

Contents

[Introduction](#)

[Prerequisites](#)

[Requirements](#)

[Components Used](#)

[Conventions](#)

[Configure](#)

[Network Diagram](#)

[Configuration](#)

[Verify](#)

[Troubleshoot](#)

[Troubleshoot on the Gateway](#)

[Troubleshoot on the Cisco ATA 186](#)

[Sample Debugs for Calls Made from the Cisco ATA 186 to the Gateway](#)

[Related Information](#)

Introduction

The Cisco Analog Telephone Adaptor (ATA) 186 is a handset-to-Ethernet adaptor that interfaces regular analog phones with IP-based Telephony networks. The Cisco ATA 186 has two voice ports that can only support legacy analog touch-tone telephones. Unlike the regular Foreign Exchange Station (FXS) ports, these cannot be interfaced with a private branch exchange (PBX), as the Cisco ATA 186 cannot send out digits on these ports. With this configuration, you can use both voice ports, with different E.164 addresses on each.

This document explains how to configure one Cisco ATA 186 to send and receive calls from one Cisco IOS[®] Gateway. The Cisco ATA 186 is a simple device which does not have call routing capability. If you have a network with more than two H.323 endpoints including the ATA itself, an H.323 gatekeeper is required to do the call routing.

Refer to [Configuring and Troubleshooting an ATA 186 with Cisco IOS Gatekeepers](#) for more information.

Prerequisites

Requirements

Ensure that you meet these requirements before you attempt this configuration:

- This document assumes that the reader is familiar with the content in the [Cisco ATA 186 Basic Configuration](#) document.
- This configuration requires the Cisco ATA 186 to be at version 2.0 or later, using the H.323 feature set.
- Ensure that there is IP connectivity between the Cisco ATA 186 and the gateway.
- Ensure that the Cisco ATA 186 is accessible through the web server method, for further

configuration.

Components Used

The information in this document is based on these software and hardware versions:

- Cisco ATA 186 with version 2.12
- Cisco 3640 with Cisco IOS Software Release 12.1(2)T

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Conventions

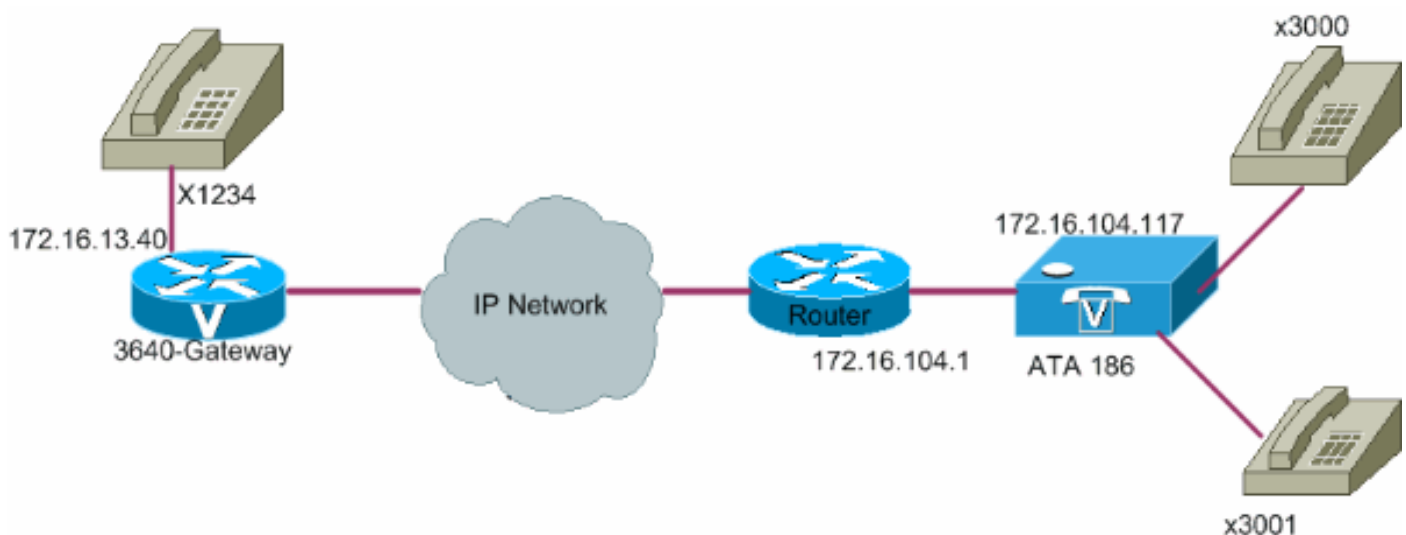
Refer to the [Cisco Technical Tips Conventions](#) for more information on document conventions.

Configure

In this section, you are presented with the information to configure the features described in this document.


Network Diagram

This document uses this network setup:



Configuration

Follow these steps:

1. Access the Cisco ATA 186 configuration window from a web browser. Go to the URL **http://ip_address_of_ata/dev**, where *ip_address_of_ata* is the IP address of the Cisco ATA 186 that you plan to register. In this example, the URL is <http://172.16.104.117/dev>. The Cisco ATA 186 configuration window appears. **Note:** The underlined fields are the relevant configured parameters for this scenario.  IP addressing can be done statically or

dynamically, as explained in the [Cisco ATA 186 Basic Configuration](#) document. In the previous figure, the static IP address is used.

2. Configure these fields in the Cisco ATA 186 configuration window: **UID0 and UID1** -

Configure the E.164 addresses of voice ports 0 and 1. Both voice ports cannot have the same E.164 address, as the Cisco ATA 186 cannot hunt if one of the ports is busy. If both voice ports are assigned the same E.164 address, the call is always sent to the first voice port. If this port is busy, the busy signal is sent to the caller. **RxCCodec and TxCodec** -

Configure the codec ID. G.723.1 - Codec ID 0 G.711a - Codec ID 1 G.711u - Codec ID 2 G.729a - Codec ID 3 In the configuration shown later in this document, the G.729r8 codec is used on the Cisco ATA 186 and on the gateway. **Note:** ATA 186 does not support two ports that use G.729a codec simultaneously. The G.729 codec can only run on one port at a time. When one port uses G.729, other port uses G.711. These combinations are allowed on the ATA 186: Two simultaneous G.723.1 codecs Two simultaneous G.711 codecs One G.723.1 and one G.711 codec One G.729A and one G.711 codec - The allocation of the G.729 resource to the FXS port is dynamic. The G.729 resource, if available, is allocated to an FXS port when a call is initiated or received. The resource is released when a call is completed. **LBRCCodec (low-bit-rate codec)** - Configure as 0 or 3, based on the chosen codec. If LBRCCodec is 0 - G.723.1 codec is available to both FXS ports at any time. Each line can maintain two G.723.1 calls in a non-conference state. Therefore, up to four G.723.1 calls can be maintained in the Cisco ATA 186. When configured for G.723, the Cisco ATA 186 supports all G.723 codecs. If LBRCCodec is 3 - G.729a is available to one of the two FXS ports on a first-come-first-served basis. The Cisco ATA 186 has only one G.729a coded. Therefore, there can be only one call at any time that uses the G.729a codec. Both ports on the Cisco ATA 186 cannot be used at the same time, if you only configure the default codec on the gateway. You need to configure a codec-class on the gateway to negotiate the second call using G.711uLaw or G.711aLaw, as G.711 is the default codec used. Without this configuration, the second call fails. When configured for G.729a, the Cisco ATA 186 supports all G.729 codecs. **NumTxFrames** - It is recommended that this field remain at the default value. It can be used when the coded byte size on the Cisco IOS gateway is changed from its default value (which is 2). This table shows the frame definition of the Cisco ATA 186: This table shows the frame sizes with the default value of 2 for NumTxFrame: **Note:** The values are equal to the frame definition of the ATA x 2, when the NumTxFrame is 2. The corresponding values can be set on the gateway using the **bytes** parameter in the **codec** command. Refer to the [codec \(dial-peer\)](#) command for more information. **Gateway** -

Configure the IP address of the gateway. Once this is done, anything that is dialed from the voice ports connected to the Cisco ATA 186 are sent to this gateway. **Note:** Use this Gateway field to define the terminating gateway with which the Cisco ATA 186 communicates for a VoIP H.323 call. If the terminating gateway is another Cisco ATA 186, then configure this gateway field with the IP address of the terminating Cisco ATA 186. If you need this Cisco ATA 186 to talk to multiple terminating devices (other Cisco ATA 186s or voice gateways), you must implement a gatekeeper in the network for E.164-to-IP address resolution for the Cisco ATA 186. Refer to [Configuring and Troubleshooting an ATA 186 with Cisco IOS Gatekeepers](#) for more information. **Note:** If port 1 of the Cisco ATA 186 needs to communicate with port 2 on the same Cisco ATA 186, then you must configure the Gateway field to its own IP address or you must configure the Cisco ATA 186 to communicate with the gatekeeper, which returns its own IP address for the call setup. **StaticIP** - IP address of the Cisco ATA. **MediaPort** - Configure the port used by the Cisco ATA 186 to send the Real-Time Transport Protocol (RTP) stream. The configured port and higher is used. This feature

is available in version 2.13 and later.

3. Click **Apply** and then reload the page. Cisco ATA 186 2.14 and later supports dial plan support for hotline and warmline functionality. **Configuration** The dial plan configuration parameter now accepts the rule `Hdnnnn`, where `d` is the post-off hook delay (in seconds - use 0-9 or a-z to specify a delay in the range of 0 to 35 seconds), and `nnnn` is the variable length phone number to call when no digit is entered for `d` seconds after the phone is taken off hook. Example 1 - H05551212 (call 5551212 immediately after the phone is taken off hook). Example 2 - H5923123456 (call 923123456 if no digit is entered for 5 seconds after the phone is taken off hook). The Cisco ATA 186 takes 10 seconds to reconfigure itself. This example shows the relevant configuration for the Cisco IOS gateway:

Verify

Use this section to confirm that your configuration works properly.

Ping the Cisco ATA 186 from the gateway. If a configuration similar to the previous example appears, the calls must go through. Ensure that there is no codec mismatch. If a codec other than G.729r8 is used, configure (erase an incoming dial peer on the gateway with) an incoming dial peer on the gateway with the codec or voice-class codec as applicable on the dial-peer 3000.

Troubleshoot

This section provides information that you can use to troubleshoot your configuration.

The [Cisco CLI Analyzer](#) ([registered](#) customers only) supports certain **show** commands. Use the Cisco CLI Analyzer to view an analysis of **show** command output.

Note: Refer to [Important Information on Debug Commands](#) before you use **debug** commands.

Troubleshoot on the Gateway

Issue the [debug voip ccapi inout](#) command to debug end-to-end VoIP calls. Issue the [debug vtsp dsp](#) command to show the digits as they are received by the voice port.

Troubleshoot on the Cisco ATA 186

When you work with Third-party gatekeepers and gateways, the troubleshooting tool on the Cisco ATA 186 is very helpful. Complete these steps to enable the Cisco ATA 186 troubleshooting tool:

1. Enter the IP address of the PC that is on the same subnet as the Cisco ATA 186 in the ATA Nprintf field.
2. The port specified after the address must be **9001**.
3. Run the **prserv.exe** program at the DOS prompt on the PC. You can download the **prserv.exe** program from the Cisco Software Center at the [ATA Software Download](#) ([registered](#) customers only) location. The **prserv.exe** program is included in the latest Cisco ATA 186 software release ZIP file.

Sample Debugs for Calls Made from the Cisco ATA 186 to the Gateway

These output examples show a successful call and an unsuccessful call, respectively:

Successful Call

D:\Documents and Settings\sshafiqu\My Documents\voice\ata>prserv.exe

logging started Wed Feb 06 18:25:27 2002

!--- Call is made from port 0, as UID0 (3000) is shown. 3000 active @0xab45555a (GK @0x3c256420)

1:00;0,0,0,0, [0]DTMF 1

[0]DTMF 2

[0]DTMF 3

[0]DTMF 4

[0]DTMF #

!--- Dual tone multifrequency (DTMF) tones for the number dialed 1234.

!--- The # symbol is not needed, as the 2.0 versions now have a 10 second timeout. Calling 1234 SCC->(0

<cmd 16> CLIP SCC->(0 0) <cmd 2> <0 0> dial<1234> block queue <- (18 1289228 0) Connect to <0xac100d28

1720>.. >>>>>> TX CALLER ID : 0x1 0x80 6

!--- Caller ID can be formatted using the CallerIDMethod field.* Q931<-0:Setup:CRV 1190 Q931->0:Proceed

Connect H245... block queue <- (19 1289228 525478) NuConnectDispatcher: 0x4a6 H245 TCP conn ac100d28 11

!--- TCP port for H.245 is 11001. CESE/MSDSE start:<0 0 0 0> capSize = 3 H245->0:Cese RemoteInputCap <1

RemoteAudioCap <4 0> RemoteAudioCap <4 11> MODE FRAME : 11 2 RemoteAudioCap <4 15> MODE FRAME : 15 2

RemoteAudioCap <4 0> Capability set accepted

!--- Capability (codec, voice activity detection [VAD]) negotiation

!--- is successful. H245->0:MSD: <rn tt> = <0x2020 60> H245->0:CeseAck H245->0:MsdAck h323.c 1837: csta

->H245<0> OLC H245<-0:LcseOpen set TX audio to G729AB/B 2 fpp

!--- The Cisco ATA 186 is prepared to communicate via the G.729ab and

!--- G.729b codecs. SetG723Mode: 2 3 H245->0:LcseOpeng H245->0:OLC mode 10 remote OpenLogicalReq

G711/G729(10) : 2 fpp

!--- The default codec for the Cisco ATA 186 is G.711. OpenRtpRxPort(0,0x0,4000):11 RTP Rx Init: 0, 0 R

>0:<0xab45555a 4000>

!--- RTP Port opened is 4000. H245->0:LcseClose : chnum 1 H245->0:LcseRelease 0: Close RTPRX H245->0:Lc

H245->0:OLC mode 14 remote OpenLogicalReq G711/G729(14) : 2 fpp OpenRtpRxPort(0,0x0,4000):12 RTP Rx Ini

0 RTP->0:<0xab45555a 4000> [0]DPKT 1st: 3725026804 3725026564, pt 18 [0]Received pi=8 in q931

Q931->0:Progress

!--- Ringback is played by the terminating gateway. H323Dispatcher : 2 3 H245->0:LcseOpenAck RTP<-

0:<0xac100d28 18712> [0]Enable encoder 18 Enable LEC adapt [0]=1 RTP TX[0]:SSRC_ID = e8c533a0 RTP Tx In

0 [0]TX SID frame (pt 18) [0]RX SID frame (pt 18) 1:30;3,0,0,0, Q931->0:Connect

!--- Call is answered. SCC:ev=12[0:0] 3 0 SCC->(0 0) <cmd 20> SCC->(0 1) <cmd 20> SCC->(0 0) <cmd 9> 30

active @0xab45555a (GK @0x3c256420) Q931->0:ReleaseComplete: reason 16, tone = 13

!--- Disconnect received from the terminating gateway. H245<-0:EndSessionCmd 1

!--- The cause is 16 (0x10), which is a normal disconnect.** 0: Close RTPRX write TCP err : 10 -33 [0:0

LBRC Res Q931<-*:ReleaseComplete write TCP err : 9 -33 SCC:ev=13[0:0] 7 1 [0:0]SCC: Disconnected

* This list defines the appropriate value per country when you configure the CallerIDMethod field:

- USA - Caller ID Method is 0x19e60.
- Sweden - Caller ID Method is 0x0ff61.
- Denmark - Caller ID Method is 0x0fde1.

Unsuccessful Calls

SCC->(0 0) <cmd 2>

<0 0> dial<258>

block queue <- (18 1289228 0)

Connect to <0xac100deb 1720>..

!--- IP connectivity issue gives TCP error. 0:30;0,0,0,0, 1:00;0,0,0,0, TCP connect err: -33 [0:0]SCC:

Disconnected

<0 0> dial<1234>

block queue <- (18 1289084 0)

Connect to <0xac100d28 1720>..


>>>>>> TX CALLER ID : 0x1 0x80 6

Q931<-0:Setup:CRV 1836

```
Q931->0:ReleaseComplete: reason 1, tone = 7
!--- The terminating gateway sends a release complete in response to the
!--- setup, with a cause code of 1, which is an unallocated or unassigned
!--- number.
!--- Note: Busy signal played by the terminating gateway.
```

```
H245<-0:EndSessionCmd 0
[0:0]Rel LBRC Res
Q931<-*:ReleaseComplete
write TCP err : 9 -33
SCC:ev=7[0:0] 3 0
SCC->(0 0) <cmd 1>
```

Related Information

- [Cisco ATA 186 Basic Configuration](#)
- [Configuring and Troubleshooting an ATA 186 with Cisco IOS Gatekeepers](#)
- [Troubleshooting and Debugging VoIP Call Basics](#)
- [Voice Technology Support](#)
- [Voice and Unified Communications Product Support](#)
- [Troubleshooting Cisco IP Telephony](#) 
- [Technical Support & Documentation - Cisco Systems](#)