Full Time Recording (SIP)
FTR 2.5.0

Installation and Administration Guide

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Document History

Date	Author	Version	Summary
4/05/2016	Murray Lum	2.5e	New SIP timer settings and update for 2.5.00 Increase default disk size to 150MB New settings to increase SIP session expiry, and disable G.729 silence suppression Added instructions for backups and alerting
24/04/2015	Murray Lum	2.5	Updated for release 2.2.40, removed APEX section, changed install to VM import, expanded management of phones and licenses
25/09/2012	Te Kairangi Katene	2.4	Removed cdr references for cm 8.5+
15/09/2010	Jamie Brown	2.3	Updated APEX section
16/08/2010	Jamie Brown	2.2	Updated section SIP Trucks
26/07/2010	Jamie Brown	2.1	Updated section on managing recordings
16/07/2010	Jamie Brown	2	Updated diagrams and troubleshooting
17/06/2010	Jamie Brown	1.0	Initial administration guide

Related Documents

Document	Description
FTR User guide	User guide

1 Scope of this document

This guide describes the administrative tasks to configure and maintain the Atea Full Time Recording (FTR) application. This uses SIP based call recording.

This document covers:

- FTR overview
- FTR VM appliance installation (pre-configured instance supplied by Atea)
- UCM configuration for the FTR application
- Recording file basic management
- Troubleshooting tips

For additional information on configuring the Cisco UCM, please see the Cisco documentation.

2 Product Overview

In SIP based call recording, recording streams get forked from agent IP phone to the recorder. The agent voice and the customer voice are sent separately. Two different methods of call recording are supported by Cisco, being automatic (full time) recording and application invoked recording. This administration guide is for full time recording.

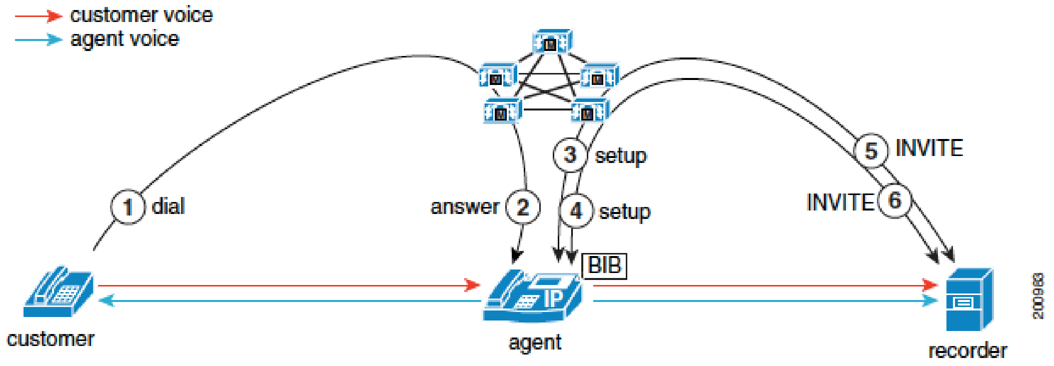


Figure 1: Full time recording

The steps for establishing a full time recording session are:

1. A customer calls into the call centre.
2. The call routes to the agent. The agent answers the call. The agent IP phone starts to exchange media streams with the customer.
3. As the agent line appearance is configured for automatic recording, the recording session for the media streams automatically gets triggered. The CUCM first makes a recording call to the built-in bridge (BIB) of the agent IP phone for the agent voice.
4. The CUCM makes a second recording call to the BIB of the agent IP phone for the customer voice.

5. The recorder receives and answers the recording call setup messages from CUCM for the agent voice using SIP. The agent IP phone forks the agent voice stream to the recorder.
6. The recorder receives and answers the recording call setup messages from CUCM for the customer voice using SIP. The agent IP phone forks the customer voice stream to the recorder.

To record conversations, set up each Cisco IP phone with a recording profile. The profile directs copies of the voice streams to the Atea FTR server. The directory number for the IP phone must also be included on the FTR server.

3 Supplied Software

The Atea FTR server appliance is usually supplied as a virtual machine using OVF or OVA files(s). This VM may be set up as a separate appliance independent of other Atea supplied applications. The resource availability can be critical for the recording appliance.

Virtual Machine guest resources

Item	Resources
Processor	2 virtual CPUs (2.1GHz min) 64-bit
RAM	4GB
Disk	150GB HDD (Resilient data store recommended)
Additional disk for recordings	Usually greater than 80GB HDD – depending on recording retention requirements. See the section on recording files for a sizing table.

4 Installing and Configuring FTR recording

4.1 Pre-requisites

Before you begin, you will need:

- Access to the CUCM to configure the SIP recording and recording profiles
- Virtual machine resources for the FTR VM (pre-configuration of VM host is **not** required)
- The VM OVF (or OVA) files supplied by Atea
- A back-up and restoration strategy for the server
- An archive strategy for the recordings
- A remote access mechanism to allow Atea Support to configure and support the appliance.

To allow Atea to create the VM please supply the information below to support@ateasystems.com, using the form at https://www.ateasystems.com/virtual-server-config/

- hostname
- IP-address (& mask & gateway)
- DNS details
- NTP IP-address
- SMTP IP-address

Forward any security documentation to support@ateasystems.com.

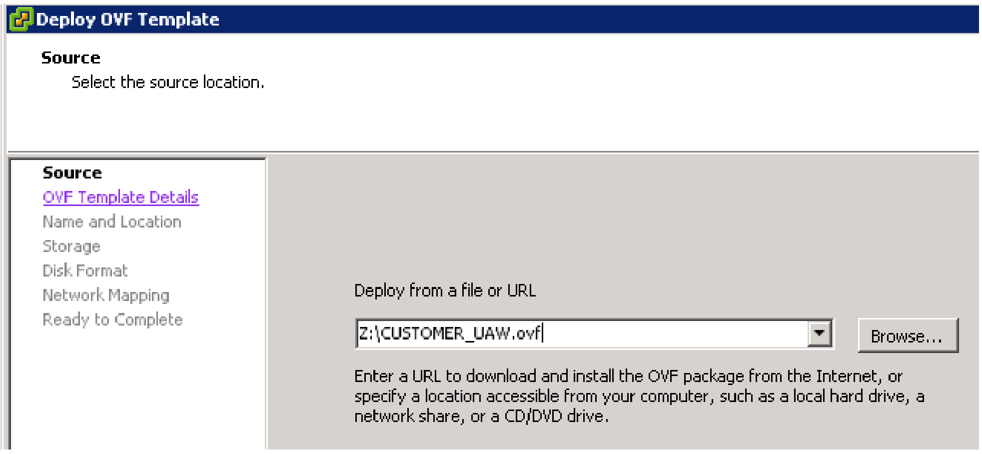
Information required later to setup and generate a license file:

- Server MAC address.

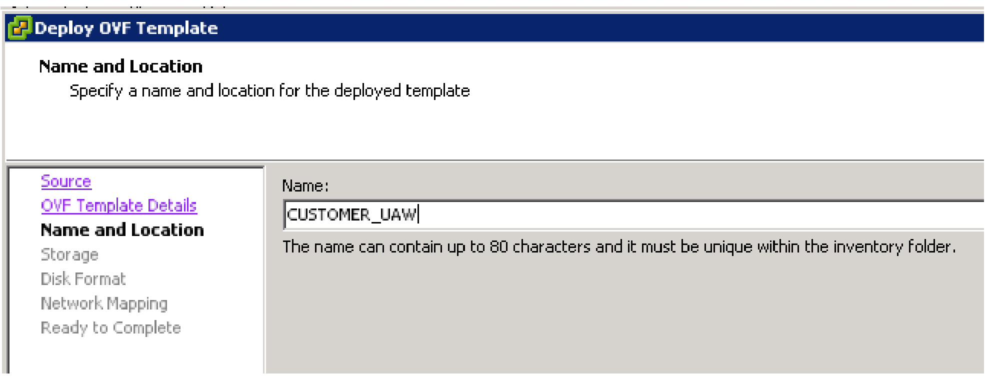
4.2 Install the Virtual Machine

The Atea FTR is normally supplied as a virtual machine, shipped in OVF format. Download and check the files provided.

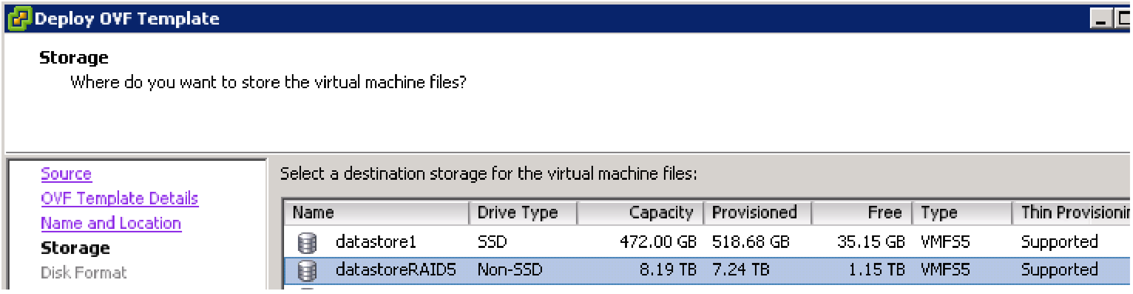
1. From your preferred client (such as the vSphere Client), navigate to "**Deploy OVF Template**"
2. Browse to the location where the downloaded files are stored and select the [filename].ovf (or .ova)



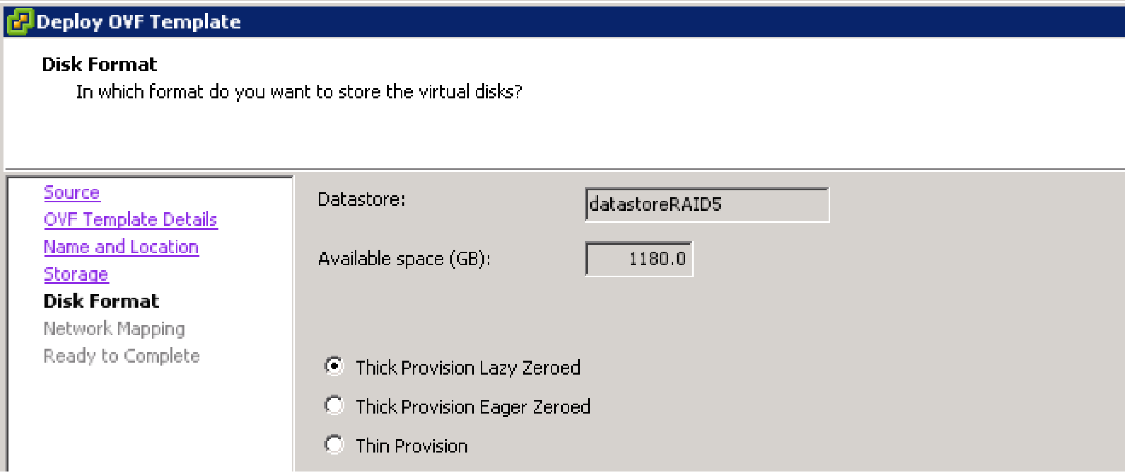
3. Enter a **name** for the instance



4. Select the data-store where you want the VM guest to reside



5. Select the **Disk Format** (*Thick Provision is recommended*)



6. Start the server instance (**Power On**)
7. Log into the console with user: **thirdparty**, and the password provided. You must change the password on first login. Enter the old password again, then the new password twice.

```
thirdparty@9.1.1.186's password:
You are required to change your password immediately (root enforced)
Last login: Wed Apr 22 11:22:53 2015 from 9.1.1.188
#################################################################
# Oracle Linux 6.6-x86_64 KS-installed [Wed Apr 22 10:35:40 NZST 2015] #
# Atea Systems Ltd - TSP                                        #
# All connections are monitored and recorded                    #
# Disconnect IMMEDIATELY if you are not an authorized user!     #
#################################################################
WARNING: Your password has expired.
You must change your password now and login again!
Changing password for user thirdparty.
Changing password for thirdparty.
(current) UNIX password:
New password:
Retype new password:
passwd: all authentication tokens updated successfully.
```

This confirms that you have access to the VM in the future for maintenance.
You may now set up both the server backup and remote access for Atea support.

4.3 Configure backups

The automatic backup cycle runs daily at 11pm.

The components backed up locally on the servers are:

- Linux configuration (7 days kept with day of week indicator from 1 to 7)
- Atea applications (7 days kept)
- Atea application properties (7 days kept)
- Oracle database (2 days kept)

Atea recommends that sFTP is used to copy the application and database backups to a network location.

The procedure to enable the sFTP copying is (see also the Atea website):

ssh thirdparty@ateaserver-ip-address

sudo su -

vim /etc/atea/scripts/sftpBackup.sh (enter the sftp server details)

Change the host, user, password and path parameters on lines 4-7

vim /etc/atea/scripts/fullBackup.sh (change line 23, DO_BACKUP_COPY=true)

Now check that the ssh rsa key is stored and connectivity is okay by connecting to the sftp server

sftp user@sftpserver (enter password to establish connection)

exit

4.4 Configure alerting

A monitor script runs every 10 mins by default and sends alerts if certain conditions are met. These include CPU, IOWAIT, Free disk and memory and whether the application can connect to the Oracle database.

To change the monitor script to send alerts to your service provider, you'll need to modify the script:

ssh thirdparty@ateaserver-ip-address

sudo su -

vim /etc/atea/scripts/monitor.sh

Change the MAILTO and MAILFROM lines with the Service-Provider specific email addresses.

5 Configure the CUCM for Recording

The recording application uses a SIP trunk to record the telephone calls. This can be either a single trunk, or multiple trunks spread across several devices for resilience. There are additional steps if there are multiple trunks.

For a single recording server, the normal setup steps are:

1. Configure the CUCM System Service Parameters
2. Configure the SIP trunk for the server
3. Configure the Route Pattern to use the recording device
4. Configure the recording profile
5. Set up the phones for recording

For several recording servers, we suggest you follow these steps:

1. Configure the CUCM System Service Parameters
2. Configure the SIP trunk for each of the servers
3. Create a route group for each recording trunk and add the SIP devices (a route group specifies the order the trunks are selected)
4. Create a route list for each server which specifies the order the route groups are used
5. Configure the Route Pattern to use the route list
6. Configure the recording profile
7. Set up the phones for recording

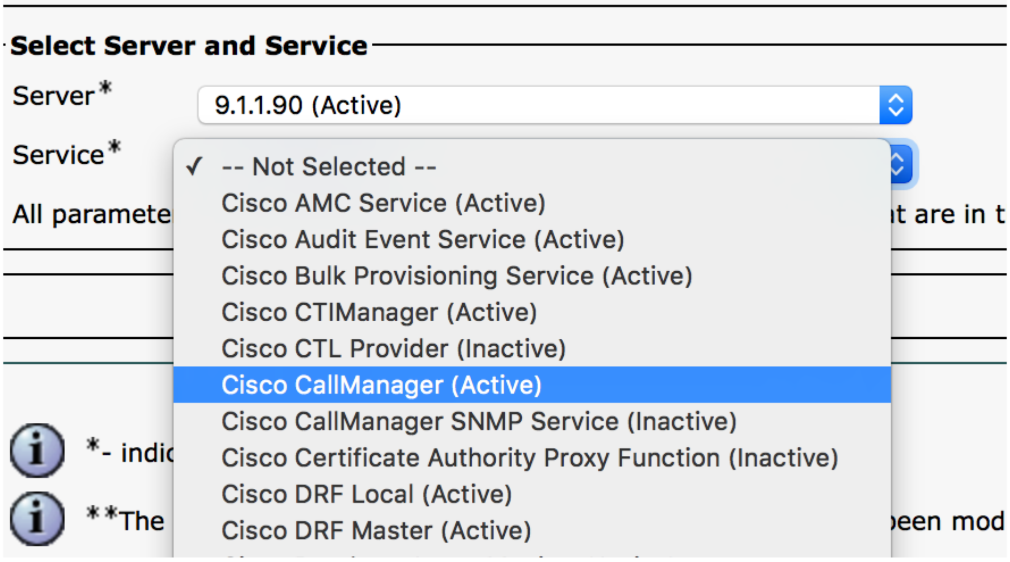
5.1 Configure CUCM System Service Parameters – single and multiple devices

Two settings must be changed here:

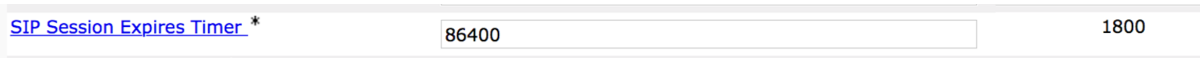
- The **SIP session expires timer** must be increased otherwise each recording is cut off at 16 minutes.
- Silence Suppression for G.729 must be stripped, otherwise some recordings will overlap when they shouldn't.

To change these settings:

1. Log onto the CUCM administration
2. Select the Server and Service **Cisco CallManager (Active)**



3. Go to **Service Parameter Configuration**
4. Set the **SIP Session Expires Timer** to **86400** (This is the number of seconds in 24 hours)



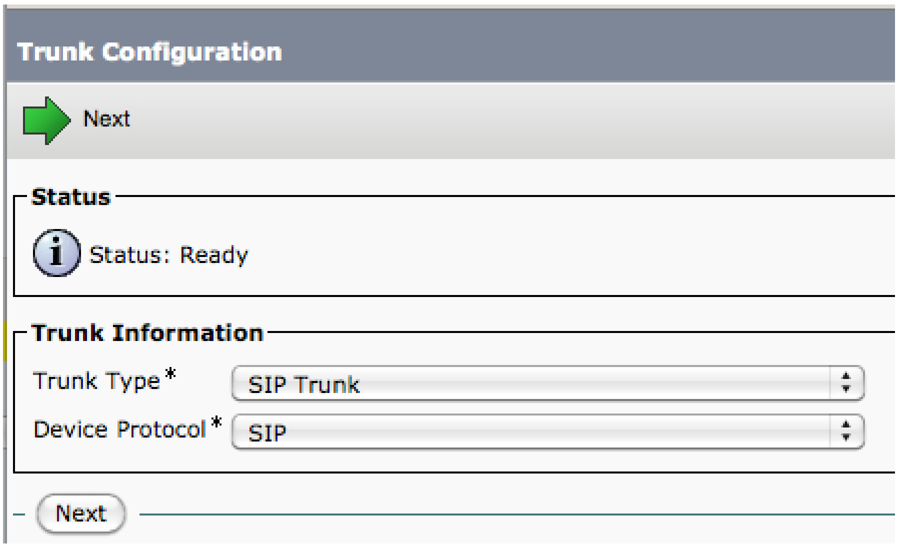
Disable silence suppression:

5. Go to the parameter **Strip G.729 Annex B (Silence Suppression) from Capabilities** and set this to **True**

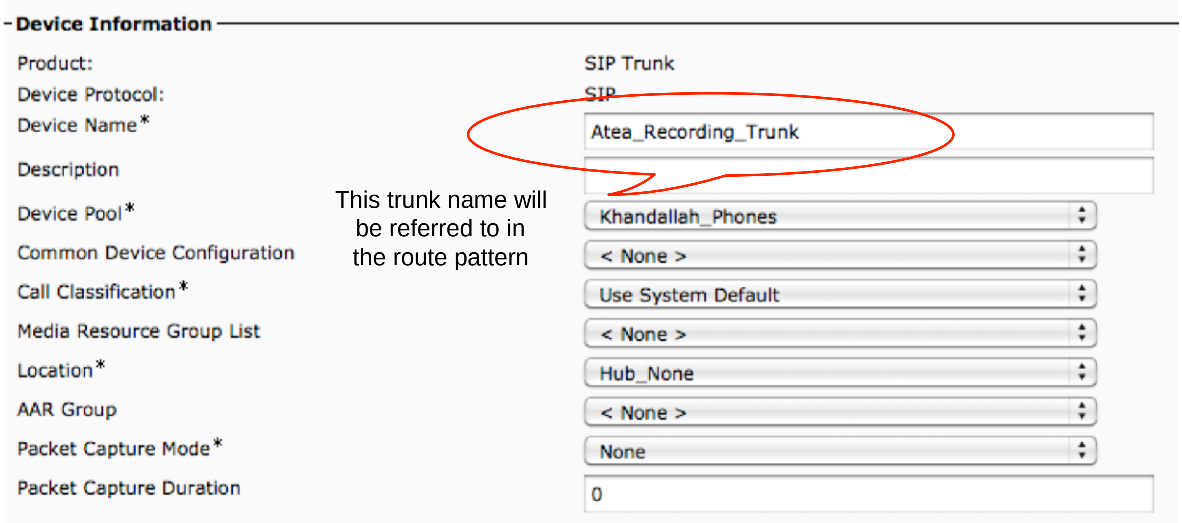
5.2 Configure SIP Trunk device – single and multiple devices

Here are the steps to create a trunk on a device. Repeat these for each recording device or FTR server that you have.

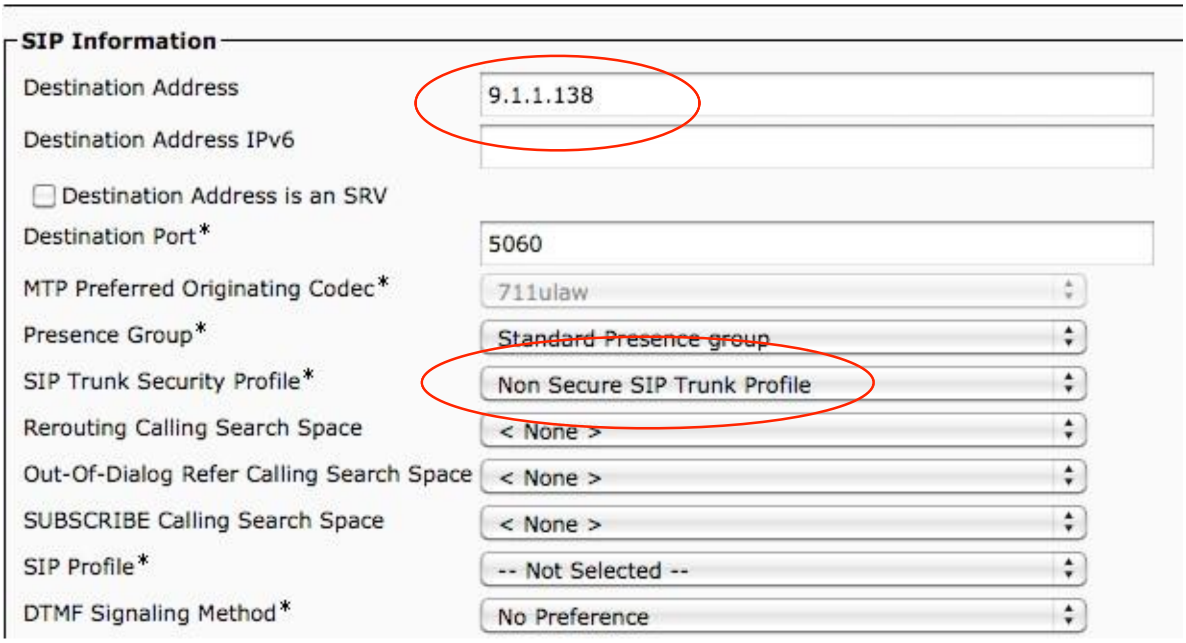
1. Create a SIP trunk. CUCM administration -> Trunk Configuration
   Trunk type = SIP trunk
   Device protocol = SIP



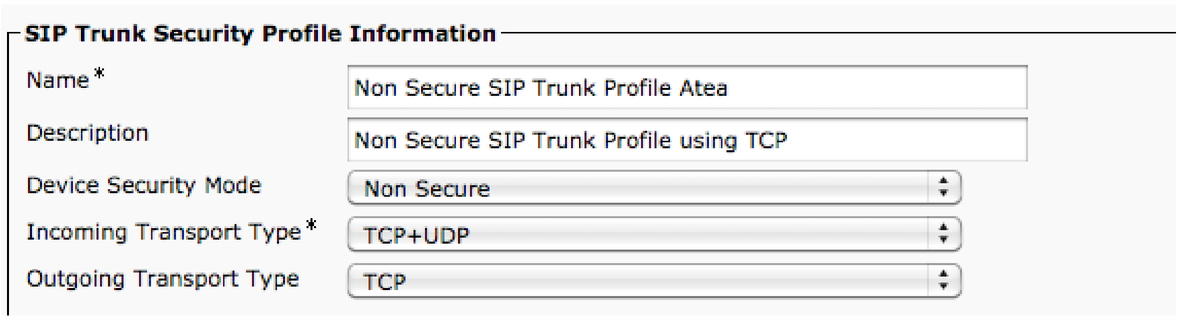
2. Set the device information
   Device Name - e.g. Atea_Recording_Trunk (this appears in the route pattern for single devices or the Route Group if there are several devices)
   Device Pool – usually the same as the phones to be recorded



3. Set the SIP information
   Destination address – this is the server with the Atea FTR application
   SIP Trunk Security Profile – this must match the setting on the Atea FTR application for which the default setting is “Non secure SIP Trunk Profile”



4. Set the SIP Trunk Security Profile Information
   The transport may be TCP or UDP. The default setting on the Atea FTR is TCP (it looks for SIP on TCP port 5060).



5.3 Create a Route Group – multiple devices only

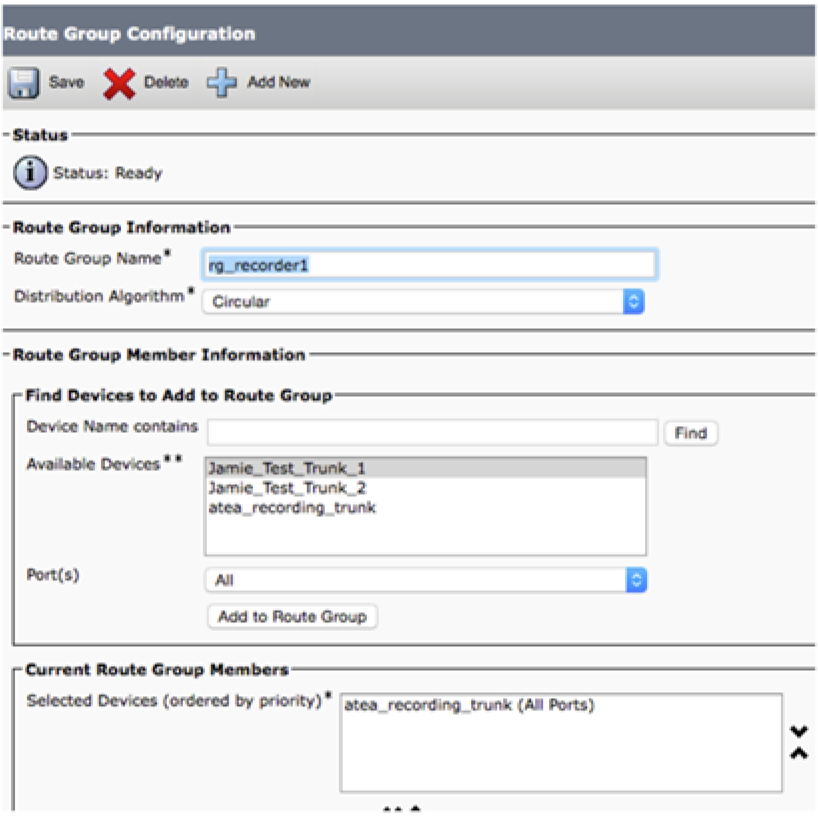
Skip this step if you have a single recording server.

For multiple devices, ensure you have configured the SIP trunk devices for each recording server.

For resilience, create a Route Group for each recording trunk on each device.

To configure the Route Group:

1. Choose Call Routing > Route/Hunt > Route Group
2. Enter a name for the route group
3. Add each device to the route group.



5.4 Create a route list for each server

Skip this step if you have a single recording server.

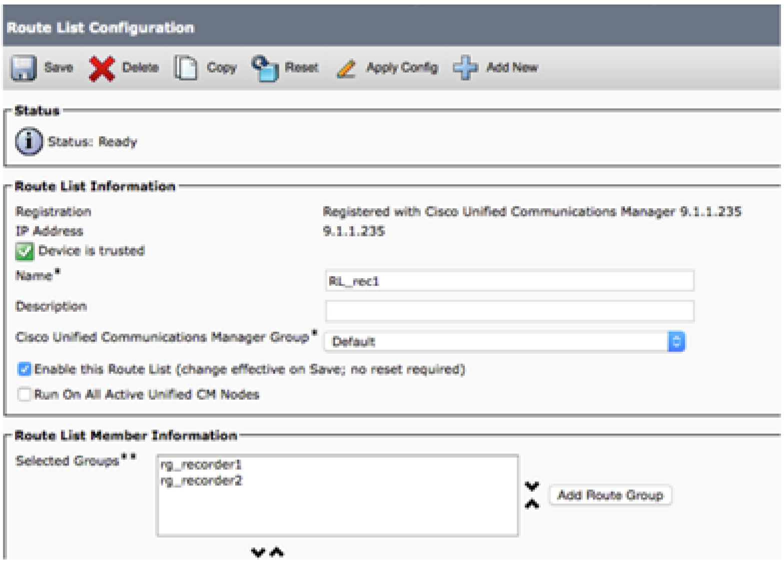
Configure a route list for each recorder to determine the priority order that the route groups apply.

Where there are say two recorders with two route groups, set the route list with the route groups in the opposite order. For example:

- route list 1: route group 1 then route group 2
- route list 2: route group 2 then route group 1

To configure the route list:

1. Choose Call Routing > Route/Hunt > Route List
2. Enter a name for the route list
3. Add each device to the route group.

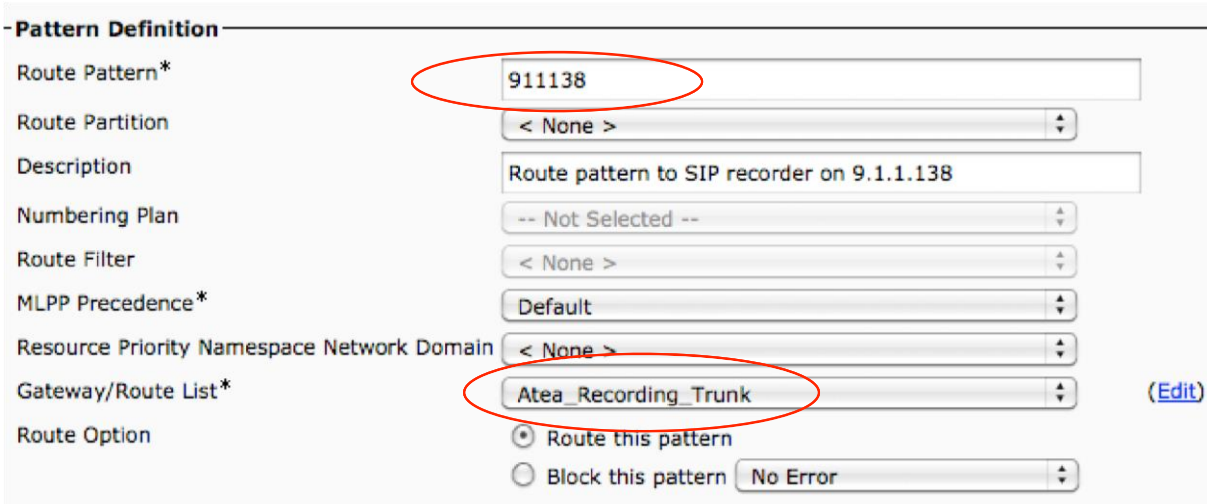


5.5 Configure a route pattern for the SIP trunk

We'll need a route pattern to direct calls to the SIP trunk gateway (or the route lists if several devices).

The convention for the route pattern is to use the destination IP address for the trunk with the period marks (".") removed. For example, the route pattern is 911138 for the SIP trunk on 9.1.1.138 as shown in the destination above.

Set the Gateway/Route List to the device name we created earlier e.g. Atea_Recording_Trunk, or the route lists where these are used.

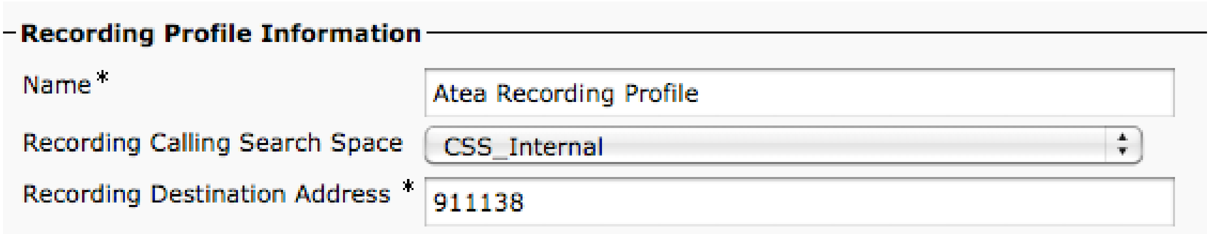


5.6 Configure the recording profile

Create a recording profile to point to the newly created route pattern. Apply the recording profile to the phone lines that are to be recorded.

A recording profile needs to be defined and pointed to the newly created route pattern. This recording profile will be applied to the phone lines that are to be recorded. The profile configuration screen can be found under Device - Device Settings screen.

1. Navigate to the Device > Device Settings screen
2. Select Recording Profile configuration and enter the details
   Name e.g. Atea Recording Profile
   Recording Calling Search Space e.g. CSS_Internal
   Recording Destination Address i.e. the name of the route pattern created above.

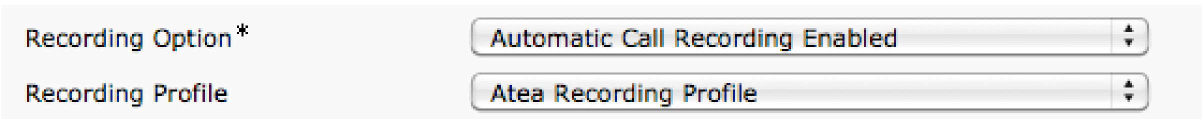


5.7 Turn on the built-in bridge on the phone

For each phone to be recorded, enable the built-in bridge. This is enabled on the device itself. On the phone configuration screen in the CUCM administration set "Built In Bridge" to "On".

5.8 Apply the recording profile to the phone line(s)

Each line that requires recording set the Recording Option to "Automatic Call Recording Enabled".

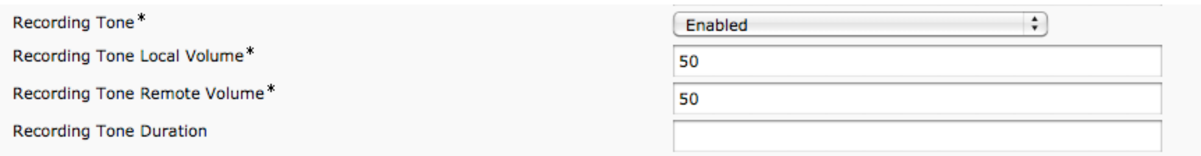


5.9 Enable the recording tone (optional)

You can set a tone to be played during calls to indicate that the call is being recorded. To enable this, **turn on** the notification tone cluster parameter (the default is off) and the setting on each individual phone.

1. Set the cluster wide service parameter to play the recording tone. From the CUCM administration, select the Cisco Call Manager service and scroll down to the call recording feature.
   Set "**Play Recording Notification Tone**" parameters to **true**.

2. Now enable the tone for each phone device. This is done on the device itself, not the individual lines. For information on these settings, refer to the Cisco documentation.



6 Recording files

6.1 General

Recording files are created in several phases. Initially, the application writes the audio header for the file, and then writes the voice-call data in real-time. When the call ends, the file is closed.

The file type is a wave file with the ".au" suffix. This type of audio file is playable by most media players. This file type is easily copied and stored. Users may copy the file to their local workstation to listen to it, or if they want to share it with a colleague for evaluation.

The recordings can searched directly from the FTR application.

6.2 Disk allocation for recordings

Recordings are stored on a separate disk mounted to the server that runs the FTR application. The suggested budget for sizing this disk is to allow 480kB per minute of recording. Recordings are stored in an uncompressed format.

Recording duration	Disk budget
1 minute	480 kB
1 hour	28.8 MB
100 hours	2.88 GB
1,000 hours	28.8 GB
10,000 hours	288 GB

6.3 Managing recordings

Recording files are located in the folder specified in the application.

Here is an excerpt from the BibCallRecorder properties file.

Property	Value	Comment
SipCallRecorder.recording_directory	/var/recording/	The path must exist

Recordings are stored separately to the call record database. The database includes a pointer to the file location.

If the recording file is moved to another location the recording will display as “not found” on the call search screen. Restoring the file back to the original folder allows the file to be played or retrieved using the normal screen to access the database.

Recordings may be archived or moved manually using operating system commands or other utilities. It is common practice to move or remove older recording files to maintain space on the disk.

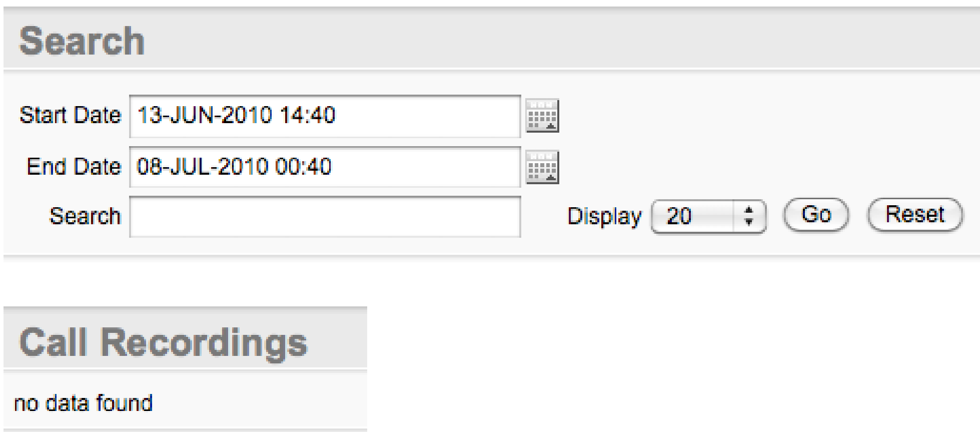
7 User Management

7.1 Accessing the FTR landing page – recording supervisors

To connect to the FTR application, use your web browser to navigate to this page –

http://[IPaddressOfServer]:8080/apexf?p=103

Enter the user name and password. This takes you a search screen for the recordings.

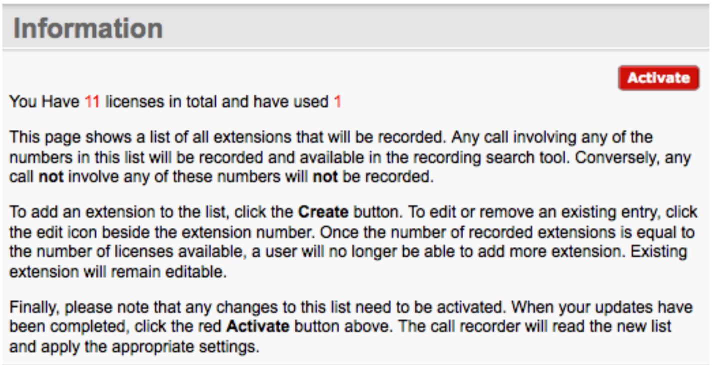


7.2 Licensing and managing the phones to be recorded

The FTR application is licensed for the number of phones to be recorded. To manage these phones, use a web browser to navigate to this page:

http://[IPaddressOfServer]:8080/apex/f?p=106

This page shows how many licenses are consumed, and the list of phones (DN directory numbers or extensions) to be recorded.

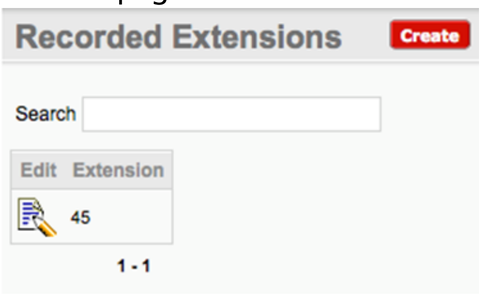


Note – in order for a phone to be recorded it must be set up in the CUCM with a recording profile, as well as this list.

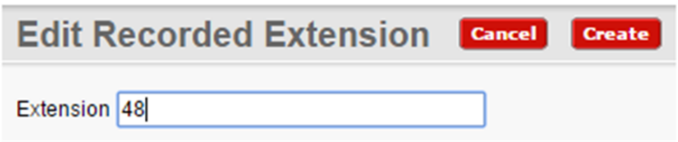
To change the size of your license, please contact Atea Systems.

To add phones to the list:

1. From the main **Manage Recorded Extensions** page, click the **Create** button to go to the next page



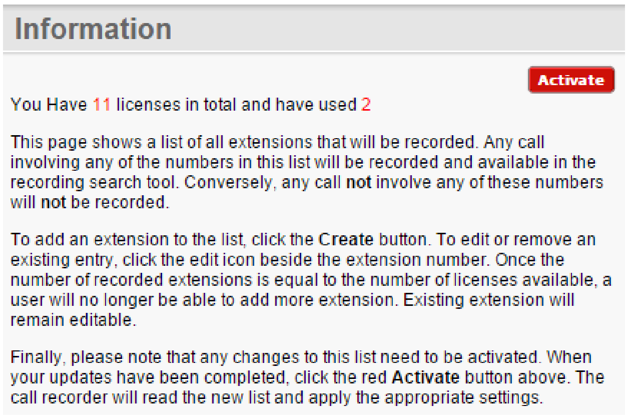
2. Enter the extension number (DN) of the phone and click the **Create** Button.



This phone will now appear in the list.

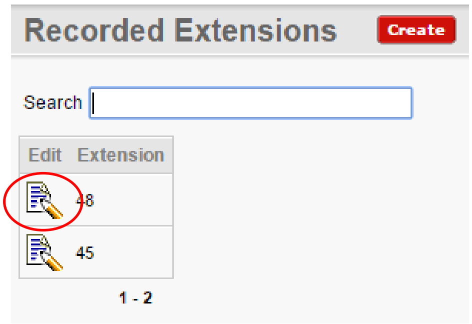
Action Processed.

3. You may continue to add or edit phones in the list within your license limit. Once you reach the limit, the Create button becomes inactive.
4. To start recordings for the updated list, click the **Activate** button

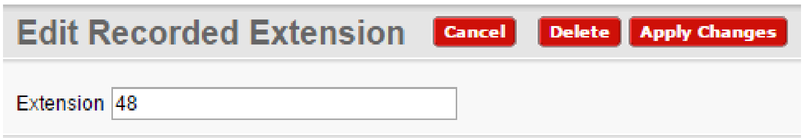


To change or delete a phone from the recording list:

1. From the main **Manage Recorded Extensions** page, click the **edit** icon next to the extension

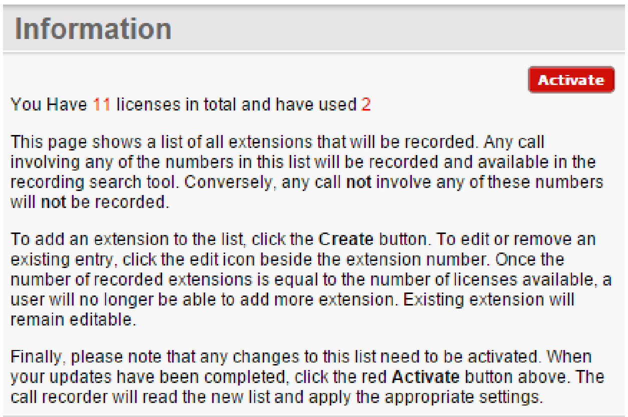


2. Edit the extension number (DN) to change it and click the **Apply Changes** button, or press the **Delete** button to remove the extension from the list.



The list will be updated.

3. Continue making changes within your license limit.
4. To start recordings for the updated list, click the **Activate** button



7.3 Additional user accounts

FTR users have access to the recording application page that allows recordings to be browsed, searched and retrieved.

There are two common methods for defining user accounts. Only one of these will be configured during the initial implementation of the solution.

1. User account defined by implementer or Atea Systems. This is a local account. To add more user accounts, contact Atea support.
2. Users are linked to LDAP/Active-Directory and access is controlled by group membership. This is usually reserved for larger organisations. To add more users, add them to the appropriate AD group.

8 Troubleshooting

Troubleshooting tips and setup tasks.

Issue	Tip
Some recordings are not displaying on the recording page	Possible issues: 1. The date range is incorrect.

Issue	Tip

	2. The recording has not been processed yet. It may take up to ten minutes for a recording to display on-screen
	3. There may be an issue with recordings not being captured.
	    a. Check the Linux directory /var/recordings/ to see if the recording files are present.
	    b. Check the directory numbers for the caller and calling party. If this is a five or six digit number that is high, contact Atea Support regarding a possible UDP port number issue.

A specific phone is not being recorded	Possible issues: 1. The phone must have the recording profile set up in the CUCM 2. The phone number (DN) must be included in the license list.
Recording files are of zero duration	Possible issues: Wrong voice codec / not enough resources FTR Solutions using G.711 only 1. On the UCM, check that the calls are set for G.711 2. If UCM transcoding is used to convert calls to G.711, check there are sufficient transcoding resources. Note: FTR does not record G.722 calls FTR solutions using g.729 option 1. Check that G.729 option is enabled 2. Check that the G.729 license file is present 3. Check that there are sufficient G.729 licenses installed (Atea) Blocked access 1. A firewall is blocking the access to the SIP trunk or to the disk with the recordings. This is a configuration issue that sometimes occurs during initial commissioning.
Recordings display as "not found" on the recording display screen	The recording file may have been moved or archived from the disk. Restore the file to the original location. Alternatively, search for the file name in the location where the recording files have been moved or archived to.
Recording disk space is full / exceeds capacity threshold	Move or archive some recording files.
Oracle database is full / approaching full	Contact Atea support to arrange purging of selected database records

Issue	Tip
Recordings are cut off after 16 minutes	Adjust the "SIP Session Expires Timer" to the value 86400 (which is 24 hours in seconds). The default setting for this means recording automatically ends at 16 minutes, as the RFC4028 re-invite is not used. This timer is a system setting under: Cisco CallManager (Active) > System > Service Parameter Configuration > SIP Session Expires Timer
Some recordings overlap falsely when using G.729 codec	Silence suppression for G.729 needs to be disabled. On the CallManager, set the parameter **Strip G.729 Annex B (Silence Suppression) from Capabilities** to **True**.