255 <i>vci</i> = 1 to 1023 <i>word</i> = optional PVC identifier (letters only); if you assign a PVC identifier, you can use it to specify this PVC when configuring network dial peers Note AAL2 encaps is not supported for ilmi and qsaal PVCs.
Step 9	Router(config-if-atm-vc)# encapsulation aal2	Sets the encapsulation of the PVC to support AAL2 voice traffic. This automatically creates channel identifiers (CIDs) 1 through 255.

Command	Purpose
<p>Step 10 Router(config-if-atm-vc)# vbr-rt <i>peak-rate</i> <i>average-rate</i> [<i>burst</i>]</p>	<p>Configures the PVC for variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>peak-rate</i>—If it does not exceed your carrier's allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>average-rate</i>—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size—<i>max calls</i> x 85 G.726 with 40 or 80 byte sample size—<i>max calls</i> x 43 G.729 with 30 byte sample size—<i>max calls</i> x 15 G.729 with 20 byte sample size—<i>max calls</i> x 22 G.729 with 30 byte sample size—<i>max calls</i> x 15 <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <ul style="list-style-type: none"> • <i>burst</i>—Set the burst size as large as possible, and never less than the minimum burst size. <p>Guidelines are as follows:</p> <p>The minimum burst size is 4 x the number of voice calls.</p> <p>The maximum burst size is the maximum allowed by the carrier.</p> <p>You can calculate the value using the calculator at the following URL:</p> <p>http://www.vnt/SPUniv/DSP/Codec_Calc1.asp</p>
<p>Step 11 Router(config-if-atm-vc)# oam-pvc [manage] [<i>frequency</i>]</p>	<p>(Optional) Configures transmission of end-to-end F5 OAM loopback cells on a PVC, optionally specifies the number of seconds between loopback cells, and optionally enables OAM management of the connection.</p> <p>The range for <i>frequency</i> is 0 to 600. The default is 10.</p>

	Command	Purpose
Step 12	Router(config-if-atm-vc)# oam retry <i>up-count</i> <i>down-count</i> <i>retry-frequency</i>	<p>(Optional) Specifies OAM management parameters for verifying connectivity of a PVC connection. This command is supported only if OAM management is enabled.</p> <ul style="list-style-type: none"> • The value of <i>up-count</i> is the number of OAM loopback cell responses received to change the PVC connection to up. The range is 1 to 600; the default is 3. • The value of <i>down-count</i> is the number of OAM loopback cell responses not received to change the PVC connection to down. The range is 1 to 600; the default is 5. • The value of <i>retry-frequency</i> is the number of seconds between loopback cells sent to verify the down state of a PVC. The range is 1 to 1000; the default is 1. <p>Note Enter the oam retry command only once with all the arguments in the order shown. The first number always specifies <i>up-count</i>; the second <i>down-count</i>, and the third <i>retry-frequency</i>.</p>
Step 13	Router(config-if-atm-vc)# end	Exits configuration mode.
Step 14	Router# show atm vc	Verifies the ATM PVC configuration.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring ATM on the Cisco 3660

Step 1	<pre>Router(config)# interface atm <slot>/ima<grp#> [subinterface-number [multipoint point-to-point]]</pre>	<p>Enters interface configuration mode to configure ATM interface 0/0 or an ATM subinterface.</p> <p>Note To configure an IMA group on each ATM interface, enter the IMA group and group number.</p> <p>The default for subinterfaces is multipoint.</p> <p><i>For all Scenarios:</i> Set up three subinterfaces for point-to-point.</p>
Step 2	<pre>Router(config-if)# pvc [name] vpi/vci</pre>	<p>Creates an ATM PVC for voice traffic and enters ATM virtual circuit configuration mode.</p> <p>Note AAL2 encap is not supported for ilmi and qsaal PVCs.</p>
Step 3	<pre>Router(config-if-atm-vc)# encapsulation aal2</pre>	<p>Sets the encapsulation of the PVC to support AAL2 voice traffic. This automatically creates channel identifiers (CIDs) 1 through 255.</p>

Step 4	<pre>Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]</pre>	<p>Configures the PVC for variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>peak rate</i>—If it does not exceed your carrier's allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>average rate</i>—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size—$max\ calls \times 85$ G.726 with 40 or 80 byte sample size—$max\ calls \times 43$ G.729 with 30 byte sample size—$max\ calls \times 15$ G.729 with 20 byte sample size—$max\ calls \times 22$ G.729 with 10 byte sample size—$max\ calls \times 43$ <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> • <i>burst size</i>—Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: <ul style="list-style-type: none"> The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: <ul style="list-style-type: none"> http://www-vnt/SPUniv/DSP/Codec_Calc1.asp
Step 5	<pre>Router(config-if-atm-vc)# oam-pvc [manage] [frequency]</pre>	<p>(Optional) Configures transmission of end-to-end F5 OAM loopback cells on a PVC, optionally specify the number of seconds between loopback cells, and optionally enable OAM management of the connection.</p> <p>The range for <i>frequency</i> is 0 to 600. The default is 10.</p>

Step 6	<pre>Router(config-if-atm-vc)# oam retry up-count down-count retry-frequency</pre>	<p>(Optional) Specifies OAM management parameters for verifying connectivity of a PVC connection. This command is supported only if OAM management is enabled.</p> <ul style="list-style-type: none"> • The value of <i>up-count</i> is the number of OAM loopback cell responses received to change the PVC connection to up. The range is 1 to 600; the default is 3. • The value of <i>down-count</i> is the number of OAM loopback cell responses not received to change the PVC connection to down. The range is 1 to 600; the default is 5. • The value of <i>retry-frequency</i> is the number of seconds between loopback cells sent to verify the down state of a PVC. The range is 1 to 1000; the default is 1. <p>Note Enter the oam retry command only once with all the arguments in the order shown. The first number always specifies <i>up-count</i>; the second <i>down-count</i>, and the third <i>retry-frequency</i>.</p>
Step 7	<pre>Router(config-if-atm-vc)# end</pre>	Exits configuration mode.
Step 8	<pre>Router# show atm vc</pre>	Verifies the ATM PVC configuration.

Step 9	Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]	<p>Configures the PVC for the variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>peak rate</i>—If it does not exceed your carrier’s allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>average rate</i>—Calculate according to the maximum number of calls (<i>max calls</i>) the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> For VoIP: <p>G.711 with 40 or 80 byte sample size: <i>max calls</i> x 128K</p> <p>G.726 with 40 byte sample size: <i>max calls</i> x 85K</p> <p>G.729a with 10 byte sample size: <i>max calls</i> x 85K</p> For VoAAL2: <p>G.711 with 40 byte sample size: <i>max calls</i> x 85K</p> <p>G.726 with 40 byte sample size: <i>max calls</i> x 43K</p> <p>G.729a with 10 byte sample size: <i>max calls</i> x 43K</p> <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <ul style="list-style-type: none"> • <i>burst</i>—Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: <ul style="list-style-type: none"> The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: <p>http://www-vnt/SPUniv/DSP/Codec_Calc1.asp</p>
Step 10	Router(config-if-atm-vc)# vcci pvc-identifier	Assigns a unique identifier to the PVC.
Step 11	Router(config-if-atm-vc)# exit	Exits ATM virtual circuit configuration mode.
Step 12	Router(config-if)# exit	Exits interface configuration mode.
Step 13	Router(config)# dial-peer voice number pots	Enter dial peer configuration mode for the POTS dial peer.
Step 14	Router(config-dial-peer)# application MGCPAPP	Initiates the MGCP protocol for the voice ports.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring Voice Band Detection Playout Settings

To configure voice band detection playout buffer delay on Cisco 2600 series and Cisco 3600 series routers, use the following commands beginning in the voice service configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# vbd-playout-delay maximum time	Specifies the maximum AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers in milliseconds. The time is set in milliseconds. The range is from 40-1700 milliseconds. The default is set to 200 milliseconds.
Step 4	Router(config-voice-service-session)# vbd-playout-delay minimum time	Specifies the minimum AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers. The time is set in milliseconds. The range is from 4-1700 milliseconds. The default is set to 4 milliseconds.
Step 5	Router(config-voice-service-session)# vbd-playout-delay mode {fixed passthrough}	Configures voice band detection playout delay adaptation mode on a Cisco router. When the vbd-delay-playout mode is set to fixed, jitter buffer is set at a constant delay in milliseconds. When the vbd-delay-playout mode is set to passthrough, jitter buffer is set to DRAIN_FILL for clock compensation. There is no default.
Step 6	Router(config-voice-service-session)# vbd-playout-delay nominal time	Specifies the nominal AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers. The time is set in milliseconds. The range is from 0-1500 milliseconds. The default is 100 milliseconds.
Step 7	Router(config-voice-service-session)# end	Exits voice-service-session configuration mode.

Configuring Subcell Multiplexing for AAL2 Voice

This section describes the configuration tasks necessary to enable AAL2 common part sublayer (CPS) subcell multiplexing when the Cisco 2600 series router or a Cisco 3660 interoperates with a voice interface service module (VISM) in an MGX switch.

To configure the Cisco 2600 series router or the Cisco 3660 to perform subcell multiplexing, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# subcell-mux number	Enables subcell multiplexing. The number is time in milliseconds. By default, subcell multiplexing is not enabled.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

Verifying the MGCP CAS PBX and AAL2 PVC Configurations

Use these commands to verify the configuration settings:

	Command	Purpose
Step 1	Router# show dial-peer voice sum	Displays the status of the dial peer. The dial peer should be active. If it is not, enter the command: Router(config-dial-peer)# no shut
Step 2	Router# show run	Displays the current configuration settings.

Configuring End-to-End Clocking


Note

The following commands can be used to configure the Cisco 3660 only when there is a TDM switch module on board. For the Cisco 2600 series routers, these commands are automatically allowed.

	Command	Purpose
Step 1	Router(config)# network-clock-participate {nm wic} slot	Enables the Cisco 2600 series router or the Cisco 3660 to receive clock signals from the VWIC by entering the keyword wic and the slot number 0 on the router.
Step 2	for Cisco 2600 series: Router(config)# network-clock-participate {nm wic} slot for Cisco 3660: Router(config)# network-clock-participate nm slot	Enables the Cisco 2600 series router or the Cisco 3660 to receive clock signals from the network module by entering the keyword nm and the slot number 1 on the router.
Step 3	Router(config) network-clock-select priority t1 slot/port	Names a source to provide timing for the network clock and to specify the selection <i>priority</i> for this clock source. The priority selection is 1 or 2 . Use the no form of this command to cancel the selection.
Step 4	Router(config) network-clock-select priority t1 slot/port	Assigns <i>priority 1</i> to ATM interface 0/0 and <i>priority 2</i> to controller 1/0.
Step 5	Router(config) voice-card slot	Enters voice-card configuration mode and sets codec complexity. For <i>slot</i> , use a value from 0 to 3 that describes the card location in the module.


Note

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring Call Admission Control for AAL2 Voice

This section describes the configuration tasks necessary to configure call admission control (CAC) for AAL2 voice. The commands and procedures in this section are common to the Cisco 2600 series and the Cisco 3660 routers.

You can configure a Cisco 2600 series router or a Cisco 3660 as either a CAC master or a CAC slave. By default, this is a CAC slave. You typically configure a CAC master at one end of an ATM trunk and a CAC slave at the opposite end. A Cisco 2600 series router or a Cisco 3660 configured as a master always performs CAC during fax/modem upspeed. A Cisco 2600 series router or a Cisco 3660 configured as a slave sends a request for CAC to the CAC master.

To configure a Cisco 2600 series router or a Cisco 3660 as a CAC master, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# cac master	Configures this Cisco 2600 series router or a Cisco 3660 as a CAC master.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

To return a Cisco 2600 series router or a Cisco 3660 to its default operation as a CAC slave, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# no cac master	Configures this Cisco 2600 series router or a Cisco 3660 as a CAC slave.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

Configuring MGCP POTS Dial Peer

To configure MGCP POTS dial peer on the Cisco 2600 series and Cisco 3660, complete the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# dial-peer voice number pots	Enters dial-peer configuration mode for the POTS dial peer.
Step 2	Router(config-dial-peer)# application MGCPAPP	Initiates MGCP for the voice ports.
Step 3	Router(config-dial-peer)# port slot/port:ds0-group	Associates the dial peer with a specific logical interface. The value of <i>slot</i> is the router location where the voice port adapter is installed. Valid entries are from 0 to 3. The value of <i>port</i> indicates the voice interface card location. Valid entries are 0 or 1. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
Step 4	Router(config-dial-peer)# exit	Exits from the dial-peer configuration mode.

Troubleshooting Tips

- For a good voice quality and to be able to make fax calls, make sure that you configure end-to-end clocking properly, that is, make sure that the T1 controllers participating in this configuration do not have any errors.
- Make sure that you do not configure bisync tunnelling protocol (BSTUN) and ATM on the Cisco 2650 router simultaneously.

Configuration Tasks for MGCP PRI Backhaul

See the following sections for configuration tasks for MGCP PRI Backhaul for Cisco 2600 series and Cisco 3660 routers:

- [Configuring MGCP CAS PBX on the Cisco 2600 Series and Cisco 3660, page 49](#) (required)
- [Configuring ATM on the Cisco 2600 Series, page 52](#) (required only for Cisco 2600 series)
- [Configuring ATM on the Cisco 3660, page 56](#) (required only for Cisco 3660)
- [Configuring Voice Band Detection Playout Settings, page 60](#) (optional)
- [Configuring Subcell Multiplexing for AAL2 Voice, page 61](#) (optional)
- [Verifying the MGCP CAS PBX and AAL2 PVC Configurations, page 61](#) (optional)
- [Configuring End-to-End Clocking, page 62](#) (required)
- [Configuring Call Admission Control for AAL2 Voice, page 62](#) (required)
- [Configuring Backhaul Session Manager, page 63](#) (required)
- [Configuring ISDN Signaling Backhaul, page 66](#) (required)
- [Configuring Fast Ethernet for Signaling Backhaul Compatibility, page 67](#) (required)
- [Configuring the Cisco VSC3000, page 68](#) (required)
- [Configuring MGCP POTS Dial Peer, page 73](#) (required)

Configuring MGCP CAS PBX on the Cisco 2600 Series and Cisco 3660

Use the following commands for configuring the Media Gateway Control Protocol (MGCP) CAS PBX on the Cisco 2600 series and the Cisco 3660 routers:

	Command	Purpose
Step 1	Router(config)# mgcp	Starts the MGCP daemon.
Step 2	Router(config)# mgcp call-agent {ipaddr / host-name} [port] [service-type type] version version-number	Configures the MGCP call agent and service type.
Step 3	Router(config-if)# mgcp sgcp restart notify	Causes MGCP to send and process SGCP RSIP messages.
Step 4	Router(config-if)# mgcp modem passthrough [cisco ca nse]	Enables the router to process fax or modem messages. VoAAL2 does not support cisco .

	Command	Purpose
Step 5	Router(config)# mgcp tse payload type	Enables the TSE payload for fax and modem messages.
Step 6	Router(config)# no mgcp timer receive-rtcp	Turns off the RTP RTCP transmission interval at the gateway.
Step 7	Router(config)# mgcp timer rtp-nse timer	Turns on the RTP NSE timeout interval at the gateway.
Step 8	Router(config)# mgcp quarantine mode process	(Optional) Turns on processing for MGCP quarantine mode.
Step 9	Router(config)# controller {t1 e1} slot/port	<i>For the CAS PBX scenarios only:</i> Selects the T1/E1 controller 1/0.
Step 10	Router(config-controller)# mode cas	<i>For the CAS PBX scenarios only:</i> Specifies that the controller will support CAS.
Step 11	Router(config-controller)# ds0-group channel-number timeslots range type signaling-type tone type addr info service service-type	<i>For the CAS PBX scenarios only:</i> Configures the T1 time slots for CAS calls. The scenarios use the following three digital signal level 0 (DS-0) definitions: <ul style="list-style-type: none"> • ds0-group 1 time slots 1-8 type e&m-immediate-start • ds0-group 2 time slots 9-16 type e&m-wink-start • ds0-group 3 time slots 17-24 type fxs-ground-start
Step 12	Router(config-controller)# exit	<i>For the CAS PBX scenarios only:</i> Exits controller configuration mode.

	Command	Purpose
Step 13	<pre>Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]</pre>	<p>Configures the PVC for the variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> <i>peak rate</i>—If it does not exceed your carrier's allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). <i>average rate</i>—Calculate according to the maximum number of calls (<i>max calls</i>) the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> For VoIP: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size: $max\ calls \times 128K$ G.726 with 40 byte sample size: $max\ calls \times 85K$ G.729a with 10 byte sample size: $max\ calls \times 85K$ For VoAAL2: <ul style="list-style-type: none"> G.711 with 40 byte sample size: $max\ calls \times 85K$ G.726 with 40 byte sample size: $max\ calls \times 43K$ G.729a with 10 byte sample size: $max\ calls \times 43K$ <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <i>burst</i>—Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: <ul style="list-style-type: none"> The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: <ul style="list-style-type: none"> http://www.vnt/SPUniv/DSP/Codec_Calc1.asp
Step 14	<pre>Router(config-if-atm-vc)# vcci pvc-identifier</pre>	Assigns a unique identifier to the PVC.
Step 15	<pre>Router(config-if-atm-vc)# exit</pre>	Exits ATM virtual circuit configuration mode.
Step 16	<pre>Router(config-if)# exit</pre>	Exits interface configuration mode.

	Command	Purpose
Step 17	Router(config)# dial-peer voice <i>number</i> pots	Enters dial-peer configuration mode for the POTS dial peer.
Step 18	Router(config-dial-peer)# application MGCPAPP	Initiates MGCP for the voice ports. You can enter the MGCPAPP keyword in either uppercase or lowercase.

Configuring ATM on the Cisco 2600 Series

This section describes the ATM configuration tasks necessary to support Voice over ATM using AAL2 on Cisco 2600 series.



Note

If any DS0 groups (CAS groups), channel groups, or clear channels are configured on T1/E1 controller 0, you must remove them before configuring VoATM. Because ATM uses all of the DS0 time slots on the controller, the ATM configuration cannot take place if any DS0s on controller 0 are used by other applications.

You must perform the VoATM configuration on the Cisco 2600 series or Cisco 3660 concentrators at both ends of the ATM link.

To configure a Cisco 2600 series or Cisco 3660 series concentrator to support VoATM on a T1/E1 trunk, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# controller {t1 e1} 0/0	Selects the T1 or E1 controller 0/0.
Step 2	Router(config-controller)# mode atm	Specifies that the controller will support ATM encapsulation and create ATM interface 0/0.
Step 3	Router(config-controller)# framing <i>framing type</i>	Specifies the T1 framing type as Extended SuperFrame (esf). When the controller is set to ATM mode, the Controller framing is automatically set to Extended SuperFrame (esf) on T1 and to crc4 on E1.
Step 4	Router(config-controller)# linecode <i>linecode type</i>	Specifies the T1 linecode type as b8zs . The linecode is automatically set to b8zs on T1 and to hdb3 on E1.
Step 5	Router(config-controller)# no shutdown	Ensures that the controller is activated.
Step 6	Router(config-controller)# exit	Exits controller configuration mode.
Step 7	Router(config)# interface <i>atmslot/port</i> [<i>subinterface-number</i> [multipoint point-to-point]]	Enters interface configuration mode to configure ATM interface 0/0 or an ATM subinterface. For subinterfaces, the default is multipoint .

	Command	Purpose
Step 8	Router(config-if)# pvc [<i>word</i>] { <i>vpi/vci</i> <i>vci</i> }	<p>Creates an ATM PVC for voice traffic and enters ATM virtual circuit configuration mode.</p> <p><i>vpi</i>= 0 to 255</p> <p><i>vci</i>= 1 to 1023</p> <p><i>word</i>= optional PVC identifier (letters only); if you assign a PVC identifier, you can use it to specify this PVC when configuring network dial peers</p> <p>Note AAL2 encap is not supported for ilmi and qsaal PVCs.</p>
Step 9	Router(config-if-atm-vc)# encapsulation aal2	<p>Sets the encapsulation of the PVC to support AAL2 voice traffic. This automatically creates channel identifiers (CIDs) 1 through 255.</p>

Command	Purpose
<p>Step 10 Router(config-if-atm-vc) # vbr-rt <i>peak-rate</i> <i>average-rate</i> [<i>burst</i>]</p>	<p>Configures the PVC for variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>peak-rate</i>—If it does not exceed your carrier's allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>average-rate</i>—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size—<i>max calls</i> x 85 G.726 with 40 or 80 byte sample size—<i>max calls</i> x 43 G.729 with 30 byte sample size—<i>max calls</i> x 15 G.729 with 20 byte sample size—<i>max calls</i> x 22 G.729 with 30 byte sample size—<i>max calls</i> x 15 <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <ul style="list-style-type: none"> • <i>burst</i>—Set the burst size as large as possible, and never less than the minimum burst size. <p>Guidelines are as follows:</p> <p>The minimum burst size is 4 x the number of voice calls.</p> <p>The maximum burst size is the maximum allowed by the carrier.</p> <p>You can calculate the value using the calculator at the following URL:</p> <p>http://www-vnt/SPUniv/DSP/Codec_Calc1.asp</p>
<p>Step 11 Router(config-if-atm-vc) # oam-pvc [manage] <i>frequency</i></p>	<p>(Optional) Configures transmission of end-to-end F5 OAM loopback cells on a PVC, optionally specifies the number of seconds between loopback cells, and optionally enables OAM management of the connection.</p> <p>The range for <i>frequency</i> is 0 to 600. The default is 10.</p>

	Command	Purpose
Step 12	Router(config-if-atm-vc)# oam retry <i>up-count</i> <i>down-count</i> <i>retry-frequency</i>	<p>(Optional) Specifies OAM management parameters for verifying connectivity of a PVC connection. This command is supported only if OAM management is enabled.</p> <ul style="list-style-type: none"> The value of <i>up-count</i> is the number of OAM loopback cell responses received to change the PVC connection to up. The range is 1 to 600; the default is 3. The value of <i>down-count</i> is the number of OAM loopback cell responses not received to change the PVC connection to down. The range is 1 to 600; the default is 5. The value of <i>retry-frequency</i> is the number of seconds between loopback cells sent to verify the down state of a PVC. The range is 1 to 1000; the default is 1. <p>Note Enter the oam retry command only once with all the arguments in the order shown. The first number always specifies <i>up-count</i>; the second <i>down-count</i>, and the third <i>retry-frequency</i>.</p>
Step 13	Router(config-if-atm-vc)# end	Exits configuration mode.
Step 14	Router# show atm vc	Verifies the ATM PVC configuration.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring ATM on the Cisco 3660

Step 1	<pre>Router(config)# interface atm slot/ima grp# [subinterface-number [multipoint point-to-point]]</pre>	<p>Enters interface configuration mode to configure ATM interface 0/0 or an ATM subinterface.</p> <p>Note To configure an IMA group on each ATM interface, enter the IMA group and group number.</p> <p>The default for subinterfaces is multipoint.</p> <p><i>For all Scenarios:</i> Set up three subinterfaces for point-to-point.</p>
Step 2	<pre>Router(config-if)# pvc [name] vpi/vci</pre>	<p>Creates an ATM PVC for voice traffic and enters ATM virtual circuit configuration mode.</p> <p>Note AAL2 encap is not supported for ilmi and qsaal PVCs.</p>
Step 3	<pre>Router(config-if-atm-vc)# encapsulation aal2</pre>	<p>Sets the encapsulation of the PVC to support AAL2 voice traffic. This automatically creates channel identifiers (CIDs) 1 through 255.</p>

Step 4	<pre>Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]</pre>	<p>Configures the PVC for variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> • <i>peak rate</i>—If it does not exceed your carrier’s allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). • <i>average rate</i>—Calculate according to the maximum number of calls the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size—<i>max calls</i> x 85 G.726 with 40 or 80 byte sample size—<i>max calls</i> x 43 G.729 with 30 byte sample size—<i>max calls</i> x 15 G.729 with 20 byte sample size—<i>max calls</i> x 22 G.729 with 10 byte sample size—<i>max calls</i> x 43 <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <ul style="list-style-type: none"> • <i>burst size</i>—Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: <ul style="list-style-type: none"> The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: <ul style="list-style-type: none"> http://www-vnt/SPUniv/DSP/Codec_Calc1.asp
Step 5	<pre>Router(config-if-atm-vc)# oam-pvc [manage] [frequency]</pre>	<p>(Optional) Configures transmission of end-to-end F5 OAM loopback cells on a PVC, optionally specify the number of seconds between loopback cells, and optionally enable OAM management of the connection.</p> <p>The range for <i>frequency</i> is 0 to 600. The default is 10.</p>

Step 6	Router(config-if-atm-vc)# oam retry <i>up-count</i> <i>down-count</i> <i>retry-frequency</i>	<p>(Optional) Specifies OAM management parameters for verifying connectivity of a PVC connection. This command is supported only if OAM management is enabled.</p> <ul style="list-style-type: none"> • The value of <i>up-count</i> is the number of OAM loopback cell responses received to change the PVC connection to up. The range is 1 to 600; the default is 3. • The value of <i>down-count</i> is the number of OAM loopback cell responses not received to change the PVC connection to down. The range is 1 to 600; the default is 5. • The value of <i>retry-frequency</i> is the number of seconds between loopback cells sent to verify the down state of a PVC. The range is 1 to 1000; the default is 1. <p>Note Enter the oam retry command only once with all the arguments in the order shown. The first number always specifies <i>up-count</i>; the second <i>down-count</i>, and the third <i>retry-frequency</i>.</p>
Step 7	Router(config-if-atm-vc)# end	Exits configuration mode.
Step 8	Router# show atm vc	Verifies the ATM PVC configuration.

Step 9	<pre>Router(config-if-atm-vc)# vbr-rt peak-rate average-rate [burst]</pre>	<p>Configures the PVC for the variable-bit-rate real-time (voice) traffic. Guidelines for setting the peak rate, average rate, and burst size are as follows:</p> <ul style="list-style-type: none"> <i>peak rate</i>—If it does not exceed your carrier’s allowable rate, set to the line rate (for example, 1536 kbps for T1-ATM). <i>average rate</i>—Calculate according to the maximum number of calls (<i>max calls</i>) the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps: <ul style="list-style-type: none"> for VoIP: <ul style="list-style-type: none"> G.711 with 40 or 80 byte sample size: $max\ calls \times 128K$ G.726 with 40 byte sample size: $max\ calls \times 85K$ G.729a with 10 byte sample size: $max\ calls \times 85K$ for VoAAL2: <ul style="list-style-type: none"> G.711 with 40 byte sample size: $max\ calls \times 85K$ G.726 with 40 byte sample size: $max\ calls \times 43K$ G.729a with 10 byte sample size: $max\ calls \times 43K$ <p>If voice activity detection (VAD) is enabled, the bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.</p> <i>burst</i>—Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows: <ul style="list-style-type: none"> The minimum burst size is 4 x the number of voice calls. The maximum burst size is the maximum allowed by the carrier. You can calculate the value using the calculator at the following URL: <ul style="list-style-type: none"> http://www-vnt/SPUniv/DSP/Codec_Calc1.asp
Step 10	<pre>Router(config-if-atm-vc)# vcci pvc-identifier</pre>	Assigns a unique identifier to the PVC.
Step 11	<pre>Router(config-if-atm-vc)# exit</pre>	Exits ATM virtual circuit configuration mode.
Step 12	<pre>Router(config-if)# exit</pre>	Exits interface configuration mode.
Step 13	<pre>Router(config)# dial-peer voice number pots</pre>	Enter dial peer configuration mode for the POTS dial peer.
Step 14	<pre>Router(config-dial-peer)# application MGCPAPP</pre>	Initiates the MGCP protocol for the voice ports.

**Note**

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring Voice Band Detection Playout Settings

To configure voice band detection playout buffer delay on Cisco 2600 series and Cisco 3600 series routers, use the following commands beginning in the voice service configuration mode:

	Command	Purpose
Step 1	<code>Router(config)# voice service voatm</code>	Enters voice-service configuration mode.
Step 2	<code>Router(config-voice-service)# session protocol aal2</code>	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	<code>Router(config-voice-service-session)# vbd-playout-delay maximum time</code>	Specifies the maximum AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers in milliseconds. The time is set in milliseconds. The range is from 40-1700 milliseconds. The default is set to 200 milliseconds.
Step 4	<code>Router(config-voice-service-session)# vbd-playout-delay minimum time</code>	Specifies the minimum AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers. The time is set in milliseconds. The range is from 4-1700 milliseconds. The default is set to 4 milliseconds.
Step 5	<code>Router(config-voice-service-session)# vbd-playout-delay mode {fixed passthrough}</code>	Configures voice band detection playout delay adaptation mode on a Cisco router. When the vbd-delay-playout mode is set to fixed, jitter buffer is set at a constant delay in milliseconds. When the vbd-delay-playout mode is set to passthrough, jitter buffer is set to DRAIN_FILL for clock compensation. There is no default.
Step 6	<code>Router(config-voice-service-session)# vbd-playout-delay nominal time</code>	Specifies the nominal AAL2 voice band detection playout delay buffer on Cisco 2600 series and Cisco 3660 routers. The time is set in milliseconds. The range is from 0-1500 milliseconds. The default is 100 milliseconds.
Step 7	<code>Router(config-voice-service-session)# end</code>	Exits voice-service-session configuration mode.

Configuring Subcell Multiplexing for AAL2 Voice

This section describes the configuration tasks necessary to enable AAL2 common part sublayer (CPS) subcell multiplexing when the Cisco 2600 series router or a Cisco 3660 interoperates with a voice interface service module (VISM) in an MGX switch.

To configure the Cisco 2600 series router or the Cisco 3660 to perform subcell multiplexing, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# subcell-mux number	Enables subcell multiplexing. The number is time in milliseconds. By default, subcell multiplexing is not enabled.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

Verifying the MGCP CAS PBX and AAL2 PVC Configurations

Use these commands to verify the configuration settings:

	Command	Purpose
Step 1	Router# show dial-peer voice sum	Displays the status of the dial peer. The dial peer should be active. If it is not, enter the command: Router(config-dial-peer)# no shut
Step 2	Router# show run	Displays the current configuration settings.

Configuring End-to-End Clocking



Note

The following commands can be used to configure the Cisco 3660 only when there is a TDM switch module on board. For the Cisco 2600 series these commands are automatically allowed.

	Command	Purpose
Step 1	Router(config)# network-clock-participate { nm wic } <i>slot</i>	Enables the Cisco 2600 series router or the Cisco 3660 to receive clock signals from the VWIC by entering the keyword wic and the slot number 0 on the router.
Step 2	for Cisco 2600 series: Router(config)# network-clock-participate { nm wic } <i>slot</i> for Cisco 3660: Router(config)# network-clock-participate { nm } <i>slot</i>	Enables the Cisco 2600 series router or the Cisco 3660 to receive clock signals from the network module by entering the keyword nm and the slot number 1 on the router.
Step 3	Router(config) network-clock-select <i>priority t1 slot/port</i>	Names a source to provide timing for the network clock and to specify the selection <i>priority</i> for this clock source. The priority selection is 1 or 2 . Use the no form of this command to cancel the selection.
Step 4	Router(config) network-clock-select <i>priority t1 slot/port</i>	Assigns <i>priority 1</i> to ATM interface 0/0 and <i>priority 2</i> to controller 1/0.
Step 5	Router(config) voice-card <i>slot</i>	Enters voice-card configuration mode and set codec complexity. For <i>slot</i> , use a value from 0 to 3 that describes the card location in the module.



Note

When verifying your ATM PVC connectivity, note that you cannot enter 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 enter the **ping** command over the data PVC.

Configuring Call Admission Control for AAL2 Voice

This section describes the configuration tasks necessary to configure call admission control (CAC) for AAL2 voice. The commands and procedures in this section are common to the Cisco 2600 series and Cisco 3660.

You can configure a Cisco 2600 series router or a Cisco 3660 as either a CAC master or a CAC slave. By default, this is a CAC slave. You typically configure a CAC master at one end of an ATM trunk and a CAC slave at the opposite end. A Cisco 2600 series router or a Cisco 3660 configured as a master always performs CAC during fax/modem upspeed. A Cisco 2600 series router or a Cisco 3660 configured as a slave sends a request for CAC to the CAC master.

To configure a Cisco 2600 series router or a Cisco 3660 as a CAC master, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# cac master	Configures this Cisco 2600 series router or a Cisco 3660 as a CAC master.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

To return a Cisco 2600 series router or a Cisco 3660 to its default operation as a CAC slave, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voatm	Enters voice-service configuration mode.
Step 2	Router(config-voice-service)# session protocol aal2	Enters voice-service-session configuration mode and specifies AAL2 trunking.
Step 3	Router(config-voice-service-session)# no cac master	Configures this Cisco 2600 series router or a Cisco 3660 as a CAC slave.
Step 4	Router(config-voice-service-session)# end	Exits configuration mode.

Configuring Backhaul Session Manager

The backhaul session manager operates on the media gateway and enables signaling applications to backhaul signaling information to a remote or local virtual switch controller (VSC), and also provides redundancy and transparent management of transport paths.

To configure the backhaul session manager, log on to the media gateway and complete the following tasks as required for your application:

- [Creating Session Sets, Session Groups, and Sessions, page 64](#) (required)
- [Changing Default Values of Session-Group Parameters, page 65](#) (optional)

Creating Session Sets, Session Groups, and Sessions

To create session sets, session groups, and sessions on the Cisco media gateway, complete the following steps starting in global configuration mode:

	Command	Purpose
Step 1	Router(config)# backhaul-session-manager	Enters backhaul session manager configuration mode.
Step 2	Router(config-bsm)# set set-name client {ft nft}	<p>Creates a session set and specifies its parameters:</p> <ul style="list-style-type: none"> • <i>set-name</i>—A word you select to identify the session-set • client—Required for PRI backhaul; specifies that the session set function as a client • Fault-tolerance option: <ul style="list-style-type: none"> – ft = fault-tolerant – nft = non-fault-tolerant <p>Note For fault-tolerant operation, you must configure more than one group in this session set. If only one group will be configured in this session-set, you must specify nft.</p> <p>Note If you configure the session set for non-fault-tolerant operation, you should also configure the Cisco VSC3000 for non-fault-tolerant operation. See the “Configuring the Cisco VSC3000” section on page 68.</p>
Step 3	Router(config-bsm)# group group-name set set-name	<p>Adds a new session group to a specified session set.</p> <ul style="list-style-type: none"> • <i>group-name</i>—A word you select to identify the new session group • <i>set-name</i>—The session-set to which you are adding the new session group <p>Repeat this step to add additional session groups to a session set.</p>

	Command	Purpose
Step 4	<pre>Router(config-bsm)# session group group-name remote_ip remote_port local_ip local_port priority</pre>	<p>Adds a session to a session group and specifies the interfaces and selection priority for the session.</p> <ul style="list-style-type: none"> • <i>group-name</i>—The session group to which you are adding this session. • <i>remote_ip</i>—IP address of the Cisco VSC3000 server at the remote end of this backhaul link. • <i>remote_port</i>—The UDP port number on the Cisco VSC3000 server at the remote end of this backhaul link; the range is 1024 to 9999. Make sure that this number is not already being used by another service on the Cisco VSC3000, such as MGCP. • <i>local_ip</i>—The IP address of the media gateway port used for signaling backhaul. • <i>local_port</i>—The UDP port number of the media gateway port used for signaling backhaul; the range is 1024 to 9999 • <i>priority</i>—The priority within the session group. The range is 0 to 9999; 0 is the highest priority. <p>Note Although the Cisco IOS software allows you to configure multiple sessions with the same priority in a session group, Cisco Systems recommends that the priority of each session be unique within a session group.</p> <p>Repeat this step to create additional sessions in a session group.</p>

Changing Default Values of Session-Group Parameters

If you need to change the default values of session-group parameters, complete the following commands as required, in backhaul-session-manager configuration mode:



Caution

Do not change the session-group parameters unless instructed to do so by Cisco technical support. Sessions might fail if the relationships among parameters are not set correctly.

Command	Purpose
<pre>Router(config-bsm)# group group-name auto-reset number-of-auto-resets</pre>	<p>Specifies the maximum number of auto resets before the connection is considered failed.</p> <p>The range is 0 to 255. The default is 5.</p>
<pre>Router(config-bsm)# group group-name cumulative-ack number-of-segments</pre>	<p>Specifies the maximum number of (RUDP) segments that will be received before sending an acknowledgement.</p> <p>The range is 0 to 255. The default is 3.</p>

Command	Purpose
Router(config-bsm)# group <i>group-name</i> out-of-sequence <i>number-of-segments</i>	Specifies the maximum number of out-of-sequence segments that will be received before an acknowledgement is sent. The range is 0 to 255. The default is 3.
Router(config-bsm)# group <i>group-name</i> receive <i>window-size</i>	Specifies the maximum window size for the receiver. The range is 1 to 65. The default is 32.
Router(config-bsm)# group <i>group-name</i> retrans <i>resend-attempts</i>	Specifies the maximum number of times reliable user data protocol (RUDP) attempts to resend a segment before declaring the connection broken. The range is 0 to 255. The default is 2.
Router(config-bsm)# group <i>group-name</i> timer cumulative-ack <i>milliseconds</i>	Specifies the maximum number of milliseconds RUDP delays before sending an acknowledgement for a received segment. The range is 100 to 65535 . The default is 300.
Router(config-bsm)# group <i>group-name</i> timer keepalive <i>milliseconds</i>	Specifies the number of milliseconds RUDP waits before sending a keepalive segment. The range is 0 to 65535. The default is 200.
Router(config-bsm)# group <i>group-name</i> timer retransmit <i>milliseconds</i>	Specifies the number of milliseconds RUDP waits to receive an acknowledgement for a segment. The range is 100 to 65535. The default is 600.
Router(config-bsm)# group <i>group-name</i> timer transfer-state <i>milliseconds</i> Router(config-bsm)# exit	Specifies the number of milliseconds RUDP waits to receive a selection of a new session from the application during a transfer state. The range is 0 to 65535. The default is 600.

Configuring ISDN Signaling Backhaul

To configure the ISDN Q.931 signaling backhaul parameters, log on to the media gateway and complete the following steps starting in global configuration mode:

	Command	Purpose
Step 1	Router(config)# controller { <i>t1</i> <i>e1</i> } <i>controller-number</i>	Enters controller configuration mode and specifies a controller for the PRI interface. Controller numbers are: 0/0, 0/1, 1/0, 1/1, 2/0, 2/1, 3/0, 3/1, 4/0, 4/1, 5/0, 5/1
Step 2	Router(config-control)# pri-group <i>timeslots</i> 1-24 service <i>mgcp</i>	Creates a serial D-channel interface for signaling backhaul and specifies control protocol MGCP for signaling backhaul. Note The controller time slots cannot be shared between backhaul and other Layer 3 protocols.
Step 3	Router(config-control)# exit	Exits from controller configuration mode.

	Command	Purpose
Step 4	Router(config)# interface serial <i>controller-number:23</i>	Enters interface configuration mode for the D-channel signaling backhaul interface. Enter a controller number that matches the controller number specified in Step 1.
Step 5	Router(config-if)# isdn switch-type <i>switch-type</i>	Configures the D-channel interface to match the ISDN switch type. Examples of ISDN switch types include: <ul style="list-style-type: none"> • primary-4ess • primary-5ess • primary-nec5 Note This can be done in either global or interface configuration mode.
Step 6	Router(config-if)# isdn bind-L3 backhaul <i>set-name</i> Note “L” is shown here for clarity. You can enter lower-case “l”.	Configures ISDN to backhaul Q.931 to the Cisco VSC3000. You must use the set name of a session set that was defined in Step 2 of the “Creating Session Sets, Session Groups, and Sessions” section on page 64.
Step 7	Router(config-if)# exit	Exits the interface configuration mode.

Repeat this procedure for each T1 interface on the media gateway that will use backhaul.

Configuring Fast Ethernet for Signaling Backhaul Compatibility

If your media gateway has 10/100 BASE-T Fast Ethernet capability, configure the Fast Ethernet interface not to use auto negotiation.



Caution

When the Fast Ethernet interface is configured for auto-negotiation, it can take up to 2 seconds for this interface to be enabled when the interface has to initialize. Two examples where the interface initializes are execution of the **no shut** command and disconnection or reconnection of the Ethernet cable. Auto-negotiation affects the traffic flow on the Ethernet interface and can, therefore, interrupt the traffic flow on existing RUDP connections, causing them to fail. To avoid these problems, the Fast Ethernet interface should not be configured for auto-negotiation. Instead, set the duplex and speed parameters according to the requirements of the network.

To reconfigure the Fast Ethernet interface for specified duplex and speed operation, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# int <i>Ethernet-port-number</i>	Enters Ethernet interface configuration mode for the specified Ethernet port.
Step 2	Router(config-if)# duplex { full half }	Configures the Ethernet port for full-duplex or half-duplex operation.

	Command	Purpose
Step 3	<code>Router(config-if)# speed {10 100}</code>	Configures the Ethernet port to operate at 10 Mbps or 100 Mbps.
Step 4	<code>Router(config-if)# exit</code>	Exits from interface configuration mode.

Configuring the Cisco VSC3000

The Cisco VSC3000 is the signaling controller software that provides call control and runs on a UNIX server such as a Sun Netra 1800. Man Machine Language (MML) is the user interface into the signaling controller software. You use this interface to configure parameters of your signaling controller software and to display information about the current settings.

To configure the Cisco VSC3000 to perform signaling backhaul, log on to the UNIX server and complete the MGCP service provisioning procedure as follows:

	Command	Purpose
Step 1	<code>mm1> prov-add:extnode:name=media-gateway-name,</code>	Assigns a name to the media gateway (the external node) at the far end of a backhaul link.
	<code>desc=media-gateway-name</code>	Provides a description of the media gateway (MG).
Step 2	<code>mm1> prov-add:ipfaspath:name=ipfaspath-name,</code>	Adds an IP path for D-channel transport (ipfaspath) from the Cisco VSC3000 to a media gateway and assigns it a path name.
	<code>extnode=media-gateway-name,</code>	Specifies the media gateway (external node) at the opposite end of the IP path; the name must match the media gateway name assigned in Step 1.
	<code>mdo=ISDN-variant,</code>	Specifies the ISDN variant. Options include: <ul style="list-style-type: none"> • ETSI_300_102 • ETSI_300_102_C1 • ATT_41459 • ATT_41459_C2 • BELL_1268 • ETSI_300_172 • BELL_1268_C3
	<code>custgrpID=customer-group-ID,</code>	Assigns a customer group ID (the dial plan to use for this connection).
	<code>side=equipment-location,</code>	Defines the Cisco VSC3000 as network side or user side. The Cisco VSC3000 is normally network side, opposite to the PBX, which is normally the user side. Enter network , or user .
	<code>desc=description</code>	Describes the function of this IP path (backhaul service to a specified media gateway, for example Backhaul service to 3660-6).
	Step 3	<code>mm1> prov-add:iplnk:name=iplink-name,</code>
<code>if=enifinterface-number,</code>		The Ethernet interface name for the Cisco VSC3000 Ethernet card (typically enif1).
<code>ipaddr=IP_Addrnumber,</code>		The IP address of the Cisco VSC3000 Ethernet port as defined in ../etc/XECfgParm.dat (for example, IP_Addr1).
<code>port=port-number,</code>		The port number on the Cisco VSC3000.
<code>pri=priority-number,</code>		The selection priority of this IP link. (1, 2 and so on; this should match the selection priority specified on the media gateway for this IP link.)
<code>peeraddr=IP-address,</code>		The IP address of the media gateway.
<code>peerport=port-number,</code>		The port number on the media gateway; does not have to match the Cisco VSC3000 port

	Command	Purpose
	<code>sigslot=slot-number,</code>	The physical card slot in the media gateway.
	<code>sigport=port-number,</code>	The PRI port number in the media gateway (= the T1/E1 controller number).
	<code>svc=ipfspath-name,</code>	The IP path that this IP link is assigned to, which matches the <i>ipfspath-name</i> assigned in Step 2.
	<code>desc=description</code>	Optional description of this IP link. For example, <code>IP link-backhaul svc 3660-6</code> could describe an IP link for backhaul service to media gateway 3660-6.
Step 4	<code>mml>prov-add:mgcppath:name=MGCP-path-name,</code>	Defines an MGCP control path. For example, <code>mgcp36606</code> could define an MGCP path to media gateway 3660-6.
	<code>extnode=ipfspath-name,</code>	Associates the MGCP control path with an IP path for D-channel transport. The <i>ipfspath-name</i> must match the <i>ipfspath-name</i> specified in Step 2.
	<code>desc=description</code>	Optional description of this MGCP control path. For example, <code>MGCP service to 3660-6</code> could describe the function of this MGCP control path.
Step 5	<code>mml>prov-add:iplnk:name=clink6,</code>	Adds an IP link for the MGCP control path.
	<code>if=enif1,</code>	The Ethernet interface name for the Cisco VSC3000 Ethernet card (typically <code>enif1</code>).
	<code>ipaddr=IP_Addrnumber,</code>	The IP address of the Cisco VSC3000 Ethernet port as defined in <code>./etc/XECfgParm.dat</code> (for example, <code>IP_Addr1</code>).
	<code>port=2427,</code>	The port used by the IP link for the MGCP control path on the Cisco VSC3000 (2427 is pre-defined for MGCP use).
	<code>peeraddr=IP-address,</code>	The IP address of the media gateway connected to this IP link.
	<code>peerport=2427,</code>	The IP port at the media gateway for this IP link (2427 is pre-defined for MGCP use).
	<code>svc=mgcp-service-name,</code>	A name of the MGCP signaling service supported by this IP link. For example, <code>mgcp36606</code> could be the name for MGCP signaling service to 3660-6).
	<code>pri=1,</code>	Selection priority for this IP link (1, 2, and so on).
	<code>desc=description</code>	Optional description of the IP link for the MGCP control path. For example, <code>MGCP link to 3660-6</code> could describe the IP link for the MGCP path to 3660-6.

**Note**

If the Cisco VSC3000 is set up for fault-tolerant operation, configure the backhaul session manager also for fault-tolerant operation. For more information, refer to the *Cisco MGC Software Release 7 Provisioning Guide*.

Verifying Configuration

- Step 1** Enter the **show isdn status** command to verify successful ISDN configuration for backhaul. The following output shows that Layers 1, 2, 3 are enabled and active. Layer 3 shows the number of active ISDN calls.

In the example below, notice that the Layer 2 protocol is Q.921, and the Layer 3 protocol is BACKHAUL. This verifies that it is configured to backhaul ISDN. Also, if you are connected to a live line, you should see that Layer 1 status is ACTIVE and that layer 2 state is MULTIPLE_FRAME_ESTABLISHED. This means that the ISDN line is up and active.

```
Router# show isdn status

*00:03:34.423 UTC Sat Jan 1 2000
Global ISDN Switchtype = primary-net5
ISDN Serial1:23 interface
    dsl 0, interface ISDN Switchtype = primary-net5
    L2 Protocol = Q.921 L3 Protocol(s) = BACKHAUL
Layer 1 Status:
ACTIVE
Layer 2 Status:
TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
Layer 3 Status:
NLCB:callid=0x0, callref=0x0, state=31, ces=0 event=0x0
NLCB:callid=0x0, callref=0x0, state=0, ces=1 event=0x0
0 Active Layer 3 Call(s)
Activated dsl 0 CCBs = 0
Number of active calls = 0
Number of available B-channels = 23
Total Allocated ISDN CCBs = 0
Router#
```

- Step 2** Enter the **show backhaul-session-manager set all** command to display all session sets. This set contains one group called grp1, and it is configured in fault-tolerant mode.

```
Router# show backhaul-session-manager set all
Session-Set
Name      :set1
State     :BSM_SET_OOS
Mode      :Fault-Tolerant(FT)
Option    :Option-Client
Groups    :1
statistics
Successful switchovers:0
Switchover Failures:0
Set Down Count 0
Group:grp1
```

Possible states are:

SESS_SET_IDLE—A session set has been created.

SESS_SET_OOS—A session has been added to session group. No ACTIVE notification has been received from the Cisco VSC3000.

SESS_SET_ACTIVE_IS—An ACTIVE notification has been received over one in-service session group. STANDBY notification has not been received on any available session group(s).

SESS_SET_STNDBY_IS—A STANDBY notification is received, but no in-service active session group available.

SESS_SET_FULL_IS—A session group in-service that has ACTIVE notification, and at least one session group in-service has STANDBY notification.

SESS_SET_SWITCH_OVER—An ACTIVE notification is received on session group in-service, that had received STANDBY notification.

- Step 3** Enter the **show backhaul-session-manager group status all** command to display the status of all session-groups.

The status is either Group-OutOfService (no session in the group has been established) or Group-Inservice (at least one session in the group has been established).

The Status (use) is either Group-Standby (the Cisco VSC3000 connected to the other end of this group will go into standby mode), Group-Active (the Cisco VSC3000 connected to the other end of this group will be the active Cisco VSC3000), or Group-None (the Cisco VSC3000 has not declared its intent yet).

```
Router# show backhaul-session-manager group status all
Session-Group
Group Name      :grp1
  Set Name       :set1
  Status          :Group-OutOfService
  Status (use)   :Group-None
```

- Step 4** Enter the **show backhaul-session-manager session all** command to display all sessions.

The State is OPEN (the connection is established), OPEN_WAIT (the connection is awaiting establishment), OPEN_XFER (session failover is in progress for this session, which is a transient state), or CLOSE (this session is down, also a transient state). The session moves to OPEN_WAIT after waiting a fixed amount of time.

The Use-status field indicates whether PRI signaling traffic is being transported over this session. The field will be either OOS (this session is not being used to transport signaling traffic) or IS (this session is being used currently to transport all PRI signaling traffic). The User-status field indicates the connection status.

```
Router# show backhaul-session-manager session all

Session information --
Session-id:35
  Group:grp1
Configuration:
  Local:10.1.2.15      , port:8303
  Remote:10.5.0.3     , port:8303
  Priority:2
  RUDP Option:Client, Conn Id:0x2
State:
  Status:OPEN_WAIT, Use-status:OOS
Statistics:
  # of resets:0
  # of auto_resets 0
  # of unexpected RUDP transitions (total) 0
  # of unexpected RUDP transitions (since last reset) 0
  Receive pkts - Total:0 , Since Last Reset:0
  Recieve failures - Total:0 ,Since Last Reset:0
  Transmit pkts - Total:0, Since Last Reset:0

  Transmit Failures (PDU Only)
    Due to Blocking (Not an Error) - Total:0, Since Last Reset:0
    Due to causes other than Blocking - Total:0, Since Last
Reset:0
  Transmit Failures (NON-PDU Only)
    Due to Blocking(Not an Error) - Total:0, Since Last Reset:0
    Due to causes other than Blocking - Total:0, Since Last
```



```

Reset:0
  RUDP statistics
    Open failures:0
    Not ready failures:0
    Conn Not Open failures:0
    Send window full failures:0
    Resource unavailble failures:0
    Enqueue failures:0

```

Configuring MGCP POTS Dial Peer

To configure MGCP POTS dial peer on the Cisco 2600 series and Cisco 3660, complete the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# dial-peer voice <i>number</i> pots	Enters the dial-peer configuration mode for the POTS dial peer.
Step 2	Router(config-dial-peer)# application MGCPAPP	Initiates MGCP for the voice ports.
Step 3	Router(config-dial-peer)# port <i>slot/port:ds0-group</i>	Associates the dial peer with a specific logical interface. The value of <i>slot</i> is the router location where the voice port adapter is installed. Valid entries are from 0 to 3. The value of <i>port</i> indicates the voice interface card location. Valid entries are 0 or 1. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
Step 4	Router(config-dial-peer)# exit	Exits dial-peer configuration mode.

Monitoring and Maintaining

Monitoring MGCP CAS PBX and AAL2 PVC Configurations

Use these commands at any time to monitor the MGCP configuration:

Command	Purpose
Router# show mgcp [connection endpoint statistics]	Displays all active MGCP connections on the router.
Router# debug mgcp [all errors events packets parser]	Turns on debugging for the gateway.
Router# clear mgcp statistics	Resets the MGCP statistical counters.

Monitoring and Maintaining Signaling Backhaul

Use the following commands as required to monitor and maintain the signaling backhaul sessions and the connection to the Cisco VSC3000:

Command	Purpose
Router# <code>clear backhaul-session-manager group</code>	Resets the statistics for all available session groups or a specified session group.
Router# <code>show backhaul-session-manager set</code>	Displays status, statistics, or configuration of all available session sets.
Router# <code>show backhaul-session-manager group</code>	Displays status, statistics, or configuration of all available session groups.
Router# <code>show backhaul-session-manager session</code>	Displays status, statistics, or configuration of all available sessions.
Router# <code>show isdn status</code>	Displays status of ISDN backhaul. If the connection to the Cisco VSC3000 is lost, the router shuts down Layer 2 so that it cannot receive more calls. When the Cisco VSC3000 connection is back up, you may use this to verify that Layer 2 was also brought back up correctly.

Configuration Examples

This section provides the following configuration examples:

- [Cisco 2600 Series, page 74](#)
 - [MGCP CAS Voice/FAX Call Examples, page 74](#)
 - [MGCP PRI Backhaul Configuration Examples, page 78](#)
- [Cisco 3660, page 82](#)
 - [MGCP CAS Call Examples, page 82](#)
 - [VoATM with AAL2 Trunking CAS Call Examples, page 84](#)
 - [VoATM with AAL2 Trunking CCS Call Examples, page 86](#)
 - [PRI/Q.931 Signaling Backhaul Examples, page 88](#)
 - [PRI/Q.931 Signaling Backhaul CAS Call Examples, page 91](#)

Cisco 2600 Series

MGCP CAS Voice/FAX Call Examples

Originating Gateway Configuration Example

```
2650-org# show run
Building configuration...

Current configuration:
```

```

!
version 12.1
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 2650-org
!
enable password lab
!
!
!
memory-size iomem 10
voice-card 1
no ip subnet-zero
no ip domain-lookup
ip dhcp smart-relay
!
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
mgcp modem passthrough voaal2 mode nse
mgcp sdp simple
mgcp default-package dt-package
no mgcp timer receive-rtcp
!
!
controller T1 0/0
mode atm
framing esf
clock source internal
linecode b8zs
!
controller T1 0/1
!
controller T1 1/0
framing esf
linecode b8zs
ds0-group 1 timeslots 1 type e&m-immediate-start
ds0-group 2 timeslots 2 type e&m-immediate-start
!
controller T1 1/1
!
interface Ethernet0/0
ip address 1.9.46.170 255.255.0.0
no cdp enable
!
interface Ethernet0/1
no ip address
shutdown
no cdp enable
!
interface ATM0/0
no ip address
ip mroute-cache
atm idle-cell-format itu
atm uni-version 4.0
atm voice aal2 aggregate-svc bandwidth 1500
atm voice aal2 aggregate-svc traffic-parameters 1500 1500 65000
atm voice aal2 aggregate-svc upspeed-number 100
atm voice aesa 111111111111.01 application aal2xgcp
atm ilmi-keepalive
pvc 0/5 qsaal
!
pvc 0/16 ilmi

```

```

!
!
ip default-gateway 1.9.0.1
ip kerberos source-interface any
ip classless
no ip http server
!
no cdp run
!
snmp-server engineID local 000000090200005054747B80
no snmp-server ifindex persist
snmp-server forwarder
snmp-server manager
!
voice-port 1/0:1
!
voice-port 1/0:2
!
dial-peer cor custom
!
!
!
dial-peer voice 1 pots
  application mgcpapp
  port 1/0:1
  forward-digits all
!
dial-peer voice 2 pots
  application mgcpapp
  port 1/0:2
  forward-digits all
!
!
line con 0
  exec-timeout 0 0
  transport input none
line aux 0
line vty 0 4
  password lab
  login
!
end
2650-org#

```

Terminating Gateway Configuration Example

```

2650-trm# show run

Building configuration...

Current configuration:
!
version 12.1
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 2650-trm
!
boot system flash

```

```

!
!
!
memory-size iomem 20
voice-card 1
  ip subnet-zero
  no ip domain-lookup
  ip host dirt 223.255.254.254
  ip dhcp smart-relay
!
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
mgcp modem passthrough voaal2 mode nse
mgcp sdp simple
mgcp default-package dt-package
no mgcp timer receive-rtcp
!
!
controller T1 0/0
mode atm
  framing esf
  clock source internal
  linecode b8zs
!
controller T1 1/0
  framing esf
  linecode b8zs
  ds0-group 1 timeslots 1 type e&m-immediate-start
  ds0-group 2 timeslots 2 type e&m-immediate-start
  ds0 busyout 24
!
!
!
!
interface FastEthernet0/0
  ip address 1.9.46.150 255.255.0.0
  duplex auto
  speed auto
!
interface FastEthernet0/1
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface ATM0/0
  no ip address
  ip mroute-cache
  atm idle-cell-format itu
  atm uni-version 4.0
  atm voice aal2 aggregate-svc bandwidth 1500
  atm voice aal2 aggregate-svc traffic-parameters 1500 1500 65000
  atm voice aal2 aggregate-svc upspeed-number 100
  atm voice aesa 222222222222.01 application aal2xgcp
  atm ilmi-keepalive
  pvc 0/5 qsaal
  !
  pvc 0/16 ilmi
  !
!
ip default-gateway 1.9.0.1
ip kerberos source-interface any
ip classless
ip route 0.0.0.0 0.0.0.0 1.9.0.1

```

```

no ip http server
!
!
snmp-server engineID local 0000000902000024B1345A0
no snmp-server ifindex persist
snmp-server forwarder
snmp-server manager
!
voice-port 1/0:1
!
voice-port 1/0:2
!
dial-peer cor custom
!
!
!
dial-peer voice 1 pots
  application mgcpapp
  port 1/0:1
  forward-digits all
!
dial-peer voice 2 pots
  application mgcpapp
  port 1/0:2
  forward-digits all
!
!
line con 0
  exec-timeout 0 0
  transport input none
line aux 0
line vty 0 4
  login
!
end
2650-trm#

```

MGCP PRI Backhaul Configuration Examples

Originating Gateway Configuration Example

```

2650-org# show run

Building configuration...

Current configuration:
!
version 12.1
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 2650-org
!
enable password lab
!
!
!
memory-size iomem 10
voice-card 1
no ip subnet-zero

```

```

no ip domain-lookup
ip dhcp smart-relay
!
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
mgcp modem passthrough voaal2 mode nse
mgcp sdp simple
mgcp default-package dt-package
no mgcp timer receive-rtcp
backhaul-session-manager
  set vsc2_set client nft
  group vsc2_grp set vsc2_set
  session group vsc2_grp 1.9.64.23 8004 1.9.46.170 8004 1
isdn switch-type primary-5ess
call rsvp-sync
!
!
!
controller T1 0/0
mode atm
framing esf
clock source internal
linecode b8zs
!
controller T1 0/1
!
controller T1 1/0
framing esf
linecode b8zs
pri-group timeslots 1-24 service mgcp
!
controller T1 1/1
!
!
!
!
interface Ethernet0/0
ip address 1.9.46.170 255.255.0.0
no cdp enable
!
interface Ethernet0/1
no ip address
shutdown
no cdp enable
!
interface ATM0/0
no ip address
ip mroute-cache
atm idle-cell-format itu
atm uni-version 4.0
atm voice aal2 aggregate-svc bandwidth 1500
atm voice aal2 aggregate-svc traffic-parameters 1500 1500 65000
atm voice aal2 aggregate-svc upspeed-number 100
atm voice aesa 111111111111.01 application aal2xgcp
atm ilmi-keepalive
pvc 0/5 qsaal
!
pvc 0/16 ilmi
!
!
interface Serial1/0:23
no ip address
ip mroute-cache
no logging event link-status

```

```

    isdn switch-type primary-5ess
    isdn incoming-voice voice
    isdn bind-13 backhaul vsc2_set
    no cdp enable
    !
    ip default-gateway 1.9.0.1
    ip kerberos source-interface any
    ip classless
    no ip http server
    !
    no cdp run
    !
    snmp-server engineID local 000000090200005054747B80
    no snmp-server ifindex persist
    snmp-server forwarder
    snmp-server manager
    !
    voice-port 1/0:23
    !
    dial-peer cor custom
    !
    !
    !
    !
    line con 0
      exec-timeout 0 0
      transport input none
    line aux 0
    line vty 0 4
      password lab
      login
    !
    no scheduler allocate
    end

2650-org#

```

Terminating Gateway Configuration Example

```

2650-trm# show run

Building configuration...

Current configuration:
!
version 12.1
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 2650-trm
!
boot system flash
!
!
!
memory-size iomem 20
voice-card 1
  ip subnet-zero
  no ip domain-lookup
  ip host dirt 223.255.254.254
  ip dhcp smart-relay

```



```

!
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
mgcp modem passthrough voaal2 mode nse
mgcp sdp simple
mgcp default-package dt-package
no mgcp timer receive-rtcp
backhaul-session-manager
  set vscl_set client nft
  group vscl_grp set vscl_set
  session group vscl_grp 1.9.64.23 8000 1.9.46.150 8000 1
isdn switch-type primary-5ess
call rsvp-sync
srcp 5555
!
!
!
!
controller T1 0/0
mode atm
framing esf
clock source internal
linecode b8zs
!
controller T1 1/0
framing esf
linecode b8zs
pri-group timeslots 1-24 service mgcp
!
!
!
!
!
interface FastEthernet0/0
ip address 1.9.46.150 255.255.0.0
duplex auto
speed auto
!
interface FastEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
interface ATM0/0
no ip address
ip mroute-cache
atm idle-cell-format itu
atm uni-version 4.0
atm voice aal2 aggregate-svc bandwidth 1500
atm voice aal2 aggregate-svc traffic-parameters 1500 1500 65000
atm voice aal2 aggregate-svc upspeed-number 100
atm voice aesa 222222222222.01 application aal2xgcp
atm ilmi-keepalive
pvc 0/5 qsaal
!
pvc 0/16 ilmi
!
!
interface Serial1/0:23
no ip address
ip mroute-cache
no logging event link-status
isdn switch-type primary-5ess

```

```

    isdn incoming-voice voice
    isdn bind-l3 backhaul vsc1_set
    no cdp enable
    !
    ip default-gateway 1.9.0.1
    ip kerberos source-interface any
    ip classless
    ip route 0.0.0.0 0.0.0.0 1.9.0.1
    no ip http server
    !
    !
    snmp-server engineID local 00000009020000024B1345A0
    no snmp-server ifindex persist
    snmp-server forwarder
    snmp-server manager
    !
    voice-port 1/0:23
    !
    dial-peer cor custom
    !
    !
    !
    line con 0
      exec-timeout 0 0
      transport input none
    line aux 0
    line vty 0 4
      login
    !
    no scheduler allocate
    end

2650-trm#

```

Cisco 3660

MGCP CAS Call Examples

Originating Gateway Configuration Example

```

3660-org# show run

!
controller T1 3/0
  framing esf
  clock source internal
  linecode b8zs
  ds0-group 1 timeslots 1 type e&m-immediate-start
!
interface ATM2/0
  ima-group 0
!
interface ATM2/1
  ima-group 0
!
interface ATM2/2
  ima-group 0
!
interface ATM2/3

```

```

    ima-group 0
  !
interface ATM2/IMA0
  mtu 17998
  ip address 2.2.2.2 255.255.255.0
  pvc 65/100
    protocol ip 2.2.2.1 broadcast
    encapsulation aal5snap
  !
  pvc 65/101
    vbr-rt 1400 1400 60000
    vcci 2
    encapsulation aal2
  !
  !
voice-port 3/0:1
  !
  !
dial-peer voice 1 pots
  application mgcpapp
  port 3/0:1
  !
  !
end

```

Terminating Gateway Configuration Example

```

3660-trm# show run

memory-size iomem 30
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
no mgcp timer receive-rtcp
isdn voice-call-failure 0
call rsvp-sync
!
controller T1 1/0
  framing esf
  clock source internal
  linecode b8zs
  ds0-group 1 timeslots 1 type e&m-immediate-start
!
interface ATM3/0
  ima-group 0
!
interface ATM3/1
  ima-group 0
!
interface ATM3/2
  ima-group 0
!
interface ATM3/3
  ima-group 0
!
interface ATM3/IMA0
  mtu 17998
  ip address 2.2.2.1 255.255.255.0
  pvc 65/100
    protocol ip 2.2.2.2 broadcast
    encapsulation aal5snap
  !
  pvc 65/101
    vbr-rt 1400 1400 60000

```

```

    vcci 2
    encapsulation aal2
    !
    !
    voice-port 1/0:1
    !
    dial-peer voice 1 pots
    application mgcpapp
    port 1/0:1
    !
    end

```

VoATM with AAL2 Trunking CAS Call Examples

Originating Gateway Configuration Example

```

3660-org# show run

!
controller T1 3/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1 type e&m-immediate-start
 ds0-group 2 timeslots 2 type e&m-immediate-start
 !
interface ATM2/0
 ima-group 0
 !
interface ATM2/1
 ima-group 0
 no scrambling-payload
 !
interface ATM2/2
 ima-group 0
 !
interface ATM2/3
 ima-group 0
 !
interface ATM2/IMA0
 mtu 17998
 ip address 2.2.2.2 255.255.255.0
 pvc 65/100
 protocol ip 2.2.2.1 broadcast
 encapsulation aal5snap
 !
 pvc 65/101
 vbr-rt 1400 1400 60000
 vcci 2
 encapsulation aal2
 !
 !
voice-port 3/0:1
 connection trunk 7200000
 !
voice-port 3/0:2
 connection trunk 7200002
 !
dial-peer voice 20 pots
 destination-pattern 7100000
 port 3/0:1

```

```

!
dial-peer voice 21 voatm
 destination-pattern 7200000
 session protocol aal2-trunk
 session target ATM2/IMA0 pvc 65/101 101
 signal-type trans
 codec aal2-profile ITUT 1 g711ulaw
 no vad
!
dial-peer voice 22 pots
 destination-pattern 7100002
 port 3/0:2
!
dial-peer voice 2003 voatm
 destination-pattern 7200002
 session protocol aal2-trunk
 session target ATM2/IMA0 pvc 65/101 102
 signal-type trans
 codec aal2-profile ITUT 1 g711ulaw
 no vad
!
!
end

```

Terminating Gateway Configuration Example

```

3660-trm# show run

memory-size iomem 30
!
controller T1 1/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1 type e&m-immediate-start
 ds0-group 2 timeslots 2 type e&m-immediate-start
!

interface ATM3/0
 ima-group 0
!
interface ATM3/1
 ima-group 0
!
interface ATM3/2
 ima-group 0
!
interface ATM3/3
 ima-group 0
!
interface ATM3/IMA0
 mtu 17998
 ip address 2.2.2.1 255.255.255.0
 pvc 65/100
 protocol ip 2.2.2.2 broadcast
 encapsulation aal5snap
!
pvc 65/101
 vbr-rt 1400 1400 60000
 vcci 2
 encapsulation aal2
!
!

```

```

voice-port 1/0:1
  connection trunk 7200000
!
voice-port 1/0:2
  connection trunk 7200002
!
dial-peer voice 20 pots
  destination-pattern 7100000
  port 1/0:1
!
dial-peer voice 21 voatm
  destination-pattern 7200000
  session protocol aal2-trunk
  session target ATM3/IMA0 pvc 65/101 101
  signal-type trans
  codec aal2-profile ITUT 1 g711ulaw
  no vad
!
dial-peer voice 22 pots
  destination-pattern 7100002
  port 1/0:2
!
dial-peer voice 2002 voatm
  destination-pattern 7200002
  session protocol aal2-trunk
  session target ATM3/IMA0 pvc 65/101 102
  signal-type trans
  codec aal2-profile ITUT 1 g711ulaw
  no vad
!
end

```

VoATM with AAL2 Trunking CCS Call Examples

Originating Gateway Configuration Example

```

3660-org# show run

!
controller T1 3/0
  mode ccs frame-forwarding
  framing esf
  clock source internal
  linecode b8zs
  channel-group 23 timeslots 24 speed 64
  ds0-group 0 timeslots 1 type ext-sig
!
interface ATM2/1
  ima-group 0
!
interface ATM2/2
  ima-group 0
!
interface ATM2/3
  ima-group 0
!
interface ATM2/4
  ima-group 0
!
interface ATM2/IMA0
  mtu 17998

```

```

ip address 2.2.2.1 255.255.255.0
no atm ilmi-keepalive
atm voice aal2 aggregate-svc upspeed-number 0
pvc 65/100
  protocol ip 2.2.2.2 broadcast
  encapsulation aal5snap
!
pvc 65/101
  vbr-rt 500 500 500
  encapsulation aal2
!
pvc 65/102
  vbr-rt 500 500 500
  encapsulation aal5mux voice
!
!
interface Serial3/0:23
  no ip address
  no keepalive
  ccs encap atm
  ccs connect ATM2/IMA0 pvc 65/102
!
voice-port 3/0:0
  connection trunk 2000
!
dial-peer cor custom
!
!
!
dial-peer voice 1000 pots
  destination-pattern 1000
  port 3/0:0
!
dial-peer voice 2000 voatm
  destination-pattern 2000
  called-number 1000
  session protocol aal2-trunk
  session target ATM2/IMA0 pvc 65/101 100
  signal-type ext-signal
  codec aal2-profile ITUT 1 g711ulaw
  no vad
!
!
end

```

Terminating Gateway Configuration Example

```

3660-trm# show run

!
controller T1 1/0
mode ccs frame-forwarding
framing esf
clock source internal
linecode b8zs
channel-group 23 timeslots 24 speed 64
ds0-group 0 timeslots 1 type ext-sig
!
interface Serial1/0:23
  no ip address
  no keepalive
  ccs encap atm
  ccs connect ATM3/IMA0 pvc 65/102

```

```

!
interface ATM3/0
  ima-group 0
!
interface ATM3/IMA0
  mtu 17998
  ip address 2.2.2.2 255.255.255.0
  no atm ilmi-keepalive
  atm voice aal2 aggregate-svc upspeed-number 0
  pvc 65/100
    protocol ip 2.2.2.1 broadcast
    encapsulation aal5snap
  !
  pvc 65/101
    vbr-rt 500 500 500
    encapsulation aal2
  !
  pvc 65/102
    vbr-rt 500 500 500
    encapsulation aal5mux voice
  !
!
!
voice-port 1/0:0
  connection trunk 2000
!
dial-peer voice 1000 pots
  destination-pattern 1000
  port 1/0:0
!
dial-peer voice 2000 voatm
  destination-pattern 2000
  called-number 1000
  session protocol aal2-trunk
  session target ATM3/IMA0 pvc 65/101 100
  signal-type ext-signal
  codec aal2-profile ITUT 1 g711ulaw
  no vad
!
!
end

```

PRI/Q.931 Signaling Backhaul Examples

Originating Gateway Configuration Example

```

3660-org# show run

memory-size iomem 30
!
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
mgcp modem passthrough voaal2 mode nse
mgcp sdp simple
mgcp default-package dt-package
no mgcp timer receive-rtcp
backhaul-session-manager
  set vscl_set client nft
  group vscl_grp set vscl_set
  session group vscl_grp 1.9.64.23 8004 1.9.47.55 8004 1
isdn switch-type primary-5ess
isdn voice-call-failure 0

```



```

call rsvp-sync
!
voice class codec 1
!
!
!
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24 service mgcp
!
interface Serial1/0:23
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 isdn bind-13 backhaul vsc1_set
 no cdp enable
!
interface ATM2/0
 ima-group 0
!
interface ATM2/1
 ima-group 0
!
interface ATM2/2
 ima-group 0
!
interface ATM2/3
 ima-group 0
!
interface ATM2/IMA0
 mtu 17998
 ip address 2.2.2.2 255.255.255.0
 no atm ilmi-keepalive
 atm voice aal2 aggregate-svc bandwidth 1536
 atm voice aal2 aggregate-svc traffic-parameters 1536 1536 65536
 atm voice aal2 aggregate-svc upspeed-number 100
 atm voice aesa AAAAAAAAAA.01 application aal2xgcp
 pvc 0/5 qsaal
!
 pvc 0/16 ilmi
!
 pvc 65/100
 protocol ip 2.2.2.1 broadcast
 encapsulation aal5snap
!
 pvc 65/101
 vbr-rt 1400 1400 60000
 vcci 2
 encapsulation aal2
!
!
voice-port 1/0:23
!
dial-peer voice 1 pots
 application mgcpapp
 forward-digits all
!
!
gatekeeper
 shutdown
!

```

```

!
line con 0
  exec-timeout 0 0
  privilege level 15
  transport input none
line aux 0
line vty 0 4
  login
!
scheduler allocate 5000 3000
end

```

Terminating Gateway Configuration Example

```

3660-trm# show run

!
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
mgcp modem passthrough voaal2 mode nse
mgcp sdp simple
mgcp default-package dt-package
no mgcp timer receive-rtcp
backhaul-session-manager
  set vscl_set client nft
  group vscl_grp set vscl_set
  session group vscl_grp 1.9.64.23 8000 1.9.48.41 8000 1
isdn switch-type primary-5ess
isdn voice-call-failure 0
call rsvp-sync
!
!
!
!
!
!
!
controller T1 1/0
  framing esf
  clock source internal
  linecode b8zs
  pri-group timeslots 1-24 service mgcp
!
interface Serial1/0:23
  isdn switch-type primary-5ess
  isdn incoming-voice voice
  isdn bind-13 backhaul vscl_set
!
interface ATM3/0
  ima-group 0
!
interface ATM3/1
  ima-group 0
!
interface ATM3/2
  ima-group 0
!
interface ATM3/3
  ima-group 0
!
interface ATM3/IMA0

```

```

mtu 17998
ip address 2.2.2.1 255.255.255.0
no atm ilmi-keepalive
atm voice aal2 aggregate-svc bandwidth 1536
atm voice aal2 aggregate-svc traffic-parameters 1536 1536 65536
atm voice aal2 aggregate-svc upspeed-number 100
atm voice aesa 99999999999.01 application aal2xgcp
pvc 0/5 qsaal
!
pvc 0/16 ilmi
!
pvc 65/100
  protocol ip 2.2.2.2 broadcast
  encapsulation aal5snap
!
pvc 65/101
  vbr-rt 1400 1400 60000
  vcci 2
  encapsulation aal2
!
!
voice-port 1/0:23
!
dial-peer voice 1 pots
  application mgcpapp
!
!
end

```

PRI/Q.931 Signaling Backhaul CAS Call Examples

Originating Gateway Configuration Example

```

3660-org# show run

memory-size iomem 30
!
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
mgcp modem passthrough voaal2 mode nse
no mgcp timer receive-rtcp
call rsvp-sync
!
voice class codec 1
!
!
!
!
!
!
!
controller T1 3/0
  framing esf
  clock source internal
  linecode b8zs
  ds0-group 1 timeslots 1 type e&m-immediate-start
!
interface ATM2/0
  ima-group 0
!
interface ATM2/1
  ima-group 0

```

```

!
interface ATM2/2
  ima-group 0
!
interface ATM2/3
  ima-group 0
!
interface ATM2/IMA0
  mtu 17998
  ip address 2.2.2.2 255.255.255.0
  no atm ilmi-keepalive
  atm voice aal2 aggregate-svc bandwidth 1536
  atm voice aal2 aggregate-svc traffic-parameters 1536 1536 65536
  atm voice aal2 aggregate-svc upspeed-number 100
  atm voice aesa AAAAAAAAAA.01 application aal2xgcp
  pvc 0/5 qsaal
  !
  pvc 0/16 ilmi
  !
  pvc 65/100
    protocol ip 2.2.2.1 broadcast
    encapsulation aal5snap
  !
!
voice-port 3/0:1
!
dial-peer voice 1 pots
  application mgcpapp
  port 3/0:1
  forward-digits all
!
end

```

Terminating Gateway Configuration Example

```

3660-trm# show run

!
mgcp
mgcp call-agent 1.9.64.23 service-type mgcp version 0.1
mgcp modem passthrough voaal2 mode nse
no mgcp timer receive-rtcp
isdn voice-call-failure 0
call rsvp-sync
!
controller T1 1/0
  framing esf
  clock source internal
  linecode b8zs
  ds0-group 1 timeslots 1 type e&m-immediate-start
!
interface ATM3/0
  ima-group 0
!
interface ATM3/1
  ima-group 0
!
interface ATM3/2
  ima-group 0
!
interface ATM3/3
  ima-group 0
!

```

```
interface ATM3/IMA0
  mtu 17998
  ip address 2.2.2.1 255.255.255.0
  no atm ilmi-keepalive
  atm voice aal2 aggregate-svc bandwidth 1536
  atm voice aal2 aggregate-svc traffic-parameters 1536 1536 65536
  atm voice aal2 aggregate-svc upspeed-number 100
  atm voice aesa 999999999999.01 application aal2xgcp
  pvc 0/5 qsaal
  !
  pvc 0/16 ilmi
  !
  pvc 65/100
    protocol ip 2.2.2.2 broadcast
    encapsulation aal5snap
  !
  !
  voice-port 1/0:1
  !
  dial-peer voice 1 pots
    application mgcpapp
    port 1/0:1
  !
end
```

Command Reference

This section documents new or modified commands. All the commands used with this feature are documented in the following:

- Cisco IOS Release 12.2 command reference publications
- MGCP CAS PBX and AAL2 PVC documentation

For further information on related documentation, see the [“Related Documents” section on page 4](#).

New Commands

- [vbd-playout-delay maximum](#)
- [vbd-playout-delay minimum](#)
- [vbd-playout-delay mode](#)
- [vbd-playout-delay nominal](#)

Modified Commands

- [subcell-mux](#)

vbd-playout-delay maximum

To enable maximum AAL2 voice band detect playout delay buffer on a Cisco router in milliseconds, use the **vbd-playout-delay** command in voice-service configuration mode. To restore the default value, use the **no** form of this command.

vbd-playout-delay maximum *time*

no vbd-playout-delay maximum *time*

Syntax Description	<i>time</i>	The time set in milliseconds. The range is from 40-1700 milliseconds.
Defaults	The default is set to 200 milliseconds.	
Command Modes	Voice-service configuration	
Command History	Release	Modification
	12.2(8)T	This command was introduced for the Cisco 2600 series and Cisco 3660 routers.
Usage Guidelines	Use the vbd-playout-delay command to enable the maximum voice band detection playout buffer delay for fax and modem in milliseconds.	
Examples	The following example shows AAL2 CPS voice band detection range set to maximum 202 being enabled: <pre>voice service voatm session protocol aal2 vbd-playout-delay maximum 202</pre>	
Related Commands	Command	Description
	voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

vbd-playout-delay minimum

To enable maximum AAL2 voice band detect playout delay buffer on a Cisco router in milliseconds, use the **vbd-playout-delay** command in voice-service configuration mode. To restore the default value, use the **no** form of this command.

vbd-playout-delay minimum *time*

no vbd-playout-delay minimum *time*

Syntax Description	<i>time</i>	The time is set in milliseconds. The range is from 4-1700 milliseconds.
Defaults	The default is set to 4 milliseconds.	
Command Modes	Voice-service configuration	
Command History	Release	Modification
	12.2(8)T	This command was introduced for the Cisco 2600 series and Cisco 3660 routers.
Usage Guidelines	Use the vbd-playout-delay minimum command to enable the minimum voice band detection playout buffer delay for fax and modem in milliseconds.	
Examples	The following example shows AAL2 CPS voice band detection range set to minimum 6 being enabled: <pre>voice service voatm session protocol aal2 vbd-playout-delay minimum 6</pre>	
Related Commands	Command	Description
	voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

vbd-playout-delay mode

To configure voice band detection playout delay adaptation mode on a Cisco router, use the **vbd-playout-delay** command in voice-service configuration mode. To disable the voice band detection palyout delay adaptation mode, use the **no** form of this command.

vbd-playout-delay mode [**fixed** / **passthrough**]

no vbd-playout-delay mode [**fixed** / **passthrough**]

Syntax Description	fixed	The mode to set jitter buffer at constant delay in milliseconds.
	passthrough	The mode to set jitter buffer passthrough is DRAIN_FILL for clock compensation.

Defaults No default behavior or values.

Command Modes Voice-service configuration

Command History	Release	Modification
	12.2(8)T	This command was introduced for the Cisco 2600 series and Cisco 3660 routers.

Usage Guidelines Use the **vbd-playout-delay mode** command to set the playout jitter buffer in milliseconds. When a voice band is detected, the call uses G.711 codec, and the playout delay values that you set are picked up. The original voice call parameters are restored after the fax or modem call is completed.

Examples The following example shows AAL2 CPS voice band detection range set fixed mode being enabled:

```
voice service voatm
  session protocol aal2
  vbd-playout-delay mode fixed
```

Related Commands	Command	Description
	voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

vbd-playout-delay nominal

To enable nominal AAL2 voice band detect playout delay buffer on a Cisco router in milliseconds, use the **vbd-playout-delay** command in voice-service configuration mode. To restore the default value, use the **no** form of this command.

vbd-playout-delay nominal *time*

no vbd-playout-delay nominal *time*

Syntax Description	<i>time</i>	The time set in milliseconds. The range is from 0-1500 milliseconds.
---------------------------	-------------	--

Defaults	The default is set to 100 milliseconds.	
-----------------	---	--

Command Modes	Voice-service configuration	
----------------------	-----------------------------	--

Command History	Release	Modification
	12.2(8)T	This command was introduced for the Cisco 2600 series and Cisco 3660 routers.

Usage Guidelines	Use the vbd-playout-delay nominal command to enable the nominal voice band detection playout buffer delay for fax and modem in milliseconds.
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Examples	The following example shows the nominal AAL2 CPS voice band detection to 202 being enabled:
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```
voice service voatm
 session protocol aal2
 vbd-playout-delay nominal 202
```

Related Commands	Command	Description
	voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

subcell-mux

To enable subcell multiplexing on a Cisco router, use the **subcell-mux** command in voice-service configuration mode. To restore the default value, use the **no** form of the command.

subcell-mux *time*

no subcell-mux *time*

Syntax Description	<i>time</i>	This is the timer coding unit value set in milliseconds. The range is from 5 - 1000 milliseconds. The <i>time</i> argument is implemented for Cisco 3660 routers.
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Defaults	The default is set to 10 milliseconds.
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Command Modes	Voice-service configuration
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Command History	Release	Modification
	12.1(1)XA	This command was introduced for the Cisco MC3810 multiservice concentrator.
	12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
	12.2(2)XB	The <i>time</i> argument was introduced for the Cisco 3660 routers.
	12.2(8)T	This command was integrated.

Usage Guidelines	Use the subcell-mux command to enable ATM adaptation layer 2 (AAL2) common part sublayer (CPS) subcell multiplexing when the Cisco router interoperates with other equipment that uses subcell multiplexing.
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Examples	The following example shows AAL2 CPS subcell multiplexing set to 15 milliseconds being enabled:
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```
voice service voatm
 session protocol aal2
 subcell-mux 15
```

Related Commands	Command	Description
	voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

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: CS and SAR. AALs differ on the basis of the source-destination timing used, whether they use CBR or 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.

AAL2—ATM adaptation layer 2. One of four AALs recommended by the ITU-T. AAL2 is used for connection-oriented services that support a variable bit rate, such as some isochronous video and voice traffic.

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.

Backhaul—A scheme where telephony signaling is reliably transported from a gateway to a Media Gateway Controller across a packet switched network.

BGW—see Business Gateway

Business Gateway—An xGCP media gateway which is a business customer premises equipment that has connection(s) to the VoIP network as well as connection(s) to the user's telephony equipment (typically a PBX, a corporate LAN or WAN). Such gateways are used to eliminate or reduce the need for individual medium (voice, data, and so forth) connectivity.

CA—see Call Agent

Call Agent—An intelligent entity in an IP telephony network which handles call control in an MGCP model Voice over IP network.

CAS—channel associated signaling. A form of signaling that the circuit state is indicated by one or more bits of signaling status sent repetitively and associated with that specific circuit. CAS is used on a T1 line. With CAS, a signaling element is dedicated to each channel in the T1 frame. This type of signaling is sometimes called Robbed Bit Signaling (RBS) because a bit is taken out (or robbed) from the user's data stream to provide signaling information to and from the switch.

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. A signaling system used in telephone networks that separates signaling information from user data. A specified channel is exclusively designated to carry signaling information for all other channels in the system.

CID—channel identifier

CLASS—Custom Local Area Subscriber Services, usually referred to as “Custom Calling” features

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. In Voice over ATM, it specifies the voice coder rate of speech for a dial peer.

Dial peer—An addressable call endpoint. In Voice over ATM, there are two kinds of dial peers: POTS and VoATM.

DS-0—digital signal level 0. Framing specification used in transmitting digital signals over a single channel at 64-kbps on a T1 facility. A 64-K B-channel on an E1 or T1 WAN interface.

DTMF—dual tone multifrequency. A type of signaling that combines two distinct frequencies to generate a tone for each digit or character dialed. Sometimes referred to as Touchtone. This analog dial signaling uses two distinct tones to represent dialing digits.

E&M—Stands for 2-wire or 4-wire interfaces with separate signaling paths (from “Ear and Mouth”, also “receive and transmit”). E&M is a trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. Cisco’s analog E&M interface is an RJ-48 connector that allows connections to PBX trunk lines (tie lines). E&M connections are also available on E1 and T1 digital interfaces.

Fault Tolerance—The level of ability within a system to operate properly even if errors occur.

FXO—Foreign Exchange Office. An FXO interface connects to the PSTN’s central office and is the interface offered on a standard telephone. Cisco’s FXO interface is an RJ-11 connector that allows an analog connection to be directed at the PSTN’s central office. This interface is of value for off-premise extension applications.

FXS—Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. Cisco’s FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment, keysets, and PBXs.

Layer 1—This describes the Physical Layer of the OSI Reference Model defined in ITU X.200. It is responsible for the electric signal being sent and received. This can be viewed as a bit stream coming in, and going out, of the system. Scope must be considered when using this term. For example, Layer 1 on a T1 is 1.544 Mbps but Layer 1 on a DS-0 timeslot in the T1 is 64 kbps.

Layer 2—This describes the Datalink Layer of the OSI Reference Model defined in ITU X.200. It is responsible for point-to-point delivery of a PDU. Layer 2 protocols have two basic classes: reliable (meaning delivery is guaranteed or an error is reported) and unreliable (meaning delivery may not occur with no indication to the upper layers).

Layer 3—This describes the Network Layer of the OSI Reference Model defined in ITU X.200. It is responsible for the network routing and delivery of a message. Examples of Layer 3 protocols include X.25 Packet Layer Protocol and the Internet Protocol. Q.931 is not considered a Layer 3 protocol because it is not concerned with routing and delivery of a message but rather the message body itself.

Media Gateway—Equipment that connects the PSTN or a PBX with the VoIP network. It is controlled by a Call Agent via MGCP. A Media Gateway terminates facilities (trunks), packetizes the PCM stream into IP/ATM and/or forwards packets into the IP/ATM network. It performs these functions in reverse order for media streams flowing from the packet network to the PSTN.

MG—see Media Gateway.

MGC—Media Gateway Controller. A Media Gateway Controller provides call control capability to handle signaling traffic from a variety of sources. It also manages connections and resources of its Media Gateways. Can also be called a Call Agent.

MGCP—Media Gateway Control Protocol

Package—A set of parameter values that define a type of voice endpoint or connection. Examples include line-package, trunk-package, dtmf-package, and atm-package.

PBX—private branch exchange. Privately owned central switching office.

PCM—pulse code modulation. Technique of encoding analog voice into a 64-kbit data stream by sampling with eight-bit resolution at a rate of 8000 times per second.

PDU—protocol data unit. OSI term for packet.

PLAR—private line, automatic ringdown. Leased voice circuit that connects two single endpoints together. When either telephone handset is taken off-hook, the remote telephone automatically rings.

POTS—plain old telephone service. Basic telephone service supplying standard single line telephones, telephone lines, and access to the PSTN.

POTS dial peer—Dial peer connected via a traditional telephony network. POTS peers point to a particular voice port on a voice network device.

PRI—primary rate interface. ISDN interface to primary rate access. Primary rate access consists of a single 64-Kbps D channel plus 23 (T1) or 30 (E1) B channels for voice or data.

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

RSIP—ReStart In Progress. MGCP command used to indicate that a span (or collection of spans) has come into service, has gone out of service, or is about to go out of service.

SPVC—soft permanent virtual circuit. A generic term for any communications medium that is permanently provisioned at the end points, but switched in the middle. In ATM, there are two kind of SPVCs: smart permanent virtual path connections (SPVPCs) and Smart permanent virtual channel connections (SPVCC).

