OAISYS SIP Trunk Integration

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OVERVIEW

OAISYS introduces the ability to record calls that originate on a SIP trunk with version 6.1. The Session Initiation Protocol (SIP) is a signaling protocol used to establish sessions for an IP network.

NOTE: OAISYS 6.1 SIP Trunk Integration will not include SMDR/CTI Integration.

For call recording purposes, a SIP session on a trunk refers to telephone calls. The OAISYS solution integrates directly with SIP trunks to record calls by capturing call data from the SIP trunk. OAISYS Trunk-side recording can record audio from T1 trunks, PRI trunks, Analog trunks, and SIP trunks.

Version 6.2 of the Recording Server Software introduces recording of SIP trunks with Matching Logic. SMDR Matching Logic can be used with Mitel 3300, Mitel 5000, Toshiba CIX, or ShoreTel to provide extension information and account codes to the OAISYS recording server.

Matching Logic is not 100% accurate, but provides a close match to the criteria entered. For example: if two calls took place at 10:23:35, lasting 30 seconds to the same outside phone number, OAISYS Matching Logic could not make a match. If a call cannot be distinctly matched, no extra information will be attached to any call. Whereas, recording TDM Trunks with CTI is accurate all the time.

Recording SIP Trunks with Matching Logic differs from recording traditional T1 or PRI trunks with CTI integration.

SMDR Matching Logic on PBXs supporting multiple state transitions

1. ***ALL EXTENSIONS*** involved with the call will be attached to the call moments after it is complete

   NOTE: The Mitel 3300 and Toshiba CIX support multiple state transitions

On other PBXs

2. ***THE LAST EXTENSION*** involved with the call will be attached to the call shortly after the call is complete

When recording on TDM trunks with CTI

3. ***ALL EXTENSIONS*** involved with a call are attached to the call record

The criteria that can be used for searching records and establishing permissions differ between SIP Trunk with Matching Logic and the TDM Trunk with CTI. See the comparison chart below:

Feature	TDM Trunk with CTI	SIP Trunk with Matching Logic
Station Information	☑	Only after the call is complete
Account Code	☑	Only after the call is complete
Start Date & Time	☑	☑
Call Duration	☑	☑
Call Direction	☑	☑
Manual Start/Stop Recording	☑	
Caller ID	☑	☑
DNIS	☑	☑
ACD Agent	☑	
ACD Group	☑	
Extra Call Information	☑	Only after the call is complete
After Call Actions	☑	
Live Call Monitoring	☑	No extension info on live calls
Screen Recording Option	☑	
Desktop Client Application	☑	

REQUIREMENTS

- OAISYS Software Version 6.1 or later
- One call on a SIP trunk at one time
  - One voice port required per call on a SIP trunk
- Network Switch with Port Mirroring
  - Recommended configuration: two destination ports
- AudioCodes USB Dongle
- AudioCodes HPX License
  - One per port
- AudioCodes driver 5.3 required.
  - Download from this location:

    ftp://ftp.oaisys.com/pub/downloads/3rdparty/Ai-Logix/5.3/

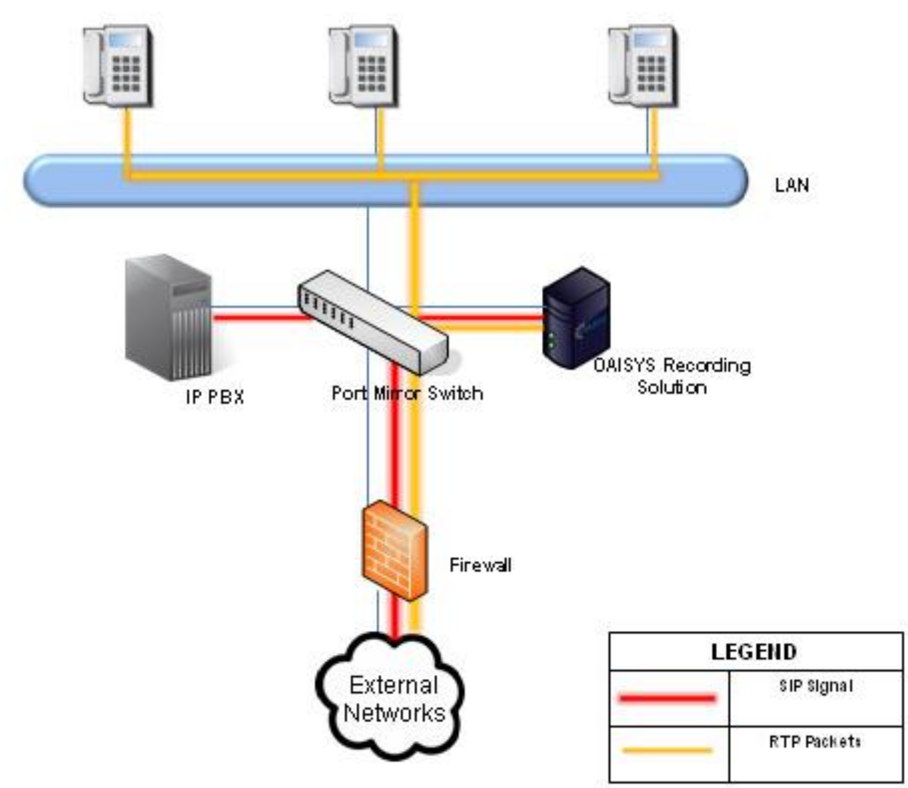
SUPPORTED PBXS

OAISYS supports recording SIP Trunk Recording for the following PBXs:

- Mitel 3300
- Mitel 5000
- Toshiba CIX
- ShoreTel
  - Contact OAISYS Product Manager at linda_gregg@OAISYS.com.

For PBXs not listed, please contact OAISYS Sales Engineering at SE@OAISYS.com.

SIP TRUNK INTEGRATION DIAGRAM



EXPECTATIONS

The information available to the OAISYS solution when recording the SIP Trunk:

- Start Date and Time
- Call Duration
- Call Direction
- ANI/DNIS (if provided by the service provider)

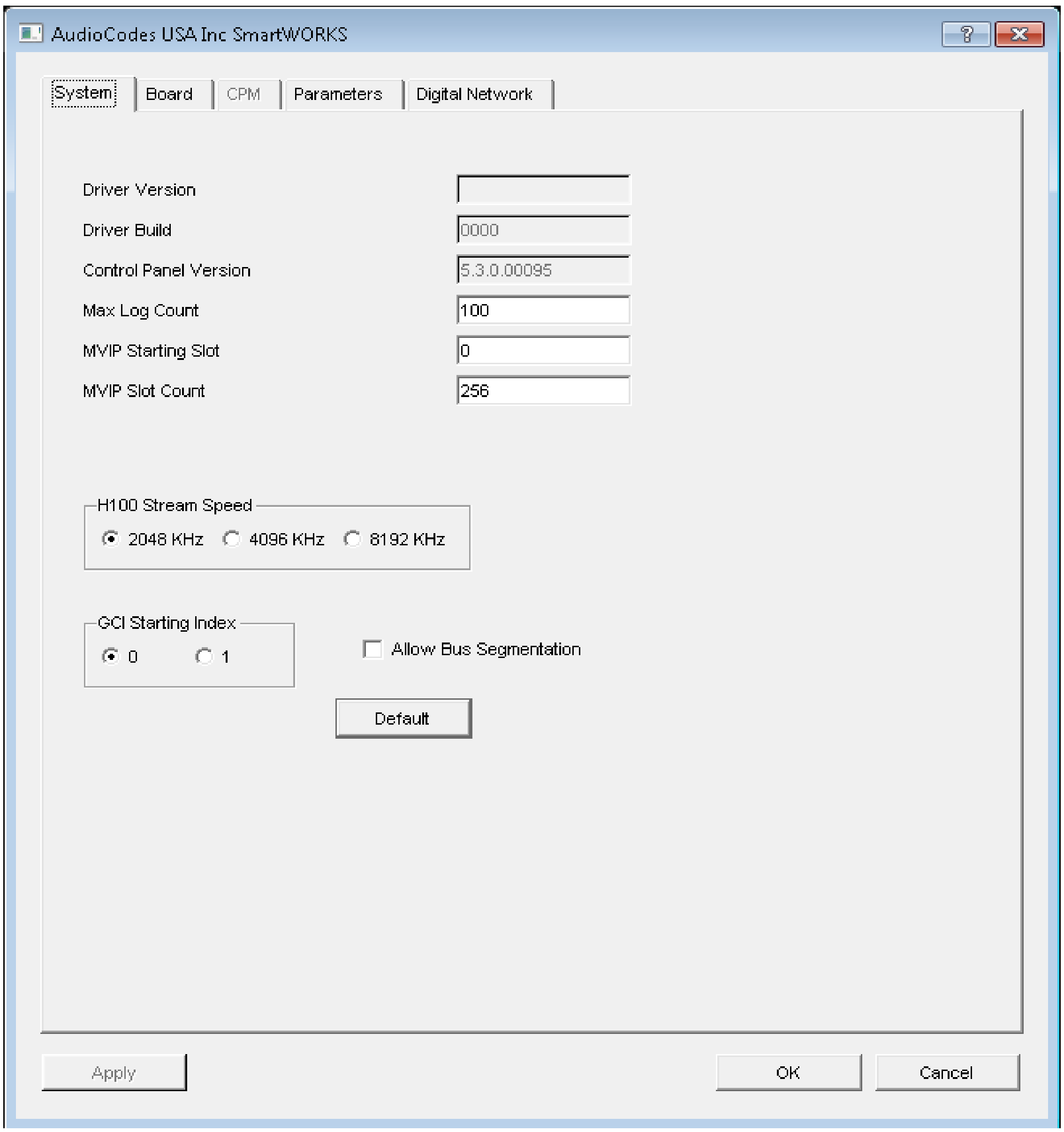
This information can be used to search for calls and can be used to enable specific permissions.

NOTE: IC calls or Peer-to-Peer calls are not recorded when using SIP Trunk Integration.

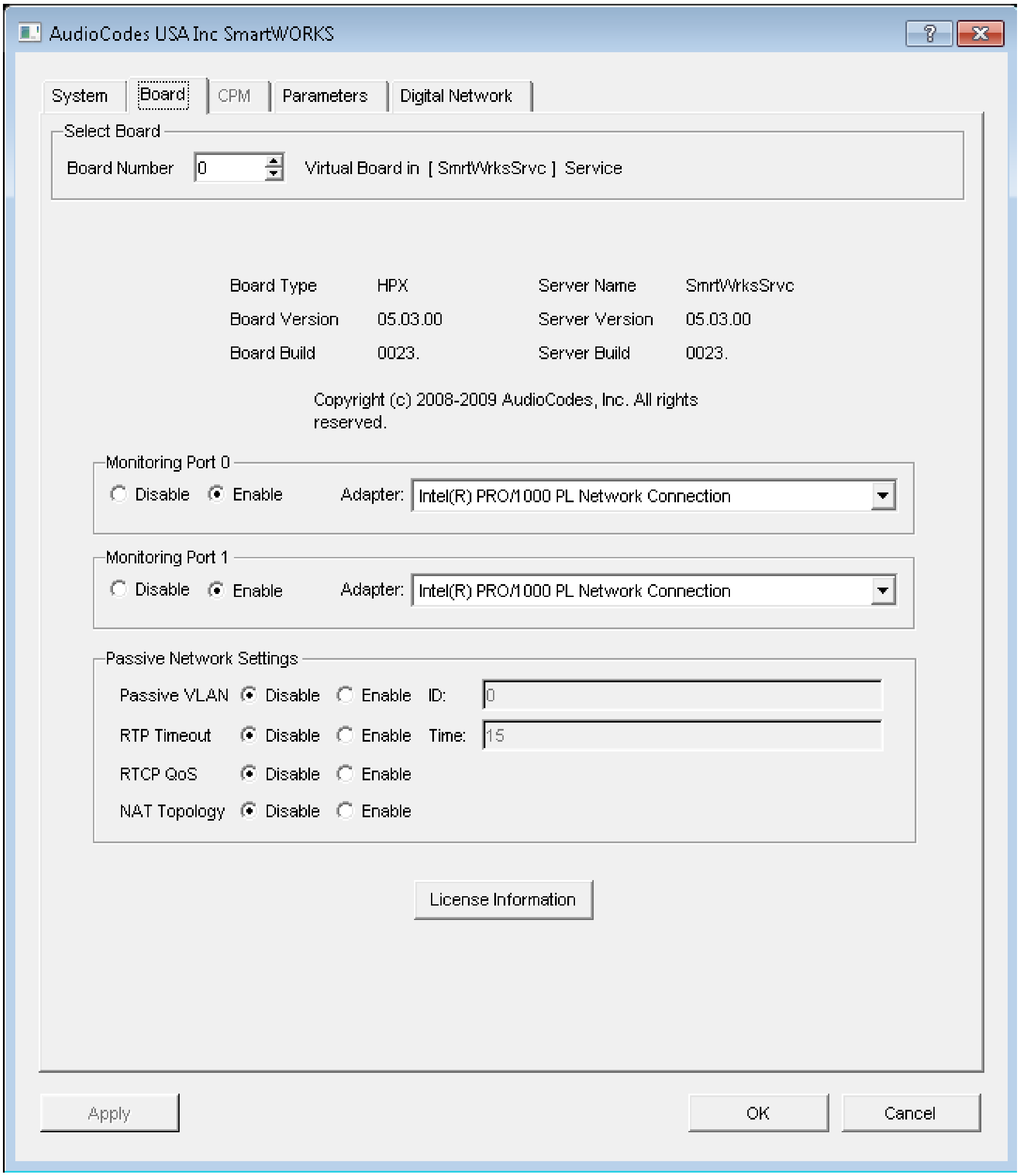
CONFIGURATION

The following information describes how to apply the AudioCodes license files and configure the OAISYS solution to record audio on SIP trunks. To use port mirroring, we recommend using a network switch that supports two destinations. The information in this guide assumes two network cards are used in the OAISYS system to separate RTP (OAISYS) traffic from SIP traffic.

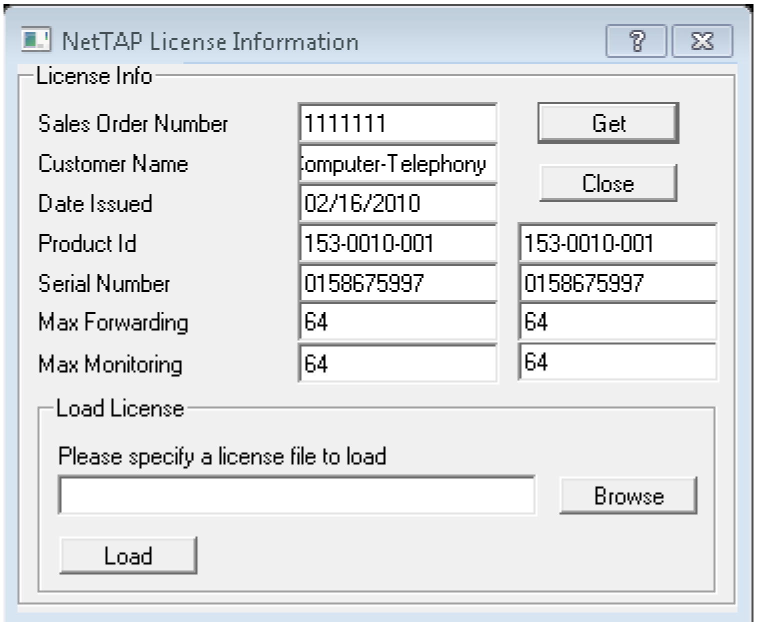
1. Open AudioCode Smart Control through the control panel.



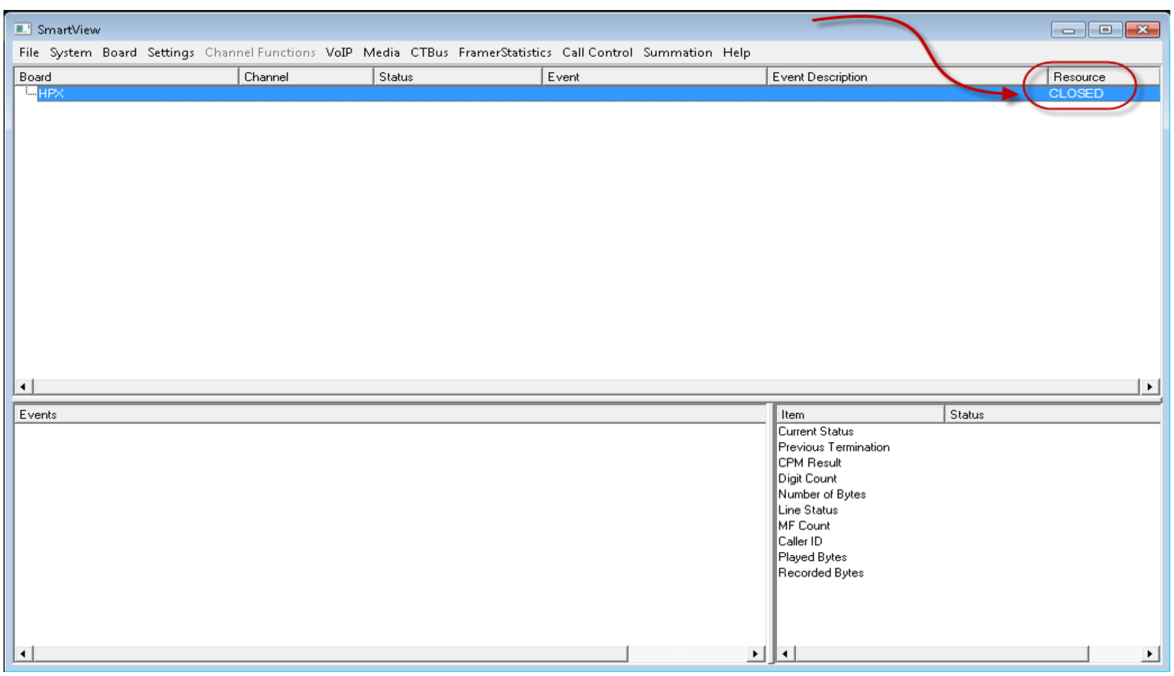
2. Smart Control Board Tab view of HPX virtual board



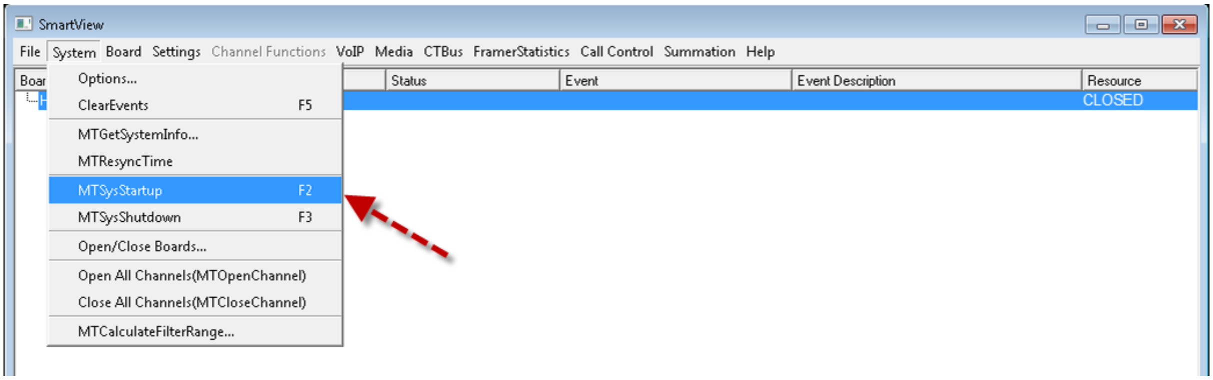
3. View of license information window



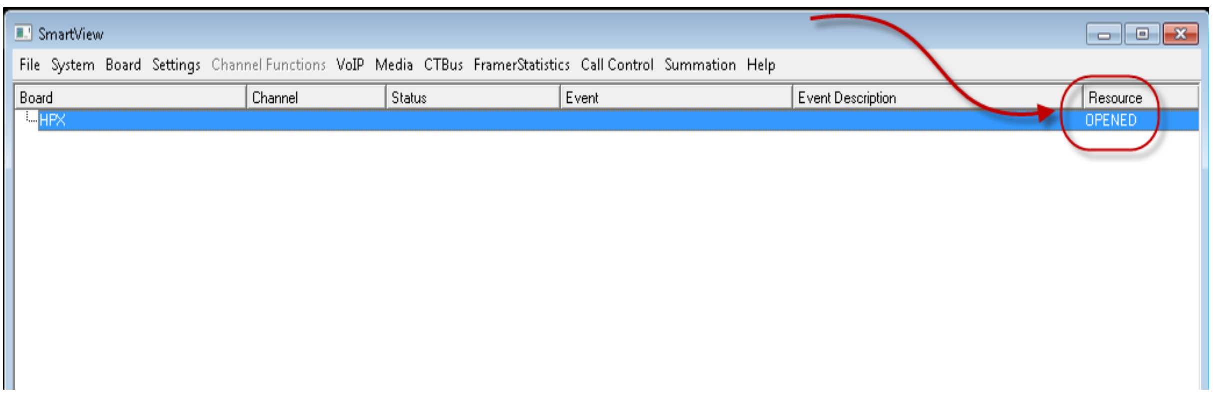
4. Next, you will need to enable UDP port 5060 for SIP, to do this:
   a. Open AudioCodes Smart View → the board will indicate "CLOSED"



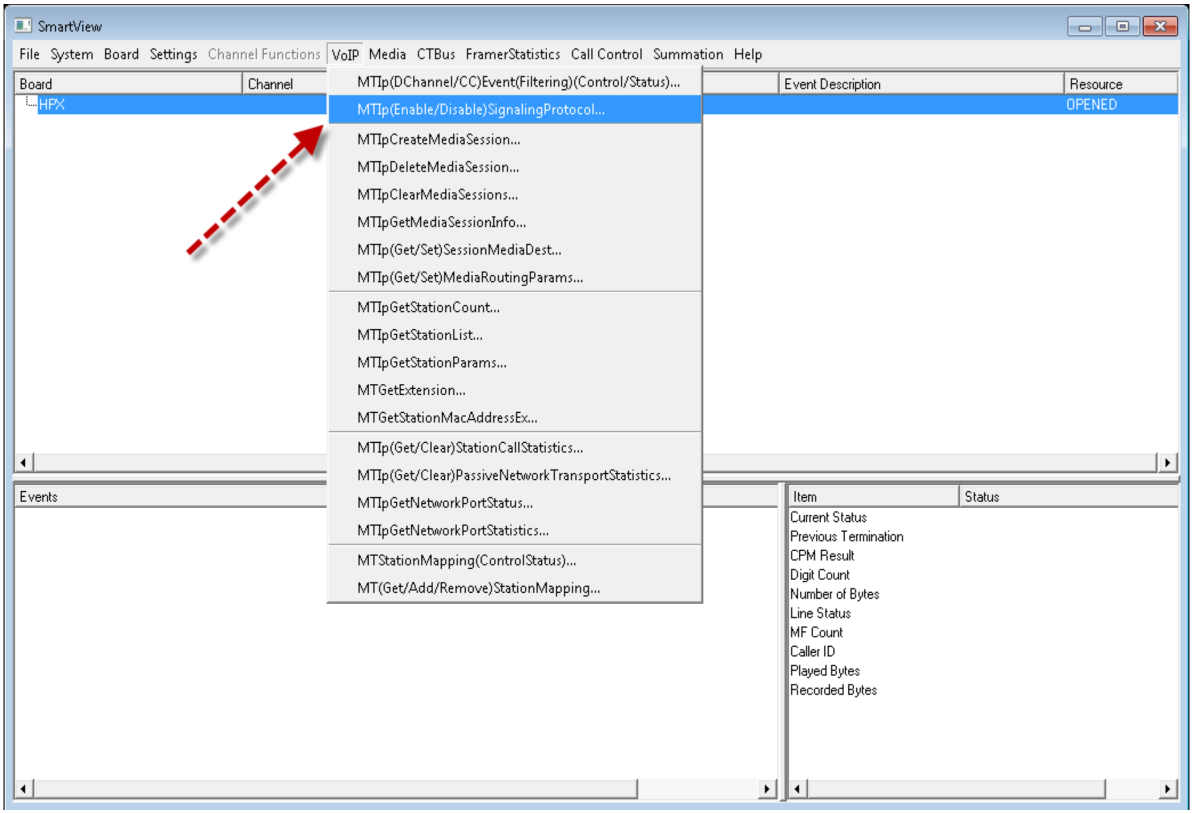
b. Open the board



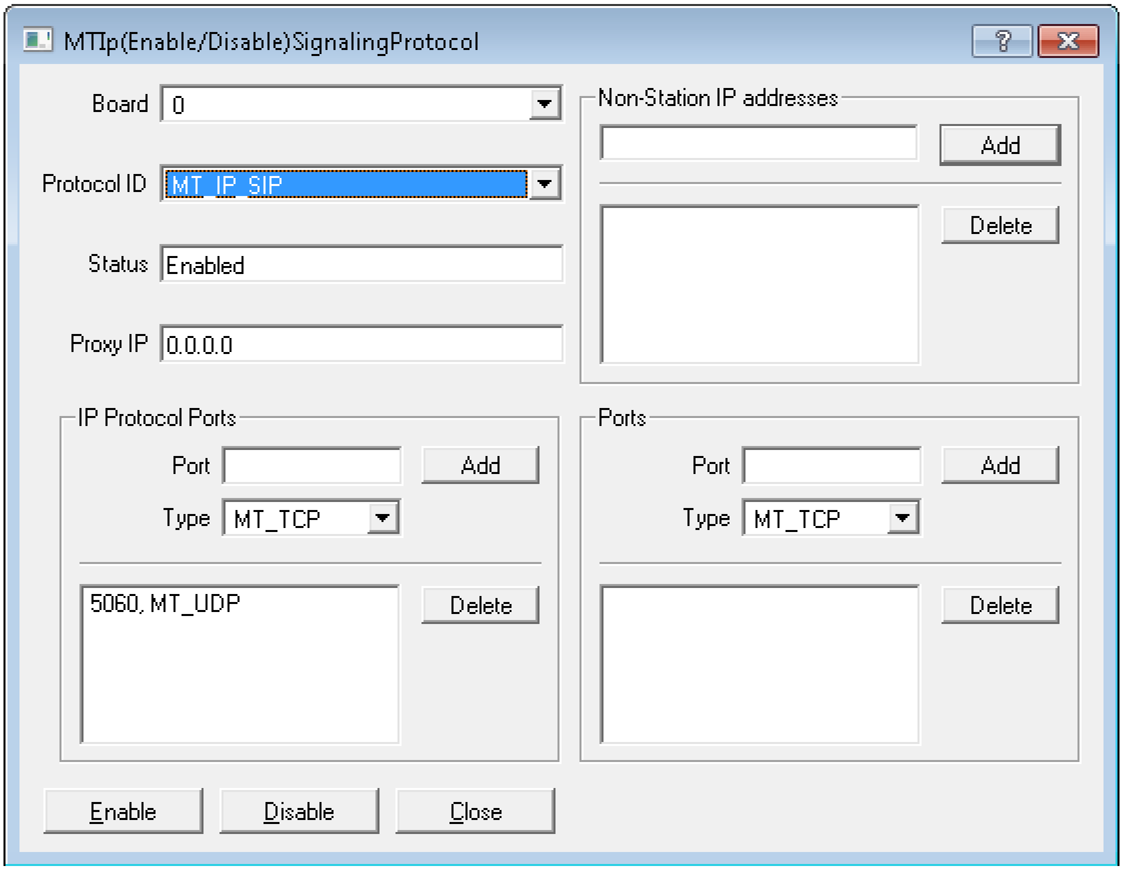
c. This shows the board in “OPEN” state



d. Open the Signaling Protocol window

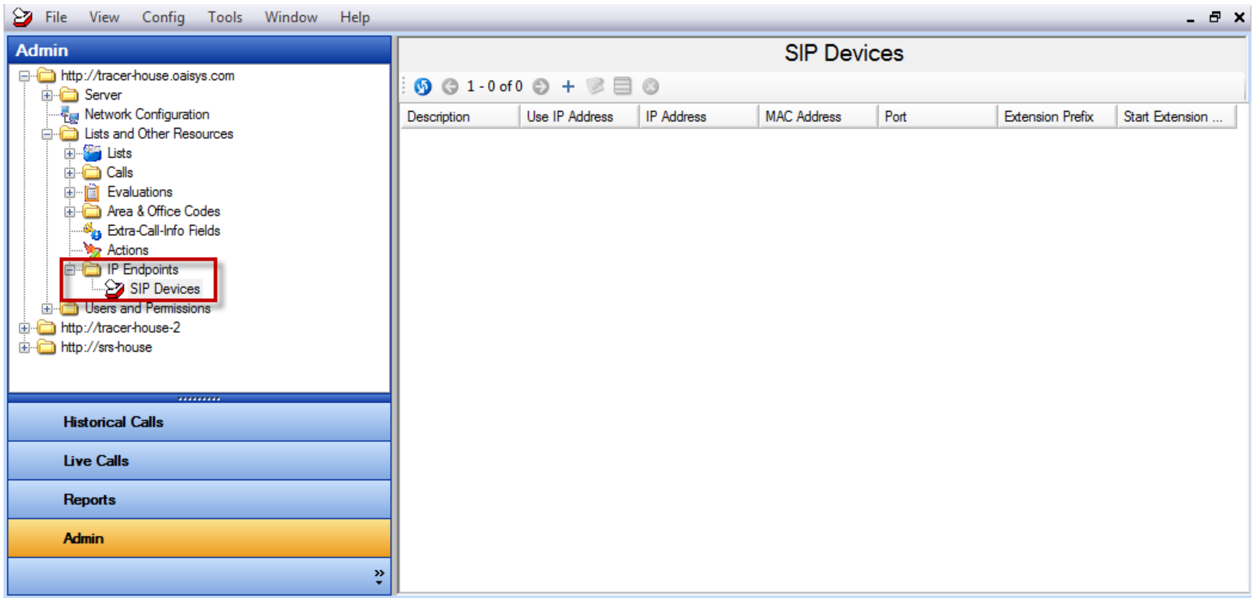


e. Enable UDP port 5060 for SIP

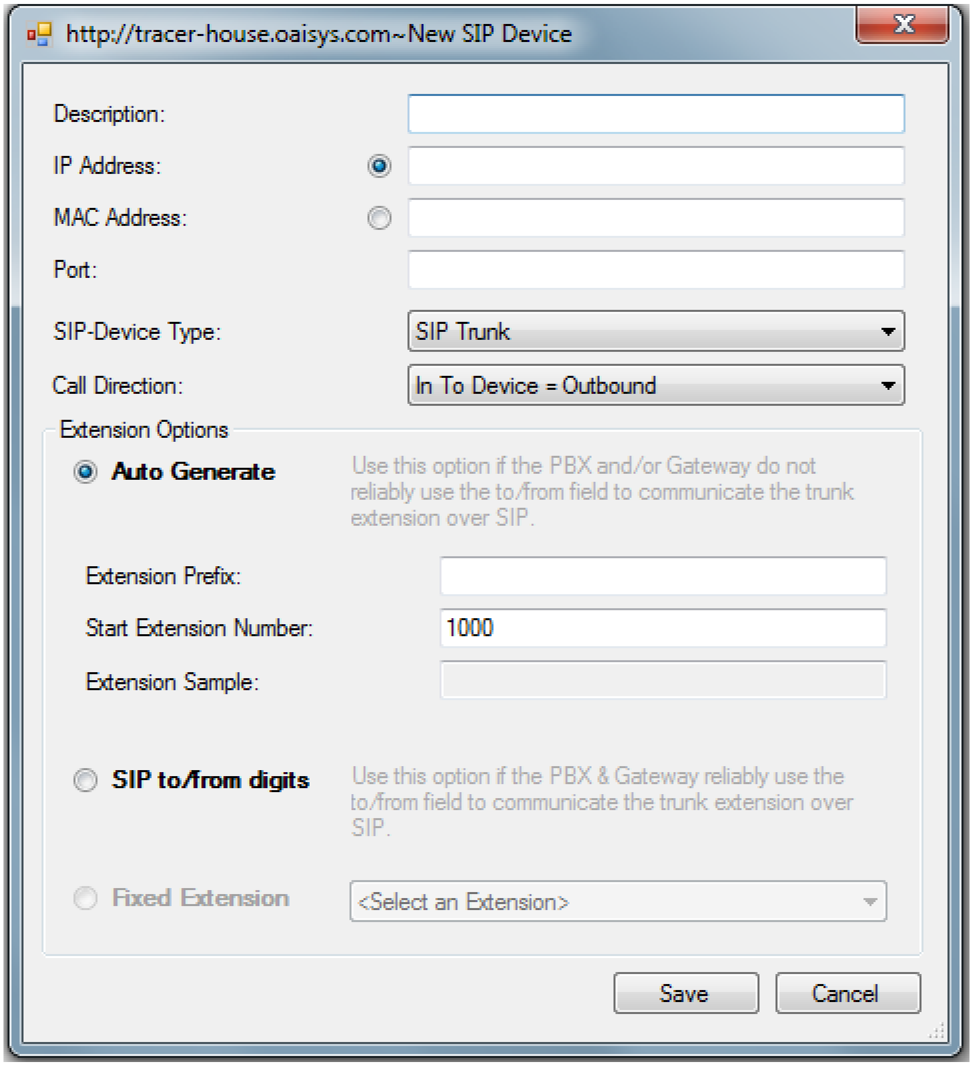


5. Open OAISYS Management Studio

6. From the Admin Tab, navigate to Lists and Other Resources → IP Endpoints → SIP Devices → add new SIP trunk settings

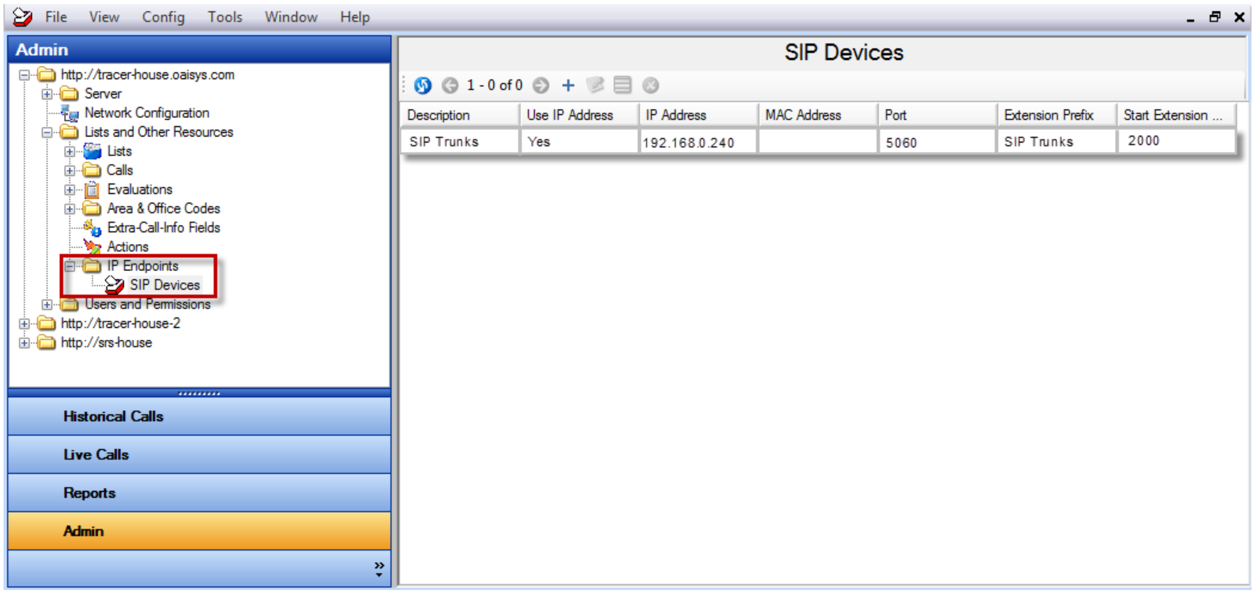


7. Click on **Add New (+)** to display the following



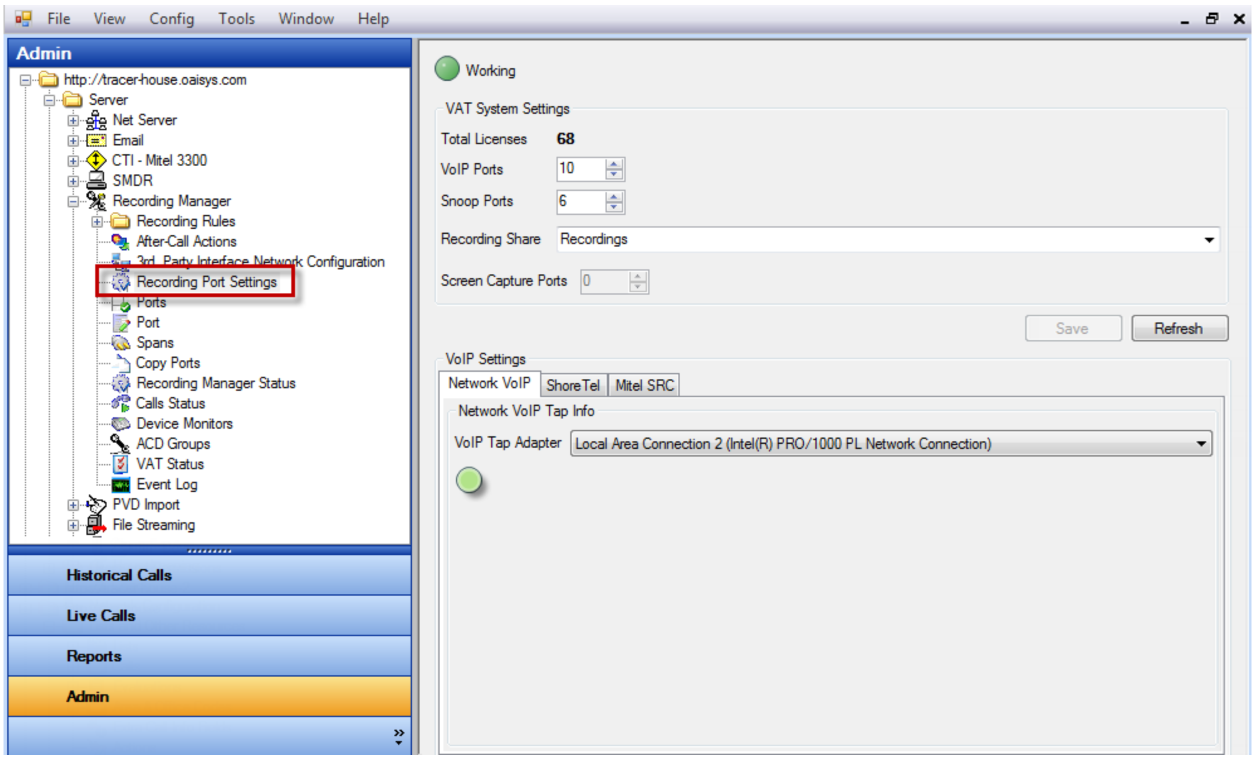
a. Enter a description
b. Enter the IP Address of the SIP Provider OR the IP Address of the Edge Device (such as the router's internal address)
c. Enter the SIP port number (default value is 5060)
d. Select Auto Generate
e. SIP to/from digits **use this only if recording SIP Trunks on a Mitel 5000**

8. The newly added SIP Trunk information will appear as follows

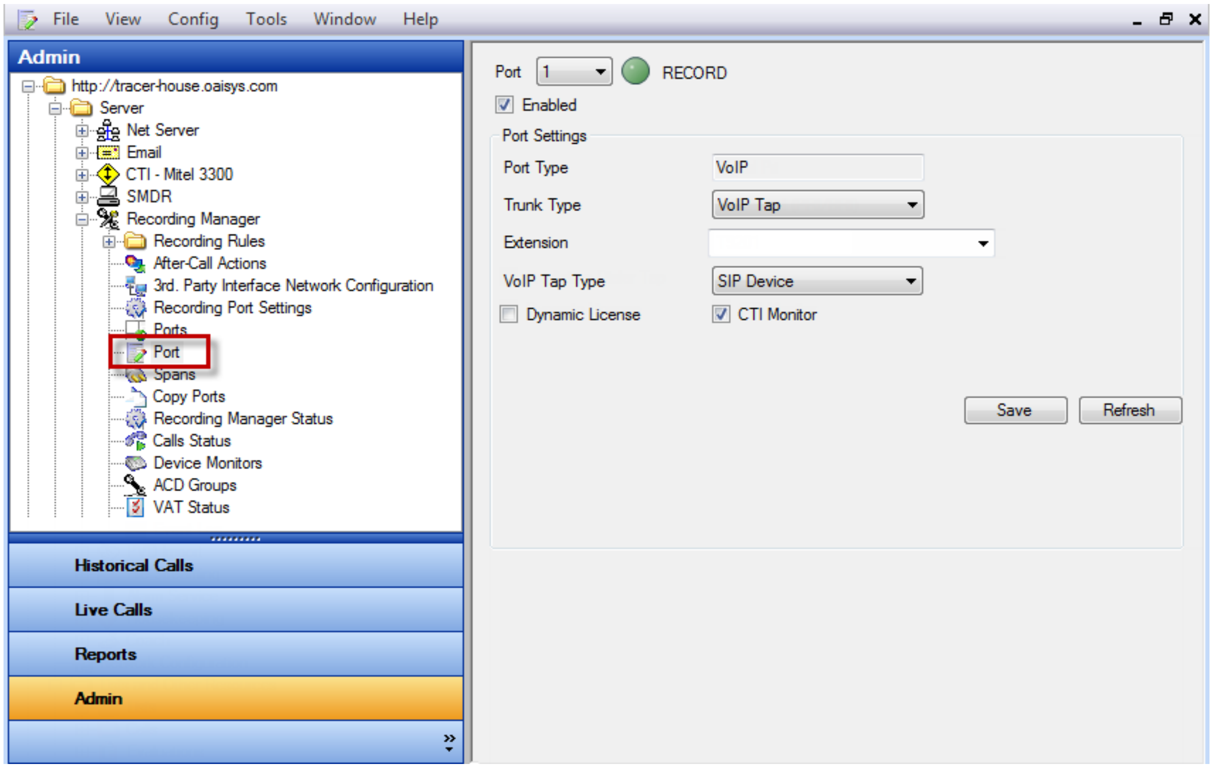


9. Add VoIP ports and select the adapter

NOTE: Per our recommendation of a second mirror port, select the network card for the RTP traffic (if there is only one network card, select it here)

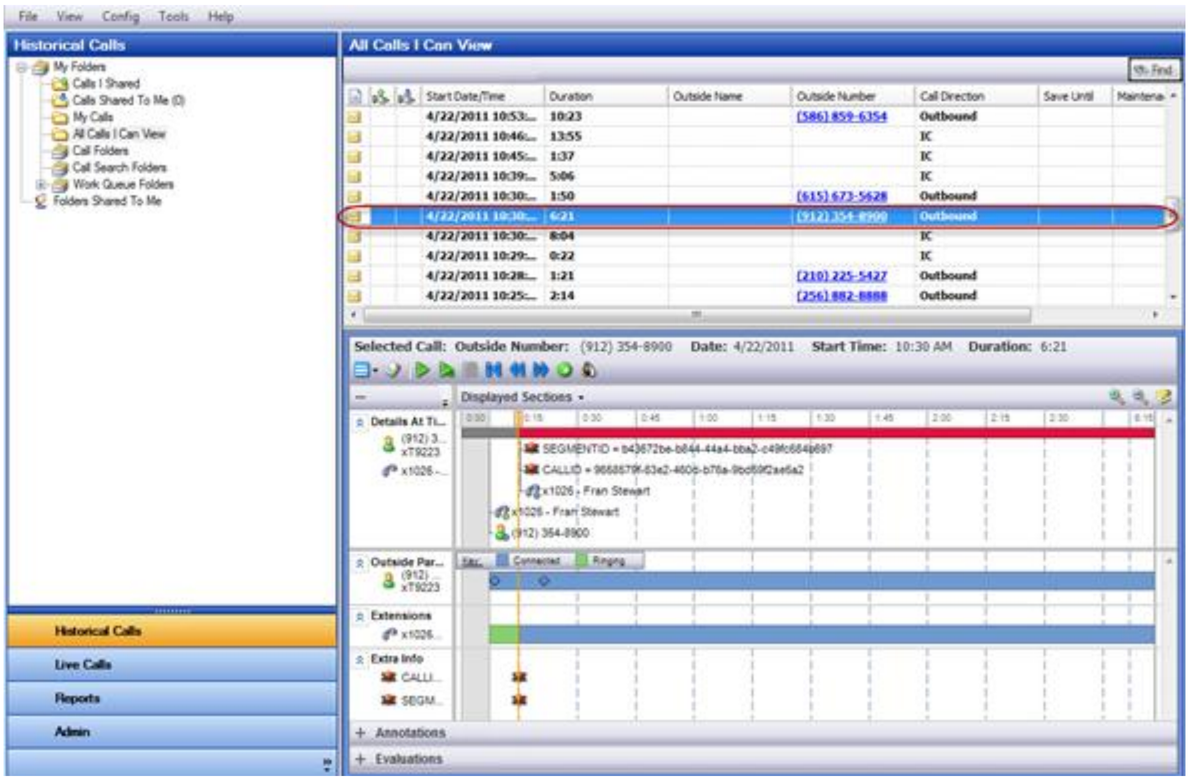


10. Configure the port



SIP CALL

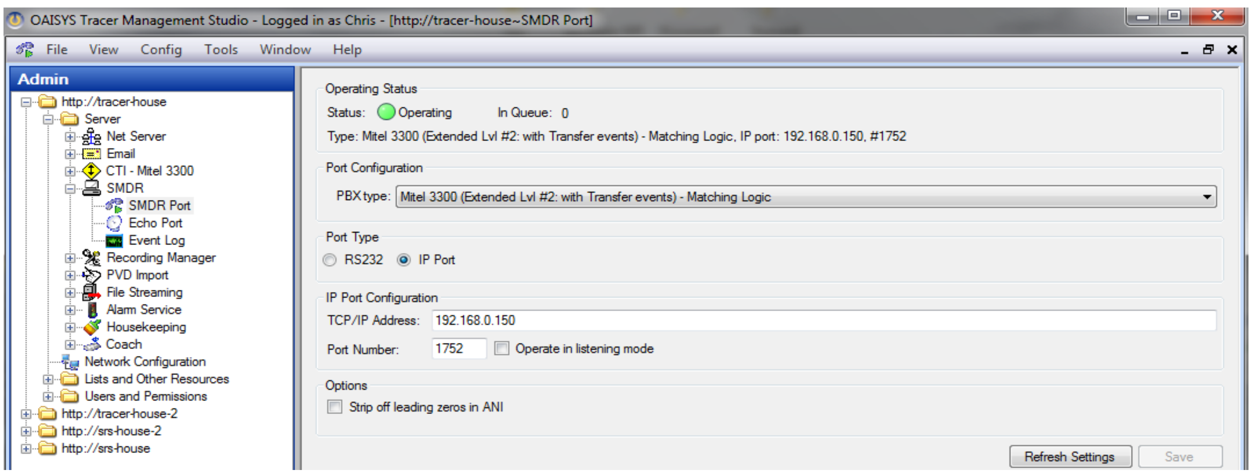
The following image shows how a SIP call appears in the OAISYS Management Studio.



SETUP MATCHING LOGIC

This portion of the document covers the basic setup of an OAISYS Recording Server that has already been configured to record SIP trunks. This assumes the server is already recording audio on the SIP channels, and it is now time to setup the Matching Logic to get extension information on those calls.

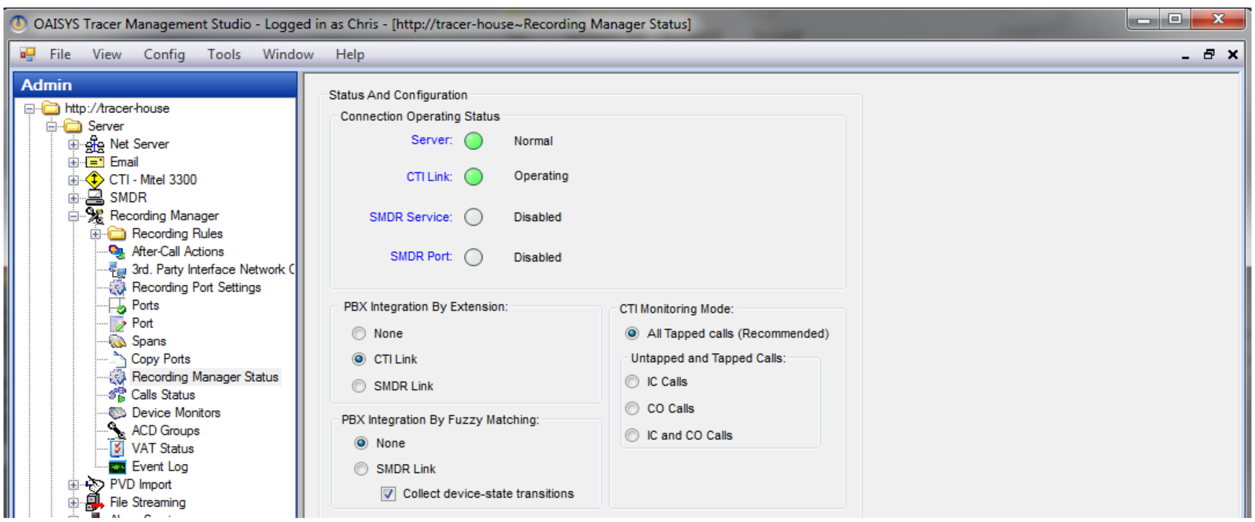
1. To configure, associate SMDR Service with the PBX type (a Mitel 3300 Matching Logic .DEF file is selected in the screen shot below), verify selection of the .DEF file that has “Matching Logic” in the title for your PBX selection.



2. Expand Recording Manager → select Recording Manager Status.

   This section is to verify that if you are using SIP Trunk only (no other recording method), you disable CTI by choosing **None** for PBX integration by extension.

3. Select PBX Integration by Fuzzy Matching (soon to be changed to Matching Logic) to SMDR link.
4. All Mitel 3300 and CTX systems typically support Device State Transitions (multiple SMDR per call) so check this box. This ensures there is only one SMDR event per call (last known extension on the call).
5. The Mitel 5000 does not support Device State Transitions.



6. Make a few test calls to ensure the extension is bound to the call recording.

   Once a call is complete, we see SMDR from the PBX and place it into an event queue. Approximately 30 seconds later, the system will run a database query to determine if any calls match the criteria based on the SMDR event to match to the call.

   - If a match is found, another query is run to add the information to the call.
   - If a match is not found initially, you will see:

     [88204 07:34:49.7] [INFO]Fuzzy match failed for SMDR call data 2527 in FuzzyMatchCallQueue 2; reason = No matches found" in the TRM events

     - The system will run another attempt after 75 seconds; this is additional time allotted for the call to complete and be entered into the database.

- The query is run three (3) times: 30 seconds, 75 seconds, and 30 minutes.

In some cases, the default hard-coded values in the timer settings need to be changed. Below are some example settings we have found are a good match:

This is to set the seconds before a call starts and after a call ends in the start time and duration window to run the query:

HKLM\Software\Computer Telephony Solutions\Recording Manager

DWORD: Voice4NetDurationWindow(20)seconds

DWORD: Voice4NetStartTimeWindow(120)seconds

These are the fuzzymatchqueue lookup timers (in seconds post call completion, so 75 seconds, if no match, we run 15 seconds later, and if no match, 69 seconds as a final attempt):

HKLM\Software\Computer Telephony Solutions\Recording Manager\FuzzyMatching

DWORD: Queue0DurationSeconds (75)

DWORD: Queue1DurationSeconds (15)

DWORD: Queue2DurationSeconds(69)

For further information or assistance, please contact Technical Support at 888-496-9040, option 4!