OrecX
Open Source Recording

Cisco BIB

Recording via Cisco BIB is an "active" recording method that does not use port mirroring. When configured on the Cisco platform, the recorder is SIP INVITEd by the CallManager and the RTP is sent directly from the phone's BIB (Built In Bridge) to the recorder.

Prerequisites

1 - The telsets need to be compatible with BIB. Most of them are, but some older phones such as the 7940/7960 are not.

Full details: https://developer.cisco.com/site/uc-manager-sip/faq/supported/

2 – Target recording server needs to be configured with **CentOS 7 64-bit**

Orkaudio configuration

OrkAudio Addons minimal version: orkaudio-addons-2.11-6061.
For earlier versions, SdpOfferAnswerMode needs to be false.

1. Configure orkaudio to use the right plugin in the orkaudio config.xml file (comment the existing plugin):

Ensure it looks like this:

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

2. Please add the following node under the top node, if it does not already exist:

```
<SipUAPlugin>
    <SipMode>CiscoBib</SipMode>
    <SdpOfferAnswerMode>true</SdpOfferAnswerMode>
</SipUAPlugin>
```

3. Restart orkaudio.

At that point, orkaudio will be ready, waiting for incoming SIP traffic on port 5060.

By default, the contact field is populated by sofia (possibly taken from the SIPREC INVITE request URI?) and we do not control which local IP is chosen. It is also possible to force the content of the Contact field with a static DNS name or an IP address:

```
<Contact>server.domain.com</Contact>
```

CUCM configuration

You need to configure your Call Manager for BIB recording and point it to the IP address of our recorder, see this:

CUCM11.5
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/featureConfig/CUCM_BK_C7DC69D3_00_cucm-feature-configuration-guide_115/CUCM_BK_C7DC69D3_00_cucm-feature-configuration-guide_115_chapter_01011.html

CUCM 11.0
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_0_1/featureConfig/CUCM_BK_FE5123E0_00_cucm-feature-configuration-guide_1101/CUCM_BK_FE5123E0_00_cucm-feature-configuration-guide_1101_chapter_01010.html

CUCM 10.5
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_1052/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_chapter_01010.html

CUCM 10.0
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_0101010.html

CUCM 9:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmfeat/CUCM_BK_CEF0C471_00_cucm-features-services-guide-90/CUCM_BK_CEF0C471_00_cucm-features-and-services-guide_chapter_0100110.html

Configure also the sip trunk with “Media Termination Point Required”

This will extract the local agent extension (as opposed to telset extension) and remote phone number as metadata, but call direction is not reported.

For call direction and more metadata, CTI integration is required.