Configure the ATA 191

Telephony Features

The following table lists the supported telephony features. Use Cisco Unified Communications Manager Administration to configure many of these features.

Table 1: Telephony Features for the ATA 191

Feature	Description	Configuration Reference
Audible Message Waiting Indicator	A stutter tone from the handset or speakerphone indicates that a user has one or more new voice messages on a line. **Note** The stutter tone is line-specific. You hear it only when using the line with the waiting messages.	For more information, refer t • *Administration Guide fo Unified Communications IM and Presence Servic Presence Service*, "Adm Overview" chapter • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter • *Feature Configuration G Unified Communication* "Audible Message Waiti chapter

Feature	Description	Configuration Reference
cBarge	Allows a user to join a nonprivate call on a shared phone line. cBarge adds a user to a call and converts it into a conference, allowing the user and other parties to access conference features. Your ATA supports Barge on a conference bridge.	For more information, refer to: • *System Configuration Guide Unified Communications M* "Configure Analog Telepho Adapters" chapter • *Feature Configuration Guide Unified Communications M* "Barge" chapter
Call forward	Allows users to redirect incoming calls to another number. Call forward options include Call Forward All, Call Forward Busy, and Call Forward No Answer.	For more information, refer to: • *System Configuration Guide Unified Communications M* "Configure Analog Telepho Adapters" chapter • *Feature Configuration Guide Unified Communications M* "Call Forwarding" chapter
Call pickup	Allows users to redirect a call that is ringing on another phone within their pickup group to their phone.	For more information, refer to: • *Feature Configuration Guide Unified Communications M* "Call Pickup" chapter
Call waiting	Indicates (and allows users to answer) an incoming call that rings while on another call. Displays incoming call information on the phone screen.	For more information, refer to: • *System Configuration Guide Unified Communications M* "Configure Analog Telepho Adapters" chapter
Caller ID	Displays caller identification such as a phone number, name, or other descriptive text on the phone screen.	For more information, refer to: • *System Configuration Guide Unified Communications M* "Configure Analog Telepho Adapters" chapter • *Administration Guide for Ci Unified Communications Ma IM and Presence Service an Presence Service,* Cisco Un Phone Configurations.

Feature	Description	Configuration Reference
Conference	• Allows a user to talk simultaneously with multiple parties by calling each participant individually. Conference features include Adhoc Conference, cBarge, and Meet-Me. • Allows a noninitiator in a standard (ad hoc) conference to add or remove participants.	For more information, refer t • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter • *Feature Configuration G Unified Communication* "Conferencing Features
Direct transfer	Allows users to connect two calls to each other (without remaining on the line).	For more information, refer t • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter • *Feature Configuration G Unified Communication* "Call Transfer" chapter
Forced authorization codes (FAC)	Controls the types of calls that certain users can place.	For more information, refer t • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter • *Feature Configuration G Unified Communication* "Speed Dial and Abbrev chapter
Group call pickup	Allows a user to answer a call that is ringing on a directory number in another group.	For more information, refer t • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter • *Feature Configuration G Unified Communication* "Call Pickup" chapter

Feature	Description	Configuration Reference
Hold/Resume	Allows the user to move a connected call between an active state and a held state. **Note** No support for resuming a call from a shared line party.	For more information, refer to: • *System Configuration Guide Unified Communications M* "Configure Analog Telepho Adapters" chapter • *Feature Configuration Guide Unified Communications M* "Secure Tone" chapter
Meet–Me conference	Allows a user to host a Meet-Me conference in which other participants call a predetermined number at a scheduled time.	For more information, refer to: • *System Configuration Guide Unified Communications M* "Configure Analog Telepho Adapters" chapter • *Feature Configuration Guide Unified Communications M* "Meet-Me Conferencing" c
Message Waiting	Defines directory numbers for message-waiting on and message-waiting off indicator. A directly connected voice-messaging system uses the specified directory number to set or to clear a message-waiting indication for a particular Cisco Unified IP Phone.	For more information refer to: • *System Configuration Guide Unified Communications M* "Configure Analog Telepho Adapters" chapter • *Feature Configuration Guide Unified Communications M* "Audible Message Waiting I chapter
Music on hold	Plays music while callers are on hold.	For more information, refer to: • *System Configuration Guide Unified Communications M* "Configure Analog Telepho Adapters" chapter • *Feature Configuration Guide Unified Communications M* "Music On Hold" chapter

Feature	Description	Configuration Reference
Privacy	Prevents users who share a line from adding themselves to a call.	For more information refer to • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter • *Feature Configuration G Unified Communication* "Privacy" chapter
Redial	Allows users to call the most recently dialed phone number by pressing the ***#** feature code.	Requires no configuration.
Shared line	Allows a user to have several devices that share the same phone number or allows a user to share a phone number with a coworker.	For more information, refer t • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter • *Feature Configuration G Unified Communication* "Manager Assistant" ch
Speed dialing	Allows users to speed dial a phone number by entering * and an assigned index code (1 to 199) on the phone keypad. Example: Press `*199` to dial the phone number with index code 199. Users assign index codes on Line configuration from the Cisco Unified Communications Manager Device window.	For more information, refer t • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter • *Feature Configuration G Unified Communication* "Speed Dial and Abbrev chapter
Time Zone Update	Updates the device with time zone changes.	For more information, refer t • *System Guide for Cisco Communications Manage* Analog Telephone Adap
Voice-messaging system	Enables callers to leave messages if calls are unanswered.	For more information refer to • *System Configuration Gu Unified Communication* "Configure Analog Tele Adapters" chapter

Product-Specific Configuration Parameters

Cisco Unified Communications Manager Administration allows you to set some product-specific configuration parameters for the ATA 191. The following table lists the configuration windows and their paths to configure the parameters.

Table 2: Configuration Information

Configuration Window	Path
Phone Configuration window	**Device** > **Phone**; Product Specific Configuration portion of window

The following table lists the configuration parameters you can set using Cisco Unified Communications Manager Administration. You can set the configuration parameters using the Phone configuration window. Options with an asterisk in the window are required.

Note Set the following ATA 191 parameters from port 1 only: IVR Password, CDP, Impedance, Input/Output Audio Level, Timers, Call Sequence, Ring1 Cadence, Ring2 Cadence, CPC Delay, CPC Duration, and MTU Size. Setting these parameters from port 2 has no effect.

Table 3: Product-Specific Configuration Parameters for the ATA 191

Parameter	Description
Line 2 Support	Enable and disable the Phone 2 port on the ATA 191. Default: Enabled
Web Access	Enable the ATA 191 to accept web connections or an HTTP client. If this option is disabled, then access to the ATA 191's internal web page is blocked. In addition, the Problem Report Tool (PRT) is disabled. Default: Disabled
HTTPS Server	Enable both HTTPS and HTTP connections to the ATA 191, or restrict connections to HTTPS only. Default: HTTPS and HTTP
Admin Password*	Set the password to access the Web Administrator interface. The password can be from 8 to 127 characters.
SSH Access	Set whether the ATA 191 accepts SSH connections. If you block SSH connections, then access to the ATA 191 is blocked. Default: Disabled
Cisco Discovery Protocol (CDP)	Enable or disable the CDP function of the ATA 191. Default: Enabled

Parameter	Description
Link Layer Discovery Protocol (LLDP)	Enable or disable LLDP on the ATA 191. Default: Enabled
LLDP Asset ID	Set the Asset ID from LLDP. The maximum length is 32.
802.1x Authentication	Enable or disable the 802.1x authentication. Default: User Controlled If the parameter is set to User Controlled, the feature is disabled on the ATA. User needs to enable it through the IVR setting on the phone that is connected to the ATA. For other values (Enabled or Disabled), the setting in CUCM takes preference.
Log Server	If using IPv4, specify an IP address and port of a remote system where log messages are sent.
IPv6 Log Server	If using IPv6, specify an IP address and port of a remote system where log messages are sent.
Remote Log	Specify where to send the log data by serviceability. If enabled, log data is copied to the location specified by the Log Server or IPv6 Log Server parameters. If disabled, log data is not copied to the log server location. Default: Disabled

Parameter	Description
Log Profile	Run the pre-defined debug command remotely: • Default—Resets the debug level to default. • Preset—Use log module settings on Phone Adapter Configuration Utility for debug flags. • Telephony—Turn on debug flag for provisioning (including auto upgrade) and call features. • SIP—Turn on debug flag for SIP messages. • UI—Turn on debug flag for key event such as DTMF, PRT, and reset button. • Network—Turn on debug flag for network events, such as DHCP, VLAN, link status change. • Media—Turn on debug flags for RTP, Fax, Tone, and SLIC-related issues. • System—Turn on debug flag for system events, such as reboot, or factory reset. • Web—Turn on debug flag for web operation and event logs. • NTP—Turn on debug flag for NTP related logs. • CDPLLDP—Turn on debug flag for CDP and LLDP logs. • Security—Turn on debug flag for security related logs.
Customer support upload URL	Provides the URL for the Problem Report Tool (PRT).
Load Server	If using IPv4, the ATA uses an alternative server to obtain firmware loads and upgrades, rather than the defined TFTP server.
IPv6 Load Server	If using IPv6, the ATA uses an alternate server to obtain firmware loads and upgrades, rather than the defined TFTP server.
Auto Barge	Auto Barge adds a user to an active call. An offhook phone automatically adds the user (initiator) to the shared line call (target), and the users currently on the call receive a tone (if configured). Barge supports conference bridges.
Echo Cancellation	Enable or Disable the use of echo canceler.

Parameter	Description
Fax Mode	The Cisco ATA 191 supports these fax modes: • Fax Pass-Through–Allows fax and modem traffic to pass through a voice port using the re-INVITE method (codec can be g711ulaw or g711alaw). • NSE Fax Pass-through g711ulaw–Allows fax traffic to pass through a voice port using the NSE method by codec g711ulaw. • NSE Fax Pass-through g711alaw–Allows fax traffic to pass through a voice port using the NSE method by codec g711alaw. • T.38 Fax Relay–Allows for a quicker protocol for fax transmission over packet networks.
Fax Error Correction Mode Override	You can set the fax error correction mode override values to one of the following settings: • Default • On • Off
FAX Disable ECAN	Set this parameter to **yes** to automatically disable Echo Canceler when FAX tone is detected.
Modem Line	If you set this parameter to **yes**, the call is treated as a modem call. The ATA191 tunes VAD, Jitter buffer, and echo canceler automatically.
Fax T38 Return To Voice	Set this parameter **yes** if voice callback is needed after the T.38 fax is completed.
Fax Tone Detect Mode	This option controls which side detects fax tone (trigger fax): • Caller Or Callee • Caller Only • Callee Only The default is Caller Or Callee.
IVR Password	ATA 191 IVR password.
Input Audio Level	Gain value of Network-to-Phone
Output Audio Level	Gain value of Phone-to-Network
Impedance	The ATA 191 provides multiple impedance values, such as 600ohm for use in the United States.

Parameter	Description
Caller Connect Polarity	Control the line polarity of the Cisco ATA FXS ports when Cisco ATA is the caller and a call is connected. Default: User forward polarity
Caller Disconnect Polarity	Control the line polarity of the Cisco ATA FXS ports when Cisco ATA is the caller and a call is disconnected. Default: User forward polarity
Callee Connect Polarity	Control the line polarity of the Cisco ATA FXS ports when Cisco ATA is the callee and a call is connected. Default: User forward polarity
Callee Disconnect Polarity	Control the line polarity of the Cisco ATA FXS ports when Cisco ATA is the callee and a call is disconnected. Default: User forward polarity
Caller ID	• BT FSK • Bellcore FSK • ETSI FSK
Call Sequence	• Bellcore FSK • ETSI FSK
Mute Progress Tone	Set this parameter to **On** to mute all progress tones on the Cisco ATA 191 during call establishment. Default setting: Off.
Ring1 Cadence	Cadence script for distinctive ring pattern. Default setting: 60(2/4).
Ring2 Cadence	Cadence script for the alternative ring pattern triggered by SIP message. Default setting: 60(.8/.4,.8/4).
CPC Delay (0-255s)	CPC(Calling Party Control) delay time in seconds after caller hangs up when the ATA 191 starts removing the tip-and-ring voltage to the attached equipment of the called party. **Note** When remote party hangs up, without CPC enabled, reorder tone will be played after a configurable delay. If CPC is enabled, dial tone will be played when tip-to-ring voltage is restored. Value range: 0–255(s). Default setting: 2(s)

Parameter	Description
CPC Duration (0-1.000s)	CPC(Calling Party Control) duration time in seconds for which the tip-to-ring voltage is removed after the caller hangs up. After that, the tip-to-ring voltage is restored and dial tone will apply if the attached equipment is still off hook. CPC is disabled if this value is set to 0. **Note** When remote party hangs up, without CPC enabled, reorder tone will be played after a configurable delay. If CPC is enabled, dial tone will be played when tip-to-ring voltage is restored. Value range: 0-1.000(s). Default setting: 0(s)
MTU Size (576-1500)	Maximum Transmission Unit (MTU) size that can be communicated in a single network layer transaction. For IPv4 only mode case, the MTU size can be set from 576 to 1500; for dual mode case, the MTU size can be set from 1281 to 1500. Value range: 576–1500 Default setting: 1500
Ring and Call Waiting Tone Specs	
Ring Waveform	Waveform for the ringing signal. Choices are Sinusoid or Trapezoid. Default setting: Trapezoid.
Ring Frequency(15-50Hz)	Frequency of the ringing signal. Value range: 15-50 (Hz). Default setting: 20.
Ring Voltage(60-90V)	Voltage of the ringing signal. Value range: 60-90 (V). Default setting: 85.
Timers	
Offhook Validation Timer (50-1000ms)	Indicates the time to validate an offhook event.
Onhook Validation Timer (50-1000ms)	Indicates the time to validate an onhook event.
Hookflash Timer (100 to 1500ms)	Indicates the time to validate a hookflash event.

Parameter	Description
Onhook Delay Timer (0 to 155ms)	Indicates the time to delay an onhook event.
Reorder Delay (0-30s)	Delay after far end hangs up before reorder tone is played.
RTP Packet Time (10-90ms)	Packet size in milliseconds for RTP. Default setting: 20.

You can access the ATA 191 web page and perform limited configuration. In Admin mode, most information and settings are available.

Add Users to Cisco Unified Communications Manager

Adding users to Cisco Unified Communications Manager allows you to display and maintain information about users. Each added user can perform these tasks:

- Access the corporate directory and other customized directories from an ATA 191.
- Create a personal directory.
- Set up speed dial and call forwarding numbers.
- Subscribe to services that are accessible from an ATA 191.

You can add users to Cisco Unified Communications Manager using this method:

- To add users individually, choose **User Management** > **End User** from Cisco Unified Communications Manager Administration.

Refer to the *Administration Guide for Cisco Unified Communications Manager and IM and Presence Service* for more information about adding users. Refer to the *System Configuration Guide for Cisco Unified Communications Manager* for details about the user information.

Emergency Call Support Background

Emergency call service providers can register an ATA's location for each IP-based phone in a company. The location information server (LIS) transfers the emergency response location (ERL) to the ATA. The ATA stores its location during registration, after the ATA restarts. The location entry can specify the street address, building number, floor, room, and other office location information.

When you place an emergency call, the ATA transfers the location to the call server. The call server forwards the call and the location to the emergency call service provider. The emergency call service provider forwards the call and a unique call-back number (ELIN) to the emergency services. The emergency service or public safety answering point (PSAP) receives the ATA's location. The PSAP also receives a number to call you back, if the call disconnects.

See Emergency Call Support Terminology, on page 13 for the terms used to describe emergency calls from the phone.

The phone requests new location information for the following activities:

- You register the ATA with the call server.
- You or the user restarts the ATA and the ATA was previously registered with the call server.
- You change the network interface used in the SIP registration.
- You change the IP address of the ATA.

If both of the location servers do not send a location response, the phone resends the location request every two minutes.

Emergency Call Support Terminology

The following terms describe emergency call support for the ATA.

- Emergency Location ID Number (ELIN)–A number used to represent one or more ATA lines that locate the person who dialed emergency services.
- Emergency Response Location (ERL)–A logical location that groups a set of ATA lines.
- HTTP Enabled Location Delivery (HELD)–An encrypted protocol that obtains the PIDF-LO location for the ATA from a location information server (LIS).
- Location Information Server (LIS)–A server that responds to a SIP-based ATA HELD request and provides the ATA location using a HELD XML response.
- Emergency Call Service Provider–The company that responds to an ATA HELD request with the ATA's location. When you make an emergency call (which carries the ATA's location), a call server routes the call to this company. The emergency call service provider adds an ELIN and routes the call to the emergency services (PSAP). If the call is disconnected, the PSAP uses the ELIN to reconnect with the ATA used to make the emergency call.
- Public Safety Answering Point (PSAP)–Any emergency service (for example, fire, police, or ambulance) joined to the Emergency Services IP Network.
- Universally Unique Identifier (UUID)–A 128-bit number used to uniquely identify a company using emergency call support.

Configure the ATA to Make Emergency Calls

Before you begin

Obtain an E911 location URL and a company ID for the ATA from your emergency calling service provider (for example, Redsky admin). You can use the same location URL and company ID for PHONE1 and PHONE2.

Procedure

Step 1 Sign into On Cisco Communication Manager Administration as an administrator.

Step 2 Configure a service profile:

a) Select **User Management** > > **User Settings** > **Service Profile**.
b) Create a new service profile with a unique name. For example, "Emergency Calling Profile".
c) Configure the fields in the section **Emergency Calling Profile**.

The **Organization ID**, **Secret**, and **Location Url** are provided by your emergency calling service provider.

For **Emergency Numbers**, enter the emergency service numbers, separated by commas. For example, `911,933`

d) Click **Save**.

Step 3 Associate an end user with the created service profile:
a) Select **User Management** > **End User**.
b) Create a new user or modify an existing user.
c) In the **Service Settings** section, select the service profile that you created from the **UC Service Profile** drop-down list.
d) Click **Save**.

Step 4 Associate a phone with the created or modified user:
a) Select **Device** > **Phone** to find an existing phone.
b) In the **Device Information** section, select **User** for the **Owner** field, and then select the user from the **Owner User ID** drop-down list.
c) Click **Save**.

Step 5 Create or modify an SIP dial rule for the emergency number:
a) Select **Call Routing** > **Dial Rules** > **SIP Dial Rules**.
b) Create a new SIP dial rule or modify an existing one.
c) If you choose to create a new SIP dial rule, select **7940_7960_OTHER** from the **Dial Pattern** drop-down list.
d) Enter a name and relevant descriptions for the SIP dial rule.
e) In the **Pattern Information** section, add patterns of the emergency number (such as, "911" and "933").
f) Click **Save**.

Step 6 Associate a phone with the created or modified SIP dial rule:
a) Select **Device** > **Phone**.
b) In the **Protocol Specific Information** section, select the SIP dial rule from the **SIP Dial Rules** drop-down list.
c) Click **Save**.

Step 7 Verify the E911 configurations on the ATA web page:
a) Select **Voice** > **Line <n>**.
b) Go to the section **Call Feature Settings**, check whether the parameter **Emergency Number** is configured as expected.
c) Go to the section **E911 Geolocation Configuration**, check whether the parameters are configured as expected.
d) Go to the section **Dial Plan**, chech whether the parameter is configured as expected.