# Avaya SBCE SIPREC Configuration for Oreka Audio Recording

This guide describes how to prepare an Avaya Enterprise SBC (ASBCE) for audio recording and/or live streaming by the Oreka software recorder by OrecX, via the SIPREC recording interface on the ASBCE. It assumes that Avaya Communication Manager, Session Manager and SBCE are pre-configured properly and you can make calls from local endpoints to the SIP Trunk through the SBCE.

## Log into the ASBCE web interface

Using a regular web browser, navigate to the following URL:

https://192.168.0.12

Where 192.168.0.12 is the IP address of the ASBCE, and log in using the appropriate credentials.

## Add Server Configuration Profile

Navigate to **Global Profiles -> Server Configuration** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up menu, e.g. "OrekaServer".

Click on **Next** and enter details in the dialogue box:

- In the **Server Type** drop down menu, select **Recording Server**.
- Click on **Add** to enter an IP address
- In the **IP Addresses / FQDN** box, type the Oreka recording server interface address.
- In the **Port** box, enter the port to be used for the listening port configured on the Oreka, e.g. 5060.
- In the **Transport** drop down menu, select **TCP**.

Click on **Next** and Enable **Heartbeat** with method **OPTIONS**

Click on **Next** to configure **Grooming** as follows:

- Enable Grooming which allows us to support multiple connections.

- No interworking profile required for Oreka

# Add Routing Profile for Recording Server

Routing information is required for routing recordings to Oreka. The IP addresses and ports defined here will be used as the destination addresses for signalling.

To define routing to the Oreka SIP Trunk, navigate to **Global Profiles -> Routing** in the main menu on the left hand side. Click on Add and enter an appropriate name in the dialogue box, e.g. “OrekaRoutingProfile”.

Click on **Next** and enter details for the Routing Profile:

- Click on **Add** to specify the IP address for the Oreka SIP trunk.
- Assign a priority in the **Priority / Weight** field. If only a single recording server is used, then choose **1**. If multiple recordings servers are configured for load balancing / auto-failover, then choose priority/weight for each server accordingly.
- Select the recording server defined earlier in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field

# Define Application Rules

An application rules needs to be defined for Oreka. To create a new Application Rules, navigate to **Domain Policies -> Application Rules**. Click on **Add** and enter an appropriate name in the pop-up menu, e.g. “OrekaApplicationRule” and select **Next**.

On the **Application Rule** pop-up windows check **In** and **Out** boxes for **Audio**, and select **Finish**.

# Define Media Rules

Audio formats need to be specified for Oreka. To create a Media Rule for Oreka, navigate to **Domain Policies -> Media Rules**. Click on **Add** and enter an appropriate name in the pop-up menu, e.g “OrekaMediaRule” and select **Next**.

On the **Media Rule** pop-up, under **Audio Encryption**, select a **Preferred Format #1** and select RTP

On the **Media Rule** pop-up, under the **Audio Codec** section, select box for **Codec Prioritization**. For **Preferred Codecs** select **PCMU**, **PCMA** and **telephone-event**, and click **>**.

# Configure UCID

UCID needs to be enabled for **Signaling Rules** that are defined for Session Manager and Oreka. Navigate to **Domain Policies -> Signaling Rules**. Select the policy for Session Manager and select the **UCID** tab. Click **Edit**, check box for **Enabled** and type in a unique value in **Node ID** field. Select **Finish** to save configuration.
Perform similar steps for Oreka signaling rule.

# Define End Point Policy Group

To define an **End Point Policy Group** for Oreka, navigate to **Domain Policies -> End Point Policy Group** and select **Add**. Click on **Add** and enter an appropriate name, e.g. “OrekaPolicyGroup” in the pop-up menu and select Next.

On the **Policy Group** pop-up, select the **Application Rule** defined earlier and select the **Media Rule,** also defined earlier Select **Finish** to save configuration.

# Define Session Policies

To define **Session Policy** for Oreka, navigate to **Domain Policies -> Session Policies** and select **Add**. Click on **Add** and enter an appropriate name in the pop-up menu, e.g. “OrekaSessionPolicy” and select **Next**.

On the **Session Policy** pop-up, select box for **Media Anchoring** and **Recording Server**.
For **Recording Type** select **Full Time**.
For **Routing Profile** select the Routing profile configured earlier.

# Define Session Flows

To define Session Policy for Oreka, navigate to **Device Specific Settings -> Session Flows** and select **Add**. Click on **Add** and enter an appropriate **Flow Name** in the pop-up menu, e.g. “OrekaSessionFlow” and select the **Session Policy** defined earlier. Select **Finish** to save the configuration.

# Define Server Flows

The End Point Server Flows allow calls to be recorded by Oreka when they are passing through Avaya SBCE to the Service Provider’s SIP Trunk. Navigate to **Device Specific Setting -> End Point Flows -> Server Flows**.

Create two Server Flows for Oreka, one to record calls coming in from Service Provider’s SIP Trunking service and another for calls coming in from Session Manager.

## Server flow 1. SIP Trunk -> Session Manager, i.e. inbound calls

Configure:

- For **Server Configuration** select the one created earlier.
- For **Received Interface** select the one used towards Service Provider for SIP Trunk
- For **Signaling Interface** select the one used towards Session Manager for SIP signaling
- For **Media Interface** select the one used towards Session Manager for RTP media
- For **Endpoint Policy Group** select the one created earlier
- For **Routing Profile** select **default** i.e. Routing profile without any IP address
- For **Topology Hiding** select **None**. The SBC will send the same information it sends to SM based on the TH set in the SM server flow.

## Server flow 2. Session Manager -> SIP Trunk, i.e. outbound calls

This server flow is similar to the previous one, but for reverse call direction, outbound call through Service Provider's SIP Trunk. The **Received Interface**, **Signaling Interface** and **Media Interface** are reversed.

Configure:

- For **Server Configuration** select the one created earlier
- For **Received Interface** select the one used towards Session Manager for SIP signaling
- For **Signaling Interface** select the one used towards Service Provider for SIP signaling
- For **Media Interface** select the one used towards Service Provider for RTP media
- For **Endpoint Policy Group** select the one created earlier
- For **Routing Profile** select **default** i.e. Routing profile without any IP address
- For **Topology Hiding** select **None**. The SBC will send the same information it sends to SM based on the TH set in the SM server flow.

### Server flow 3. Remote worker calls.

If Remote Workers connect to SBCE via external network interface different from the Service Provider's SIP Trunk, then an additional server flow is required to record Remote Worker calls:

- For **Server Configuration** select the one created earlier
- For **Received Interface** select the one used towards Remote Workers
- For **Signaling Interface** select the one used towards Session Manager for SIP signaling
- For **Media Interface** select the one used towards Session Manager for RTP media
- For **Endpoint Policy Group** select the one created earlier
- For **Routing Profile** select **default** i.e. Routing profile without any IP address
- For **Topology Hiding** select **None**. The SBC will send the same information it sends to SM based on the TH set in the SM server flow.

## Configure Oreka OrkAudio for ASBCE SIPREC recording

Enable sipua plugin and disable the default plugin:

*# vi /etc/orkaudio/config.xml:*

Ensure it looks like this:

*<!--<CapturePlugin>libvoip.so</CapturePlugin>-->*
*<CapturePlugin>liborksipua.so</CapturePlugin>*

Add section, if not already present:

*<SipUAPlugin>*
    *<SipMode>SiprecAcme</SipMode>*

*<SdpOfferAnswerMode>true</SdpOfferAnswerMode>*
*<SupportFeatures>resource-priority,siprec</SupportFeatures>*
*</SipUAPlugin>*

Then restart orkaudio

# Configure Avaya AES/TSAPI integration

Avaya AES/TSAPI can be used to enrich the metadata as well as ensure correct segmentation of recordings, so that there is one recording per intervening agent.
This is described in a separate document. Please contact support@orecx.com about this.