Ātea FTR Full Time Recording

Administration & Configuration Guide

Version 4.1

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2. Document History

Date	Contributor	Description
18 August 2025	Callum Kātene	Initial draft for FTR version 4.

3. Document Purpose

This document details the components and architecture of Ātea Full Time Recording (FTR) and describes system connectivity and commissioning tasks.

4. Product Overview

Ātea FTR is a software solution that records phone calls made on Cisco IP Phones, writing the call media to audio files stored on the FTR server(s). In addition, metadata about the calls is captured in the embedded database. Appropriately authorised users can log into the FTR portal to listen to, and download, the recordings.

Recording groups are supported. FTR administrators can create groups of phones and assign those groups to FTR users. When a user logs in to FTR they will only see the recordings of the phones that are members of the groups they have access to.

Ātea FTR uses two distinct techniques to record phone calls. *Monitor-API Call Recording (MCR)* uses CTI control to invoke recording and *SIP Forking/Recording Profile* recording uses the phones' SIP Forking capabilities.

All Cisco phones to be recorded ***must*** have the *Built-In-Bridge* feature enabled, irrespective of the recording technique in use.

The two recording techniques described here can be used simultaneously to achieve *High Availability.*

Ātea FTR uses the CUCM AXL API to 1) interrogate system configuration, 2) authenticate users, and 3) to enable and disable call recording on specific phones.

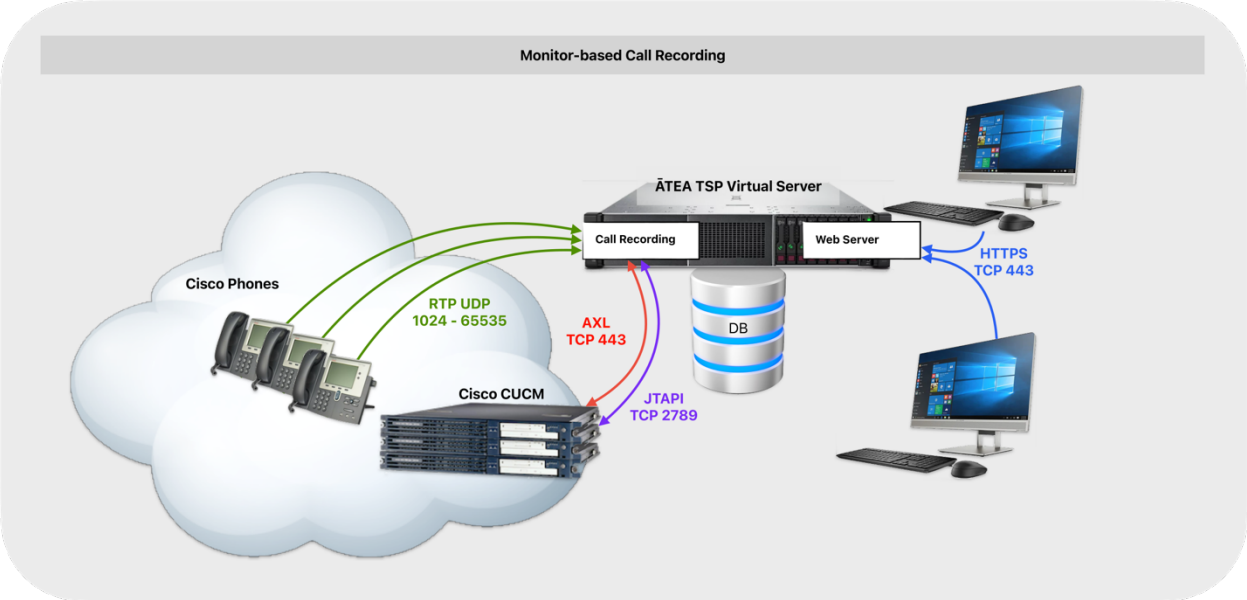
a. Monitor-API Call Recording (MCR)

Ātea *Monitor-API Call Recording (MCR)* is a CTI application that uses Cisco's silent monitor capability to capture phone call media. MCR makes use of a CTI Route Point and a CTI Port as the monitoring call endpoints. When a call starts on a recorded phone, MCR initiates a monitoring "call" from the CTI Port to the phone. Once the monitoring call is answered, MCR transfers the call to the CTI Route Point, which records the call media, writing it to a local file.

All metadata relating to the call is written to the embedded database and provides the details accessed via the FTR portal.

To enable MCR call recording the following Cisco CUCM elements are required:

1) An application user with the appropriate CTI and AXL permissions.
2) A CTI Port with one unique DN. The DN on the CTI Port must be configured with a Monitoring Calling Search Space that will allow for calls to all phones that are to be recorded.
3) A CTI Route Point with one unique DN. The CTI Port must have the necessary permissions to be able to transfer calls to this CTI Route Point.



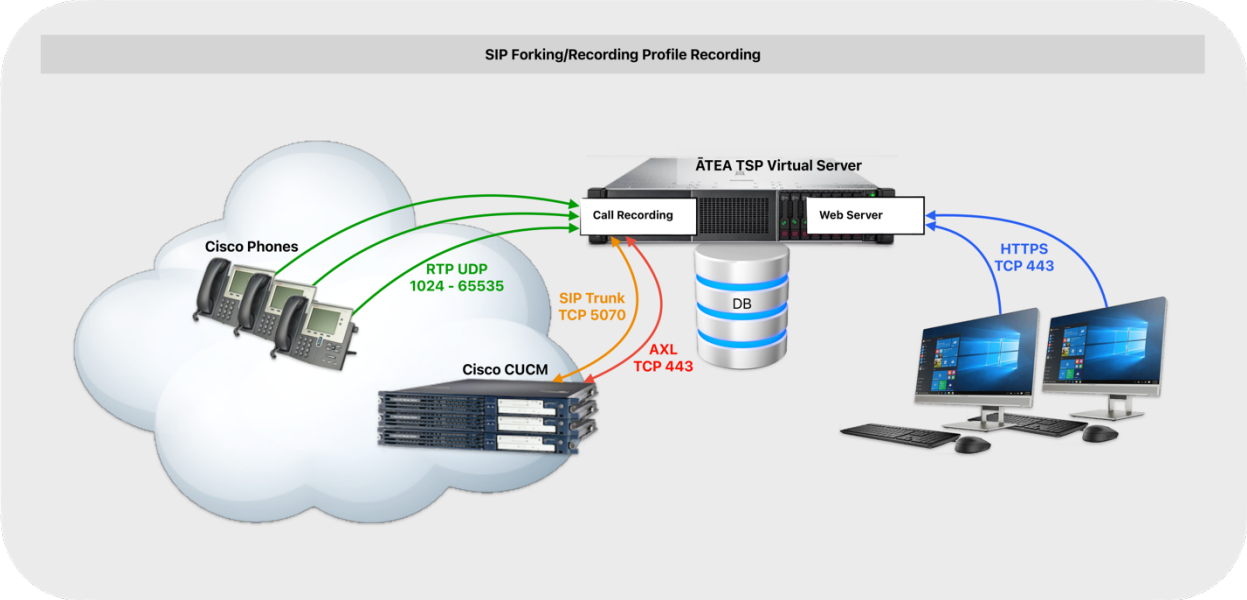
b. SIP Forking/Recording Profile Recording

Cisco IPPhone handsets have a call recording facility that uses the phones' SIP forking capability to send call media to a SIP trunk. Ātea FTR provides a SIP server suitable for connection to CUCM SIP trunks for the purposes of recording.

To enable SIP forking recording the following CUCM elements are required:

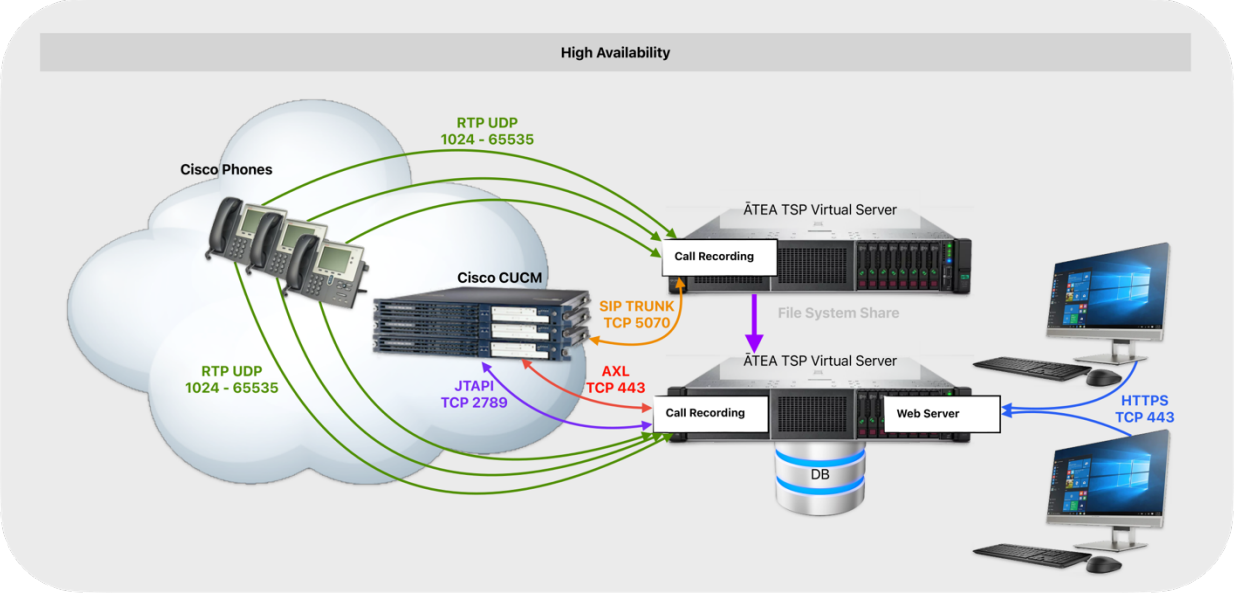
1) A SIP Trunk pointing to TCP port 5070 on the Ātea FTR server.
2) A unique Route Pattern whose destination[1] is the SIP Trunk.
3) A Recording Profile that points to the Route Pattern.
4) Automatic Call Recording enabled[2] on the lines of all phones to be recorded, specifying the Recording Profile.

[1] If required, route lists and route groups can be used to provide resiliency
[2] While sip-forking recording is enabled in CUCM, note that FTR will ***not*** record calls on unlicensed phones, even if they have call recording enabled.



c. High Availability

High Availability can be achieved by using both recording techniques at the same time. To ensure full redundancy, a second Ātea TSP server is deployed. MCR recording is performed by the original server, and SIP-forking recording is performed by the additional server. Calls recorded on this second server are written to the local disk but available via the MCR server through an NFS file share. All metadata is written to the database on the primary server.



5. Connectivity

a. AXL and Call Media

Ātea FTR uses the CUCM AXL API for device control and user authentication. The AXL service on the CUCM publisher is accessed via the HTTPS protocol (TCP 443).

Call media is sent from the phones to the Ātea FTR server using the RTP protocol (UDP port range: 1024 – 65535). A single media stream per call is sent.

b. Monitor-API Call Recording (MCR)

In addition to the connectivity requirements above, Ātea MCR utilises TCP port 2789 for CTI control.

c. SIP Forking/Recording Profile Recording

In addition to the requirements listed above under AXL and Call Media, SIP Forking recording provides a SIP server on port TCP 5070.

d. High Availability

The connectivity requirements for High Availability are simply the combination of all connectivity mechanisms listed above.

6. Requirements

Ātea FTR records calls using either the Sip Forking or the Monitoring capability of the Cisco handsets. Therefore, only handsets that support these features using a Built-In Bridge are supported.

7. Configuring CUCM

a. AXL User Permissions

Ātea FTR requires read/write access via the CUCM AXL API. An application user is required that has the following roles:

```
1) Standard AXL API Access
2) Standard CCM Admin Users
```

b. CTI User Permissions

If MCR recording is required, the application user also needs the following roles:

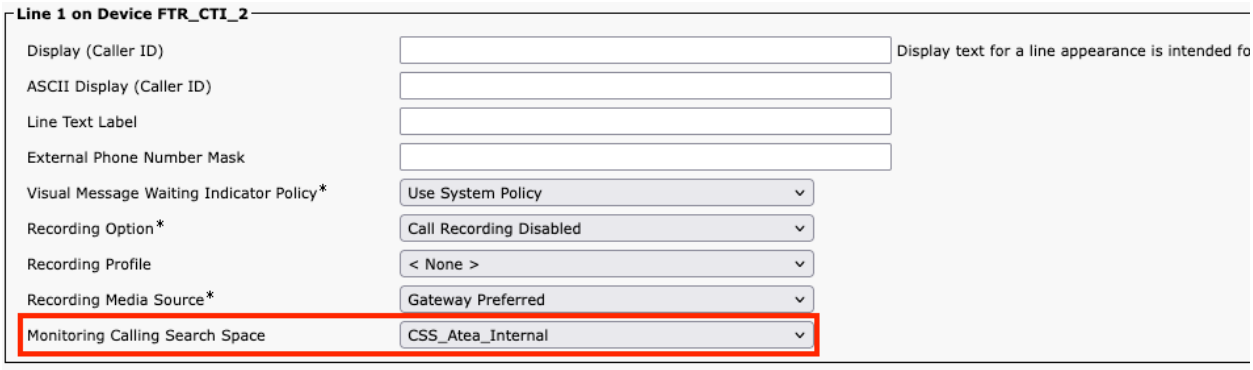
```
1) Standard CTI Allow Call Monitoring
2) Standard CTI Allow Control of All Devices
3) Standard CTI Enabled
4) Standard CTI Control of Phones supporting Connected Xfer and conf
```

c. CTI Port

MCR requires a single CTI Port that has a single, unique DN as line 1. All settings on the CTI port are standard except that line 1's Monitoring Calling Search Space must be set to a CSS that includes the partitions of all the phone lines to be recorded.

This CTI Port must be associated with the application user.

The screenshot below shows the Monitoring Calling Search Space on the line configuration:



d. CTI Route Point

MCR requires a single CTI Route Point with a single DN with a unique number. The line's partition must be such that the CTI Port Can transfer calls to it.

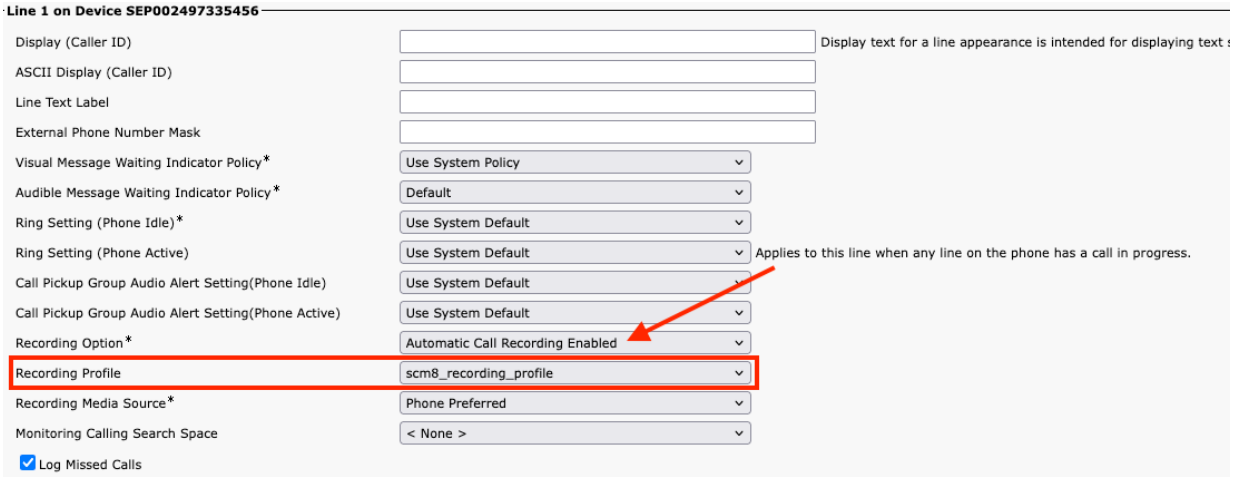
This CTI Route Point must also be associated with the application user.

e. SIP Forking/Recording Profile Recording

Cisco CUCM supports automatic recording, which involves the triggering of dual SIP calls to a predetermined destination for the purposes of capturing call media.

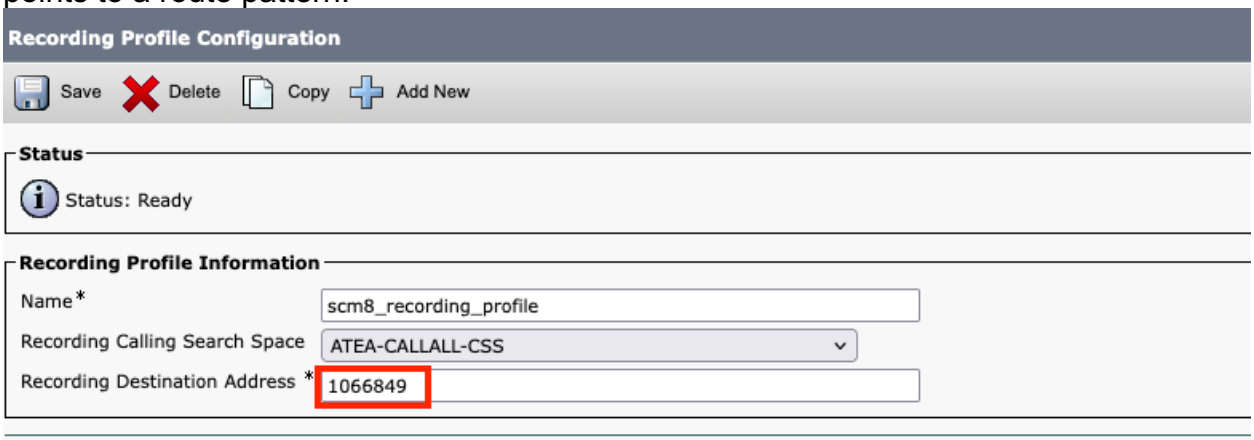
Phone Configuration

Enabling automatic call recording is a matter of setting the feature in the configuration for the phone's line and selecting a *Recording Profile.*



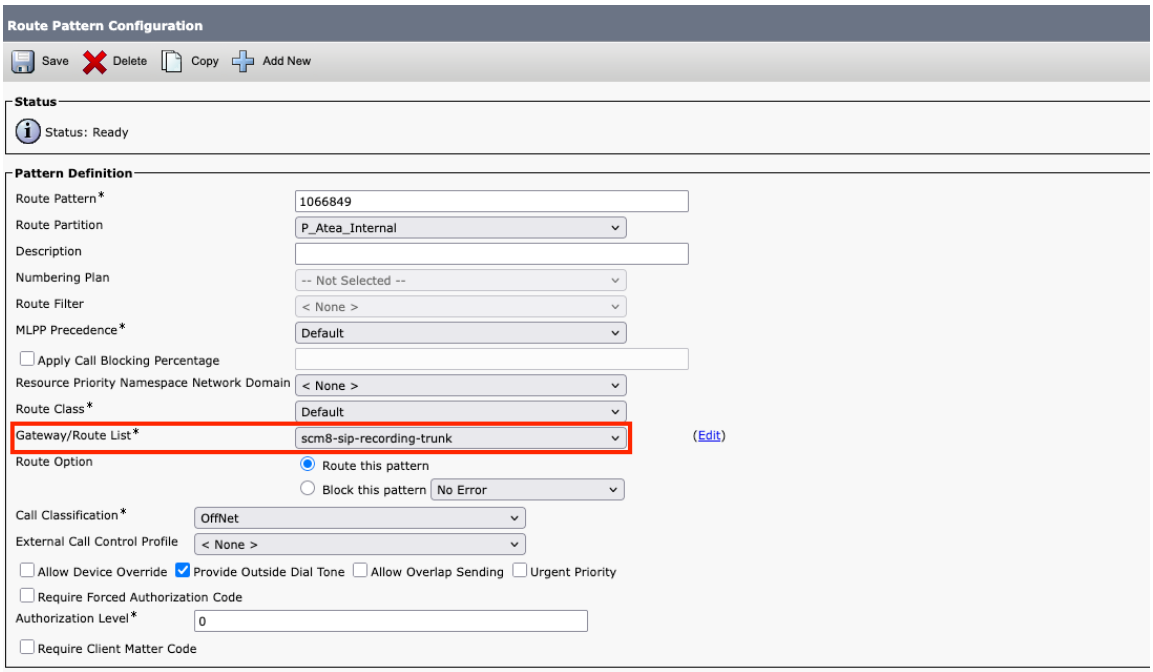
Recording Profile

The recording profile specifies a destination address: This is the number that will be called by the phone for the calls to convey the recording media. In this case, the number points to a route pattern.



Route Pattern

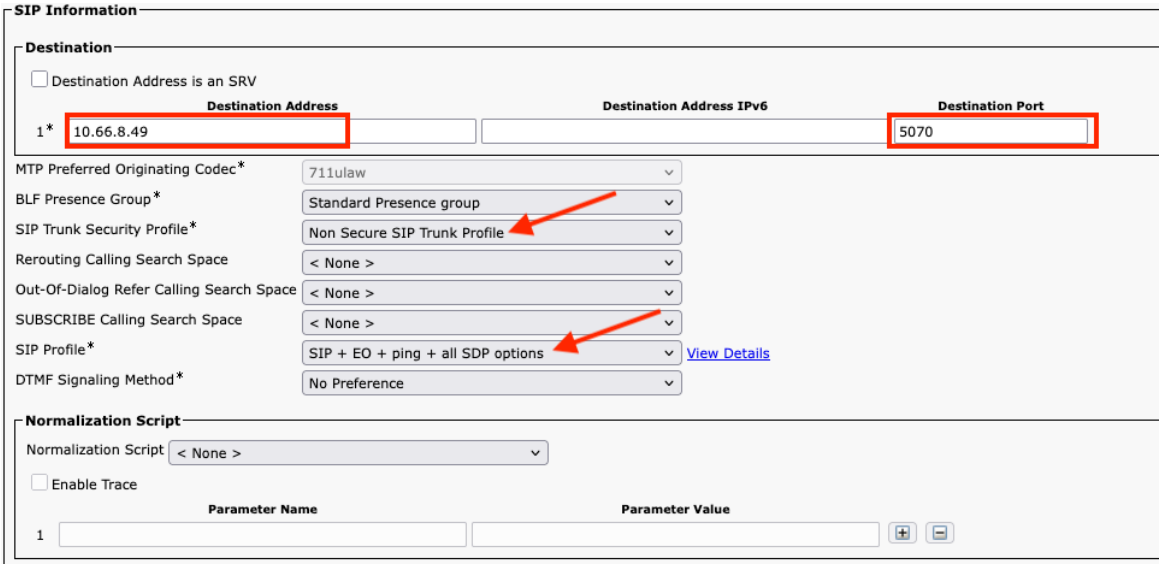
The route pattern referenced by the Recording Profile determines where the recording calls will be routed to. The *Gateway/Route List* configuration element specifies the SIP trunk[3] that the recording calls should be routed to.



[3] Alternatively, a route list can be specified here.

SIP Trunk

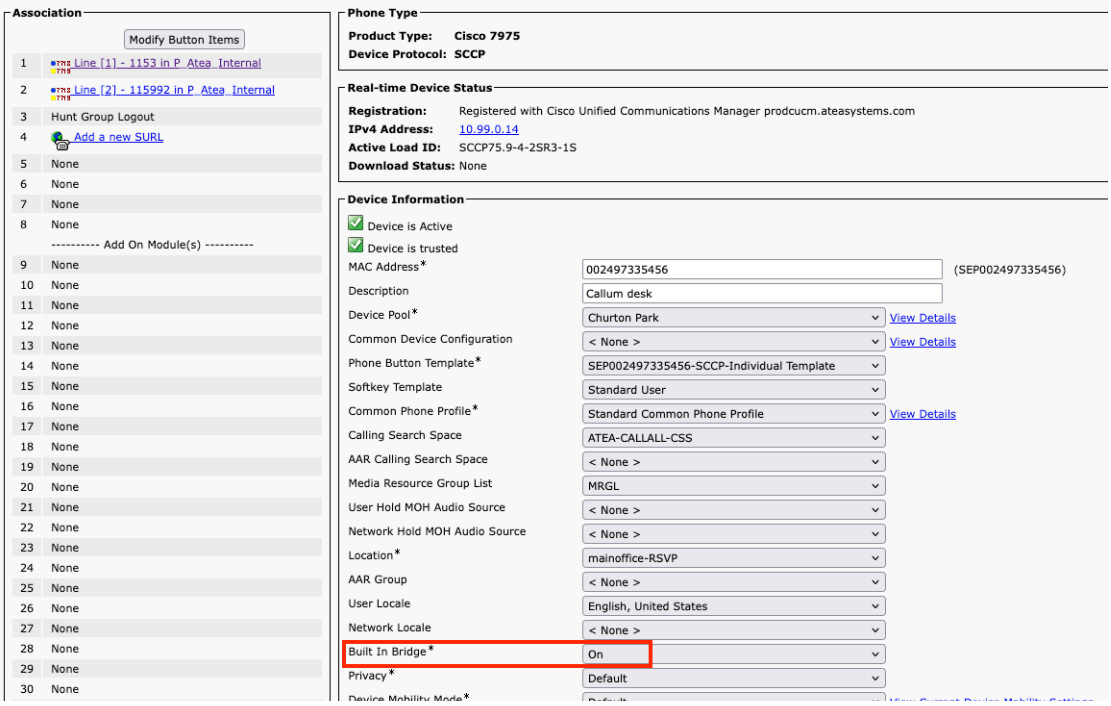
The SIP trunk needs to be configured with the SIP details of the Ātea FTR server. In the *SIP Information* section, specify the destination address and port. Also set the *SIP Trunk Security Profile* and the *SIP Profile* as per the screenshot below.



8. Phone Handsets

Ātea FTR utilises the recording and monitoring capabilities of the Cisco phones – If the phone endpoint does not support these functions, Ātea FTR cannot record their calls.

In addition to enabling recording on the phones' lines, the phones' *Built-In Bridge* feature must be enabled, as per the screenshot below.



9. About Recording Files

Call recordings require 480 KB[4] of disk space per minute of conversation. However, Ātea FTR can also convert the recorded files to either .mp3 or .ogg format after the recording is completed. These formats provide compression that can achieve up to an 8 to 1 storage saving.

10. Glossary of Terms

Term	Definition
AXL (Application eXtension Language)	A Cisco API that allows external applications to read and write to the Cisco Unified Communications Manager (CUCM) configuration database over HTTPS.
Built-In Bridge (BIB)	A feature on Cisco IP Phones that enables the phone to fork media streams for monitoring or recording purposes. Required for both MCR and SIP Forking.
CTI (Computer Telephony Integration)	Technology that allows interactions on a telephone and a computer to be integrated or coordinated.
CTI Port	A virtual endpoint used by CTI applications to initiate or receive calls as if it were a physical device.
CTI Route Point	A special type of virtual directory number used to queue and redirect calls within CTI applications.

[4] G.711 ulaw/alaw is the standard recording format. Other formats are available.

Term	Definition
CSS (Calling Search Space)	A list of partitions that a device or line can access to complete a call. Determines which directory numbers a phone can reach.
CUCM (Cisco Unified Communications Manager)	Cisco’s call control platform for voice and video communications.
DN (Directory Number)	A number assigned to a phone line or CTI port/route point, used to identify the line for routing purposes.
FTR (Full Time Recording)	Ātea’s solution for continuous or automatic recording of Cisco IP Phone calls, including metadata capture and user portal access.
High Availability (HA)	A system design approach ensuring operational continuity in the event of component failure by using redundancy.
MCR (Monitor-API Call Recording)	A call recording method using Cisco CTI and silent monitoring features to capture audio streams.
NFS (Network File System)	A network protocol that allows file access over a network as if they were local, used here for accessing call recordings across servers.
Partition	A logical grouping of directory numbers in CUCM. Used in combination with CSS to control call routing and access.
Recording Profile	A CUCM configuration that defines how and where a recorded phone sends media streams for SIP Forking recording.
RTP (Real-time Transport Protocol)	A standard protocol for delivering audio and video over IP networks. Used to transmit call media from phones to the FTR server.
Route Pattern	A CUCM configuration element used to route calls to external or internal destinations via trunks or route lists.
SIP (Session Initiation Protocol)	A protocol used to establish, modify, and terminate multimedia sessions including VoIP calls.
SIP Forking	A feature that allows a phone to send duplicate media streams to a SIP destination (e.g., for recording purposes).
SIP Trunk	A virtual connection between CUCM and an external SIP server that enables the sending of SIP calls or media streams.