Cisco ATA186 and Cisco ATA188

1 Important Notes

- Check the *SIP 3rd Party Validation Website* for current validation status. The *SIP 3rd Party Component Validation Website* can be viewed at:

  http://www.testlab.inin.com

- The difference between the ATA186 and the Cisco ATA188 is that the ATA188 has an internal Ethernet switch.
- The ATA 186/188 does not support two ports using G.729 codec simultaneously. The G.729 codec can only run on one port at a time. If one port is using G.729, the other port can only use G.711. The following combinations are allowed on the ATA 186/188:
  1. Two simultaneous G.723 codecs
  2. Two simultaneous G.711 codecs
  3. One G.723 and one G.711 codec
  4. One G.729 and one G.711 codec (G.729 is available on first-come first-serve basics)

2 Vendor Documentation

CiscoATA186 Data Sheet
http://www.cisco.com/warp/public/cc/pd/as/180/186/prodlit/at186_ds.htm
CiscoATA188 Data Sheet
http://www.cisco.com/warp/public/cc/pd/as/180/186/prodlit/at188_ds.htm
Firmware upgrades:
http://www.cisco.com/cgi-bin/tablebuild.pl/ata186
Directions for downloading firmware to the ATA186 (note that the manual method, using the program ata186us.exe, works well):
http://www.cisco.com/en/US/products/hw/gatecont/ps514/products_administration_guide_chapter09186a00800c4d1d.html#40915
Information about faxing:
http://www.cisco.com/en/US/products/hw/gatecont/ps514/products_administration_guide_chapter09186a00801e0e66.html#63198

3 Versions Verified

Supported Version: 3.2.1

4 Install

Firmware upgrades can be found at:
http://www.cisco.com/cgi-bin/tablebuild.pl/ata186

Directions for downloading firmware to the ATA186 (note that the manual method, using the program ata186us.exe, works well):
http://www.cisco.com/en/US/products/hw/gatecont/ps514/products_administration_guide_chapter09186a00800c4d1d.html#40915

The manual method is quite easy:

- The *ata186us.exe* executable is contained in the zip file that contains the firmware.
- Type *ata186us.exe* from a command prompt. It will give you all the options for the *ata186us.exe* program. Usually, the only argument you will need for the *ata186us.exe* program is the name of the firmware (which ends in .zup).
- When you run the *ata186us.exe* program with the correct arguments, it will print what command you must use on the ATA186 for it to request the firmware.

DHCP: By default, the ATA186 will have DHCP enabled. The IP address can be heard by following these steps:

- Plug an analog phone into the phone1 jack
- Go off-hook
- Press the button on the top of the ATA186 unit
- Press 21# (the system will use text-to-speech to announce the current IP address)

5 Required Post Installation Steps

6 Configuration

Line Appearances: Up to 2 analog phones can be connected to a single ATA186. Each phone is equivalent to a station in the Interaction Administrator.

Call Appearances: 1.

Using *persistent* connections allows the phone to handle more call appearances than the phone is physically capable. This is done by using the *persistent* connections and the Interaction Center Client. To manage more calls than the phone is capable (for instance an operator want to handle up to 20 simultaneous calls), check the *Persistent* checkbox in the Station configuration in Interaction Administrator. The Interaction Client can be used to manage a large number of calls while the phone will be the audio device for the calls. The phone will show one call (from the Interaction Center) while the Interaction Client will be used to manipulate the calls.

The ATA186 is configured using a web browser. All of the following parameters can be entered by typing the IP address of the ATA186 into the address window of a browser followed by /dev. For example, if the IP address is 10.0.0.1 then type http://10.0.0.1/ to access the parameters. The IP address can be determined by following the steps outlined above.

Simplest Example of a Cisco ATA 186/188 Configuration

First, do the Install section above (get the SIP firmware and load it).

Second, configure these values (*GkOrProxy, UID0,…)* below from the instructions below.

GkOrProxy: address of the Interaction Center

UID0: unique extension of the phone (i.e. 7406)

NATIP: if you are behind a firewall/router at home, configure this value as well.

SIPRegOn/SIPRegInterval: if you are behind a firewall/router at home, configure this value as well.

Third, configure a SIP station in Interaction Administrator

Station Connection SIP address: – UID0 @ [IP address of the phone]

Station Identification SIP address: – UID0 (preferred) OR *UID0* @ *GkOrProxy*

ATA 186/188 Configuration	
Parameter	Description
Proxy Section	
OutBoundProxy	Primary proxy. This is the IP address where the phone will send its requests. *Recommended Setting*: • [Most common] For managed phones not using a proxy, leave *OutBoundProxy* blank. • For managed phones and using a proxy, set *OutBoundProxy* to that proxy. The proxy will be configured to forward calls originated from the managed phone to the Interaction Center by using the *GkOrProxy* parameter. • For unmanaged phones, set *OutBoundProxy* according to your site's specifications.
GkOrProxy	The IP phone will set this address in the **host portion** of the TO and FROM headers in the SIP messages. Recommended Setting: • [Most common] For managed phones and not using a proxy, *GkOrProxy* should be set to the Interaction Center and the *OutBoundProxy* should not be set. • For managed phones and using a proxy, *GkOrProxy* should be set to the Interaction Center and the *OutBoundProxy* should be set to the proxy's address. • For unmanaged phones, set *GkOrProxy*

	according to your site's specifications.
AltGk	Alternate or Backup Proxy. This is the IP address where the phone will send its requests if no system responses to the original SIP request. This proxy will be used in all cases, i.e. if any proxy (the OutBoundProxy or the GkOrProxy) fails to respond. This value is used when one of the following are true: • Using 2 or more proxies • Using no proxies and using Interaction Center switchover • Using no proxies and using a local gateway or emergency (911) gateway The IP phone will set this address in the **host portion** of the FROM header (not the TO header) in the SIP messages.
SIPRegOn SIPRegInterval	Registration Setting *SIPRegOn* to 1 will cause a REGISTER message to be sent every *SIPRegInterval* seconds. Recommended Setting: • For managed phones, *SIPRegOn* should be set to 1. • For managed phones and not using a proxy, *SIPRegOn* should be set to 1. • For unmanaged phones, set *SIPRegOn* according to your site's specifications. If behind a firewall/router and using NAT, set this low, according to your firewall spec (usually around 180 seconds).
Identification Section	
UID0 (for first port) UID1 (for second port)	This value is sent as the **user portion** of the address in SIP FROM header for phone 0. This field should be unique and is limited to 31 characters or digits. *Interaction Center Note*: The *UID0* and *GkOrProxy* are the values configured in the station identification address in Interaction Administrator for managed phones. For example, if *UID0* is "7406" and *GkOrProxy* is "10.1.6.149", then the value of the identification address would be either "7406" (preferred) OR "sip:7406@10.1.6.149:5060". See the "*SIP Application Note"* for details. *Important*: Since *UID0* and *GkOrProxy* are used for identification with the Interaction Center, one of these fields must be unique. If using a proxy, *GkOrProxy* must be set to the Interaction Center for all the managed phones, thus *UID0* must be unique. Therefore, it is suggested that *GkOrProxy* always be unique, if you are not using a proxy.
NAT	
NATIP	NAT IP If using the Cisco ATA device behind a firewall/router at home, this is the value your ISP has given your firewall/router.

Media Section	
LBRCodec	Low-bit rate codec. The Cisco ATA 186 can support two simultaneous G.723 calls or one G.729A call. When using G.729A, the second line must use G.711 ulaw or alaw. *Values*: [default] 0 (G.723), 3 (G.729a) *Recommended Setting:* • Network specific
RxCodec	Receiving audio codec preference. *Values*: [default] 0 (G.723[1]), 1 (G.711 alaw), 2 (G.711 ulaw), 3 (G.729a[2]) *Recommended Setting*: • Network specific. 1 G.723 can be selected only if LBRCodec is set to 0 2 G.729a can be selected only if LBRCodec is set to 3
TxCodec	Transmitting audio codec preference. *Values*: [default] 0 (G.723[1]), 1 (G.711 alaw), 2 (G.711 ulaw), 3 (G.729a[2]) *Recommended Setting*: • Network specific. 1 G.723 can be selected only if LBRCodec is set to 0 2 G.729a can be selected only if LBRCodec is set to 3
AudioMode	Bitmap for the audio operating mode. The lower 16 bits apply to phone 0 and the upper 16 bits apply to phone 1. *Values*: [default] 0x00150015; enables silence suppression and DTMF negotiation via SDP *Recommended Setting:* • 0x00140014; disables silence suppression and enables DTMF negotiation via SDP. Silence suppression enabled produces very loud white noise during silence.
Supplemental Features	
CallFeatures	Call features is a 32-bit bitmap value. The lower 16-bits are used for phone 0 and the upper 16-bits are used for phone 1. *Values*: [default] 0xffffffff *Recommended Setting:* • 0xffffffff (all enabled) Bit Definitions: • bit 0 and 16 - Forward unconditional • bit 1 and 17 – Forward on Busy • bit 2 and 18 – Forward on No Answer • bit 3 and 19 – CLIR and CLIP • bit 4 and 20 – Call waiting • bit 5 and 21 – 3-way calling • bit 6 and 22 – Blind transfer • bit 7 and 23 – Consult transfer • bit 8 and 24 – Caller ID • bit 9 and 25 – Call return • bit 10 and 26 – Message waiting indication • bit 15 and 31 – FAX mode

Call Forwarding (service enabled through phone)	
Forward Unconditional	Press #72, the number you want to forward call to; then press # again
Forward When Busy	Press #74, the number you want to forward call to; then press # again
Forward On No Answer	Press #75, the number you want to forward call to; then press # again
Cancelling Call Forwarding	Press #73

<table>
<tr><th colspan="2">DialPlan</th></tr>
<tr><td>To stop the ATA from transmitting the terminating “#” in the INVITE:</td><td>Change the dial plan from the default of:

"*St4-|#St4-|911|1>#t8.r9t2-|0>#t811.rat4-|^1t4>#.-"

to

"*S>#t4-|#St4-|911|1>#t8.r9t2-|0>#t811.rat4-|^1t4>#.-"</td></tr>
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