Mitel MiVoice Business

Integration Guide

Encore Workforce Optimization Solution
Version 7.0 or later

November 15, 2018

For Dealer and Customer Use Only

DVSAnalytics, Inc.
17255 N. 82nd Street, Suite 120, Scottsdale, AZ 85255
Phone: (480) 538-2020 • Fax: (480) 538-2021
Email: info@DVSAnalytics.com
Web: www.DVSAnalytics.com

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Introduction

The Encore system integrates with the Mitel MiVoice Business platform. This integration allows the Encore system to successfully perform the following functions:

- Audio Collection – Capture the audio that needs to be recorded.
- Recording Control – Receive the necessary events that signal when the Encore must start and stop recording.
- Data Capture – Receive data associated with the call.

The Encore system can record calls on the Mitel MiVoice Business platform without using MiTAI but the recording controls and data capture are limited; configuration for this integration is not covered in this document.

Supported Data Capture

The following is a list of supported data elements that can be collected with each recording. Not every element is applicable for each call. For example, if the call being recorded is an internal call, the Trunk ID field is blank. For a description of each data element, refer to "Appendix 1: Glossary" on page 14.

- ACD Name
- ACD Number
- Agent ID
- ANI
- Call Direction
- Call ID
- Call Type
- Consultation Call
- DNIS
- Extension
- Hold Duration
- Hunt Group
- Other Party Name
- Other Party Number
- PFCallID*
- Recorded Party Name
- Recorded Party Number
- Trunk ID

**Only captured if using the PrairieFyre ACD and Reporting package. For more information about PrairieFyre, see your Mitel dealer.*

Supported Recording Features

The following matrix should be used to determine which audio collection is best for your business needs. If you find that more than one collection method will work for you, talk to your Encore representative about which method is more cost-effective. For a description of each feature, refer to "Appendix 1: Glossary" on page 14.

RECORDING FEATURE	AUDIO COLLECTION METHOD				
	STATION-SIDE TDM	TRUNK-SIDE TDM	STATION-SIDE RTP PACKET CAPTURE (PASSIVE INTERFACE)	SUBSCRIPTION-BASED SRC AUDIO STREAM [1]	TRUNK-SIDE SIP PACKET CAPTURE
Max. Recording Ports per Server [2]	192	288	400	400	288
Record External Calls	YES	YES	YES	YES	YES

RECORDING FEATURE	AUDIO COLLECTION METHOD				
	STATION-SIDE TDM	TRUNK-SIDE TDM	STATION-SIDE RTP PACKET CAPTURE (PASSIVE INTERFACE)	SUBSCRIPTION-BASED SRC AUDIO STREAM [1]	TRUNK-SIDE SIP PACKET CAPTURE
Record Internal Calls	YES		YES	YES	
Record Encrypted Calls		YES		YES	YES [3]
Related Call Lookup	YES	YES	YES	YES	
Pause/Resume on Hold	YES	YES	YES	YES	
Beep Tones				YES	
Hot Desk		YES		YES	YES
Key System/MultiCall Groups	YES	YES	YES	YES	YES

1. *For SRC, only phones registered in the SRC are recorded. This method requires licenses to be purchased from Mitel.*
2. *Small Business Servers are limited to 72 ports.*
3. *If the encryption occurs at the station, Encore can record the encrypted calls. If the encryption occurs at the trunk Encore cannot record encrypted calls.*

Software and Hardware Requirements

SYSTEM	SOFTWARE REQUIREMENTS

Mitel system	• All audio collection methods
	  ○ Mitel MiVoice Business v7.0 or later
	    – No special requirements
	  ○ Mitel Communications Director (MCD) v4.x to v6.x
	    – A MiTAI/TAPI Computer Integration license
	• Subscription-based SRC Audio Stream
	  ○ Mitel Secure Recording Connector (SRC) v1.0.27 or later
	  ○ Mitel Standard Linux v7.0 or later
	  ○ 1 SRC tap license is required for every phone that needs to be simultaneously recorded. For instance, if a customer has 20 phones that need to be recorded but they are split into two shifts so that 10 phones are used during the day and 10 phones are used during the evening, the maximum simultaneous recordings would be 10, resulting in the need for 10 SRC tap licenses.

Encore system	• No specific requirements

SYSTEM	HARDWARE REQUIREMENTS

Mitel system	• Station-side TDM
	  ○ Must allow Encore to tap at the punchdown block
	• Trunk-side TDM
	  ○ Must allow Encore to tap at the trunk
	• Station-side RTP Packet Capture
	  ○ Span port on network to route all RTP traffic for recorded stations to Encore server
	  ○ DHCP IP address reservation or static IP assignment for each station to be recorded
	• Subscription-based SRC Audio Stream
	  ○ A separate SRC server
	• Trunk-side SIP Packet Capture
	  ○ Span port on network to route all SIP and RTP traffic to Encore server
	  ○ Session Border Controller

Encore system	• Station-side TDM
	  ○ AudioCodes NGX PCIe card
	• Trunk-side TDM
	  ○ AudioCodes DP PCIe card

Documentation Overview

This document provides integration information for a specific phone system. It helps a user to understand the features and benefits of the integration as well as what needs to be configured on the phone system. Conventions used in this guide include:

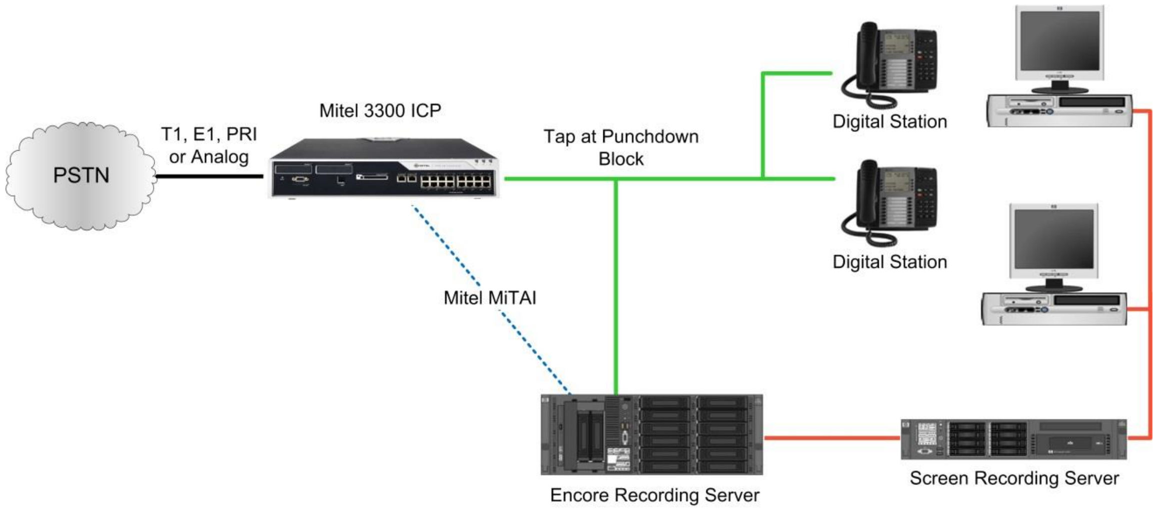
1. Computer commands needed to complete a task appear like this: **Sample** (in black)
2. Keyboard strokes that need to be entered appear like this: `[Sample]`

Overview

This section provides an overview of each audio collection method. For simplicity sake, the diagrams only display a single Encore server but there can be multiple Encore servers depending on the number of stations to be recorded.

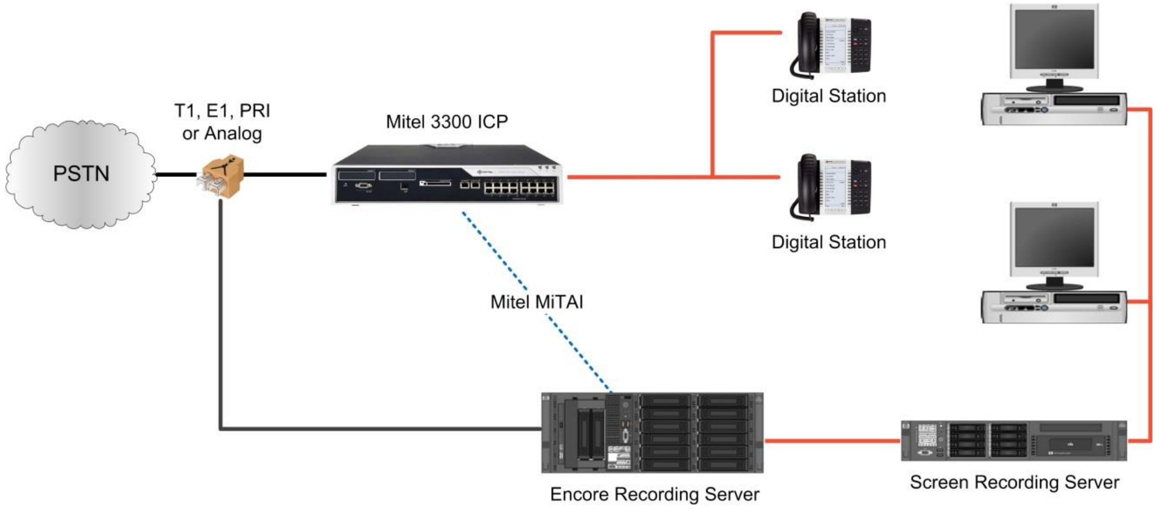
Station-side TDM

The Station-side TDM method uses a passive tap on the phones connected to the Mitel 3300 ICP to collect audio. These may be analog or digital phone sets. This passive tap is connected to the recording boards in the Encore server. Based on events received from the recording boards or from events received from MiTAI, the Encore server collects the audio on the recording boards and associates data with the call from the MiTAI messages.



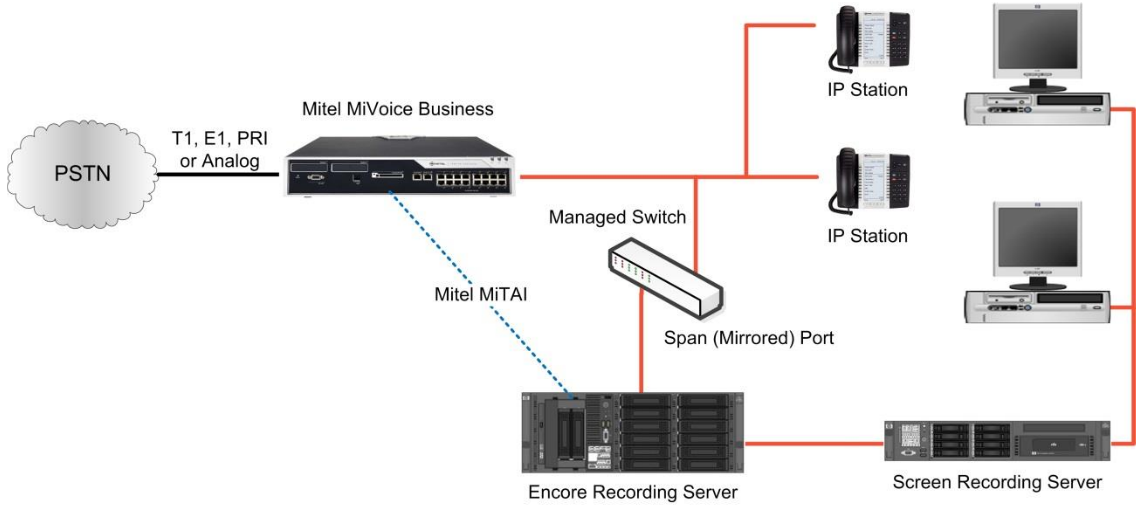
Trunk-side TDM

The Trunk-side TDM method uses a passive tap on the telephony trunks that connect the Mitel 3300 ICP PBX to the PSTN. The trunks can be T1, E1, or Analog. This passive tap is connected to the recording boards in the Encore server. The audio is collected via the passive tap. Based on events received from the recording boards or from events received from MiTAI, call recordings can be started or stopped, and the data associated with the call record.



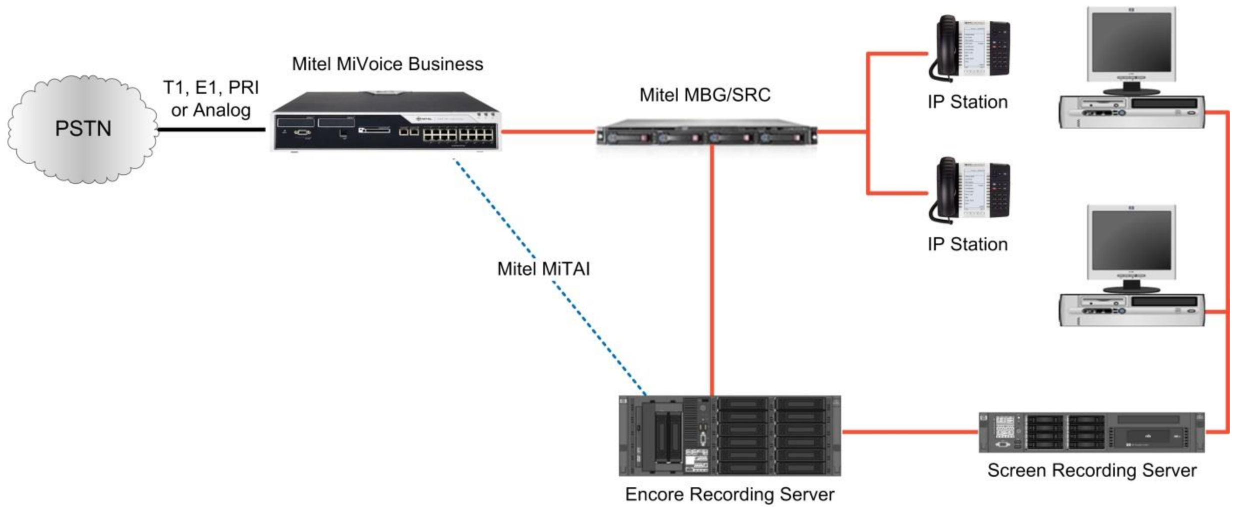
Station-side RTP Packet Capture

The Station-side RTP Packet Capture method uses a span port to collect the RTP audio packets directly from the network segment that includes the VoIP traffic. Based on events received from MiTAI, the Encore server collects the RTP packets for a specific IP or MAC address and converts the RTP data to an audio recording file. Encore collects data associated with the call from the MiTAI messages.



Subscription-based SRC Audio Stream

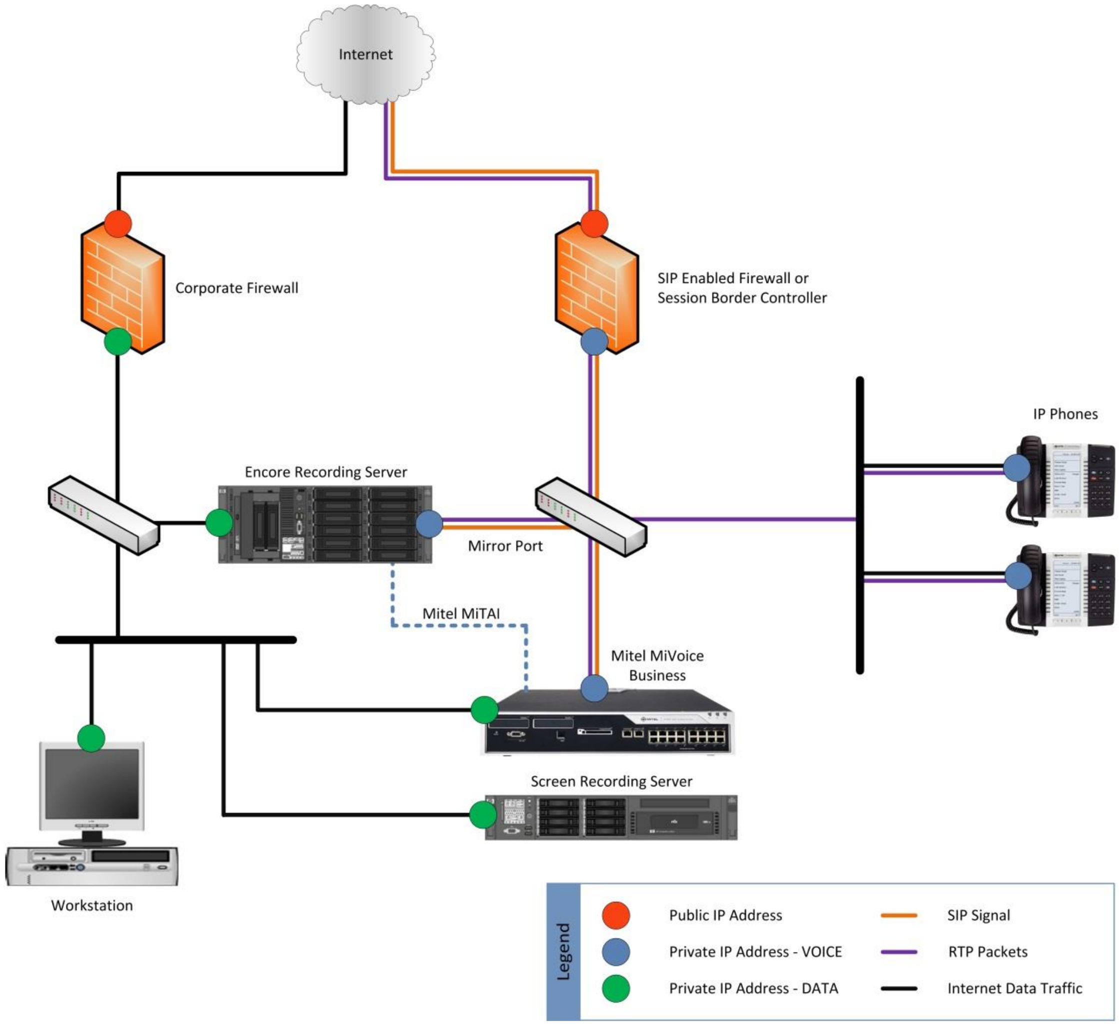
The Subscription-based SRC Audio Stream method uses the RTP packets sent from the Mitel MBG/SRC via the network to record audio. The start/stop events and call data are sent by MiTAI to the Encore system. Encore sends tap requests to the MBG/SRC server and begins recording the audio from the SRC stream.



> **NOTE** If, under very rare circumstances, the Mitel MBG/SRC server is rebooted while calls are connected, the Encore server continues to display these calls as recording until:
>
> 1. The timeout has expired, or
> 2. The first call made after the MBG/SRC reboot, on the extension in question, has terminated.
>
> In scenario #2, the audio of the next call from this extension that immediately follows the MBG/SRC reboot is appended to the original call in progress at the time of the MBG/SRC reboot. Therefore, the reviewer would hear a portion of the original call, immediately followed by the first call made after the MBG/SRC reboot for the extension in question. Any call data for the appended call is lost.

Trunk-side SIP Packet Capture

The SIP Trunk recording method uses a span port but employs the SIP Trunk Audio Server to collect the SIP packets and the RTP packets from the trunk. Based on the SIP trunk events, the recorder starts and ends recordings for SIP trunk calls. If the call terminates to a monitored device, Encore collects data associated with the call from the MiTAI messages; otherwise Encore only collects call data present in the SIP message.



NOTE When recording SIP trunks, DVSAnalytics prefers that all SIP trunk traffic goes through a Session Border Controller (SBC), such as a Cisco CUBE, Ingate SIParator, etc. The SBC's LAN-side port must terminate to a network switch that can provide a SPAN/mirror port to the Encore server.

If an SBC cannot be provided, then the SIP trunk provider must be able to guarantee that a single IP Address will be used for all SIP and RTP media packets for both inbound and outbound calls. The LAN-side port of the device used for SIP trunk traffic must terminate to a network switch that can provide a SPAN/mirror port to the Encore server.

Configure Mitel System

The steps to configure the Mitel system are included in this section. The screenshots of the MCD/MiVoice Business platform are from MCD v4.1 SP1 and MiVoice Business 8.0 and the screenshot from Mitel SRC is from version 2.2.14.0. These screenshots may differ from the version installed on your system. It is assumed that you have a working knowledge of the Mitel systems, and only need specific configuration assistance.

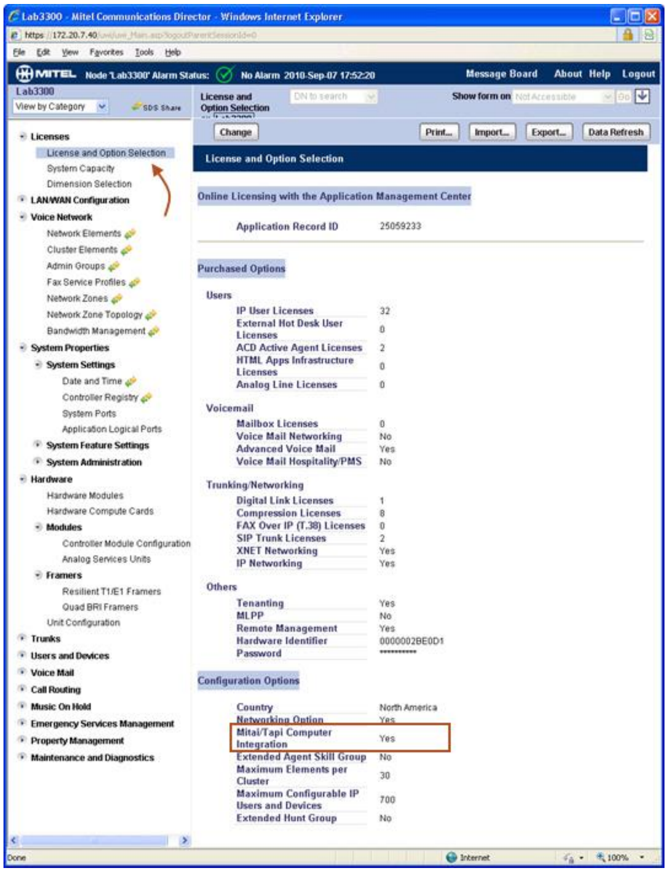
Step 1 is performed for all audio collection methods. Step 2 is only performed if using the Subscription-based SRC Audio Stream method.

Step 1: Configure the MCD/MiVoice Business Software

For MCD v4.x to v6.x only

Expand **Licenses** from the menu on the left and then select **License and Option Selection**.

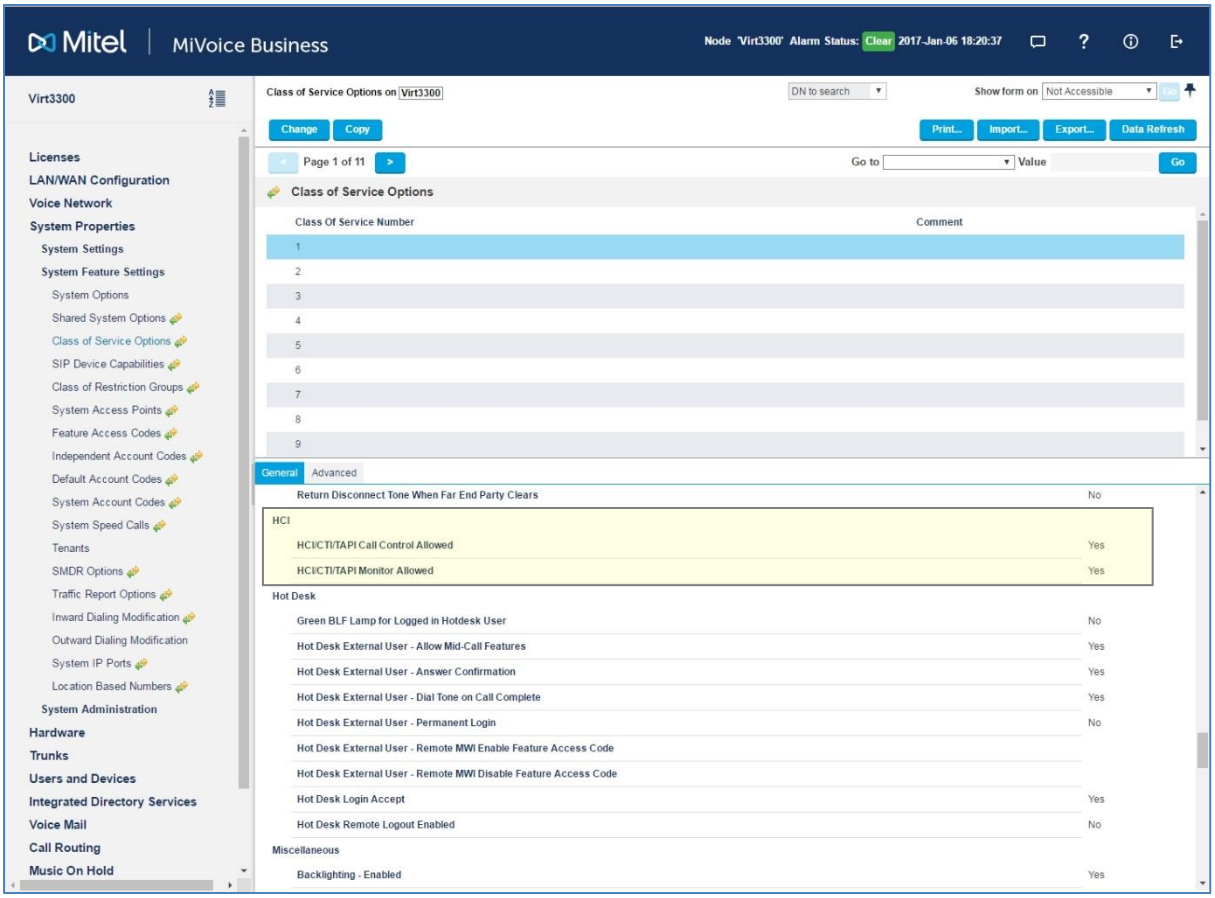
Scroll to the **Mitai/Tapi Computer Integration** option and set it to **Yes**. This option may only be available if you have purchased a MiTAI license from Mitel. Then click **Save**.



For all Mitel systems

Next, it is important to set a few parameters for the class of service assigned to each phone that needs to be recorded. In the MiVoice Business software, expand **System Properties** and then expand **System Feature Settings**. Select **Class of Service Options**.

The window now changes to show the class of service numbers that have been configured.



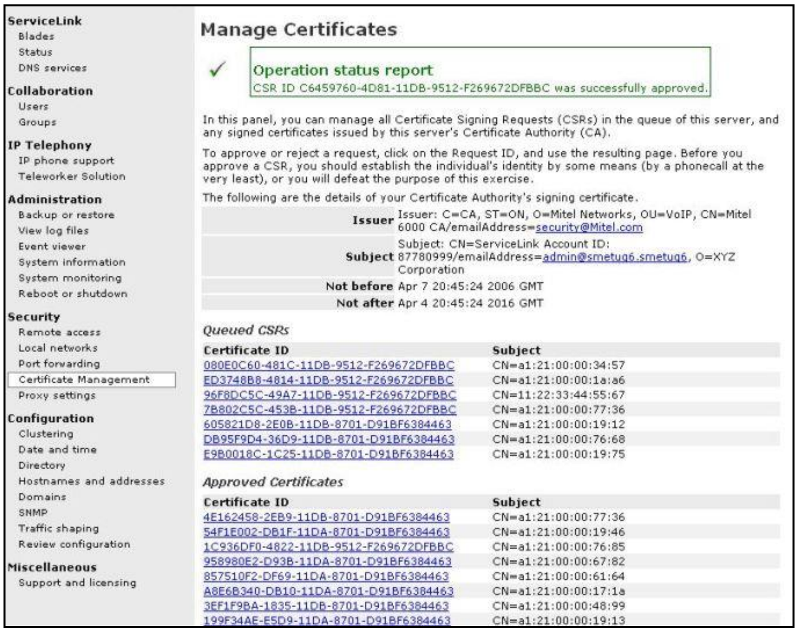
Click the class of service number that is assigned to phones that must be recorded and verify the following two parameters are set as show below:

- **HCI/CTI/TAPI Call Control Allowed** parameter set to **Yes**
- **HCI/CTI/TAPI Monitor Allowed** parameter set to **Yes**

If the parameters need to be changed, click **Change** at the top of the window, make the appropriate changes, then click **Save** to save the changes. Verify these parameters are set correctly for every class of service that is assigned to a phone that must be recorded.

Step 2: Grant a Certificate on the MBG/SRC Server

Only complete this step if using Mitel MBG/SRC to obtain audio. When the Encore server connects to the MBG/SRC server for the first time, it enrolls with the MBG/SRC server to establish a trust relationship by issuing a certificate request. The system administrator responsible for the MBG/SRC server must approve the enrollment by granting a certificate via the **Certificate Management** window in the **Server Manager** on the MBG/SRC server.



Call Handling Scenarios

This section explains how different calls are displayed in Encore. The samples in this section are from a station-side recording system and it is assumed that all stations involved in the calls are configured to be recorded.

Certain situations affect how recordings are created and how they can be located using the Related Call Lookup feature:

- **Hold** – When a call is put on hold, the recording is suspended. When the call is retrieved, the audio is appended to the recording to create one audio recording.
- **Consultation Call** – If an agent is on a call and then places a consultation call, the first call is put on hold and the recording is suspended. Assuming the called party is also using a recorded phone, the consultation call is recorded as two separate recordings – one for each extension. When the agent hangs up the consultation call and retrieves the caller, the two recordings end and the first recording resumes; the second portion of the recording is appended to the first portion. All three recordings have different Segment IDs (SID) and share the same Related ID (RID).
- **Blind Transfer** – When a call is blind transferred (also called an unannounced transfer), the first recording ends after the agent presses the transfer button and hangs up the handset. The second

recording begins when the second agent answers the transferred call. The second recording ends when the second agent hangs up the call. Separate SIDs are associated with each recording and they usually share the same RID. If the call is transferred to an ACD queue or Hunt Group, it may not be possible to show the relationship between the recordings and the same RID may not be associated with both recordings.

- **Conference Call** – When an agent decides to bring a third party into a current call, the agent usually puts the caller on hold to first consult with the third party. The first recording of the agent and the outside caller suspends during the consultation call. Assuming the third party is using a recorded phone, the consultation call creates two recordings – one for the agent and another for the third party. After the consultation call ends, the first recording resumes when the agent retrieves the caller and it ends when the agent hangs up. The recording of the third party continues until the third party hangs up.
- **Internal Call** – If both extensions are monitored by Encore, two recordings are created – one for each extension. The party who initiates the call is treated as the agent for data collection purposes.

External Inbound Call

Recordings: 1 | SID: 1 | RID: 1

Extension 5002 receives an external inbound call with SID A and hangs up when the call is complete. This call creates one recording and one RID even though no other calls are associated with it.

External Inbound Call with Supervised Transfer

Recordings: 3 | SID: 3 | RID: 1

1. Extension 5002 receives an external inbound call. Recording 1 begins with SID 1.
2. The agent presses the transfer button which puts the caller on hold and suspends Recording 1. The agent then makes a consultation call to extension 5025. Recording 2 for extension 5002 begins with SID 2 and Recording 3 begins for extension 5025 with SID 3. When extension 5002 hangs up, Recordings 1 and 2 end.
3. Now the caller is transferred to the agent at extension 5025. Recording 3 continues.
4. When the agent at extension 5025 hangs up, Recording 3 ends.

The same RID is associated with all recordings to show they are related.

External Inbound Call to ACD

Recordings: 1 | SID: 1 | RID: 1

ACD Path 5300 contains ACD Skill Group 5200. Agent with login ID 5101 is logged into ACD Skill Group 5200 at extension 5006 and is available to take ACD calls.

An external call is routed to ACD Path 5300 and answered by the agent at extension 5006. A recording with SID 1 begins and, when the call ends, the recording stops. The ACD Number, ACD Name, Hunt Group, and Agent Login ID are associated with the recording.

External Outbound Call

Recordings: 1 | SID: 1 | RID: 1

Extension 5002 makes an external outbound call with SID 1 and hangs up when the call is complete. This call creates one recording and one RID even though no other calls are associated with it. The Call Direction for the recording shows as Outgoing. The dialed number (with the dialed prefix) is stored in the ANI and Other Party Number fields.

Internal Call

Recordings: 2 | SID: 2 | RID: 1

Extension 5002 makes an internal call to extension 5009 (both extensions are monitored by Encore). A recording is created for each monitored extension and each recording is assigned a different SID. Both recordings are assigned the same RID to show they are related to each other.

External Inbound Call with Blind or Unannounced Transfer

Recordings: 2 | SID: 2 | RID: 1

1. Extension 5002 receives an external inbound call which starts Recording 1 with SID 1.
2. The agent transfers the caller to extension 5009 without consulting the agent at extension 5009. Recording 1 ends when 5002 hangs up his phone.
3. Recording 2 with SID 2 begins when 5009 answers the call. It ends when the agent hangs up her phone.

The same RID is associated with each recording to show they are related.

Consultation Call

Recordings: 3 | SID: 3 | RID: 1

1. Extension 5002 receives an external inbound call which starts Recording 1 with SID 1.
2. The agent puts the caller on hold, suspending Recording 1, and makes a consultation call to extension 5025 which starts Recording 2 with SID 2 to record extension 5002. This also starts Recording 3 with SID 3 to record extension 5025 in the consultation call.
3. When the agent at 5002 hangs up the consultation call, Recording 2 ends. When the agent at 5025 hangs up, Recording 3 ends.
4. The agent at extension 5002 then retrieves the original call and Recording 1 with SID 1 resumes.
5. When extension 5002 hangs up with the caller, Recording 1 ends.

The same RID is associated with all recordings to show they are related.

Conference Call

Recordings: 3 | SID: 3 | RID: 1

1. Extension 5010 receives an external inbound call which starts Recording 1 with SID 1.
2. The agent at extension 5010 puts the caller on hold and makes a consultation call to bring a supervisor at extension 5008 into the call. This suspends Recording 1. Recording 2 with SID 2 begins to record extension 5010 on the consultation call and starts Recording 3 with SID 3 to record the supervisor at extension 5008.
3. When the supervisor at extension 5008 joins the call, Recording 2 ends. Recording 1 resumes and appends the audio to the first portion of the recording. Recording 3 continues.
4. When the supervisor at extension 5008 hangs up the call, Recording 3 ends.
5. When the agent at extension 5010 hangs up the call, Recording 1 ends.

The same RID is associated with all recordings to show they are related.

Appendix 1: Glossary

abandoned call

An incoming call which is answered by the ACD but terminated by the caller before it is answered by an agent.

ACD

Automatic Call Distributor. An application that answers calls and directs them to a predetermined queue, or line, of waiting calls. In most cases, the ACD ensures that the first call in is the first call answered. It also determines which agent receives a call based on predetermined criteria such as idle time or availability and generates reports on call volume and distribution.

agent

A person who handles phone calls. Other variations include operator, attendant, representative, customer service representative (CSR), telemarketer, phone sales representative (TSR), and so on.

ANI

Automatic Number Identification. For inbound calls, this is the phone number from which the customer is calling (may not be supported by the trunk). For outbound calls, this is the dialed number.

automated attendant

A voice processing system that answers calls with a recording and then enables callers to press touch-tone buttons to navigate through a menu system to a person, department, or voice mail.

beep tones

This feature is available for the Subscription-based SRC Audio Stream method and plays a beep tone every 15 seconds when a call is recorded. The agent and the customer hear the beep but it is not audible in the recording.

call direction

The direction is either incoming (inbound) or outgoing (outbound).

call ID

A unique identifier used by the database to locate each recording. The call identifier for a recording can be viewed in Encore.

call record

An entry in a database that holds the data associated with a call.

call type

The call type is either internal or external.

consultation call

A call that is made while the customer (original call) is on hold. In the database, the **Consultation Call** field shows **Yes** when the recording is a consultation call.

digital recording

A method of recording that converts analog sound into a series of pulses that are translated into binary code, which is read by computers.

DNIS

Dialed Number Identification Service. For inbound calls, this is the number the customer dialed or the agent's extension number (may not be supported by the trunk).

encrypted calls

Calls that have the audio RTP packets encrypted. This prevents 3rd party applications, such as the Encore system, from using the RTP packets for recording.

extension

The number associated with a person's station. Extension and station are sometimes used interchangeably.

external calls

In these calls, the calling or called parties are outside the PBX.

hold duration

The sum of all hold durations that occurred during the recording. This is not captured for the Trunk-side SIP Packet Capture audio collection method.

Hot Desk

Each Hot Desk user has a user identifier (which is the user's directory number) and a pin number to log into the system. When logging into a phone that is Hot Desk enabled, the user takes complete control of the set (including line keys, soft keys, etc.) The set now has a new prime directory number—the user's directory number. The registration directory number is unavailable as long as the user is logged into the phone. When the user logs out of the phone, the registration directory number (with line keys, soft keys, etc.) is restored, and the user directory number becomes unavailable.

hunt group

If the call is a hunt group call, then the number of the hunt group is stored in this database field. If the call is not in a hunt group, then the field is blank.

inbound

Calls which are received/answered by a recorded party.

internal calls

Calls made between extensions on the same PBX.

key system/multicall groups

Key system groups and multicall groups let multiple phones share the same extension number. Incoming calls ring all the idle stations, and the stations stop ringing when one member answers the call. Only one member of a key system group can use the line at one time; a multicall group allows simultaneous users.

When a member of a key system group answers a call, the line becomes busy (only one member can use the line at one time); however, when a member of a multicall group answers a call, the line becomes idle (all group members can use the line simultaneously).

When a member of a key system group places a call on hold, the call can be retrieved by any member of the group. When a member of a multicall group places a call on hold, the call can be retrieved only by the set that placed the call on hold.

media encrypted calls

Calls that have the audio RTP packets encrypted. This prevents 3rd party applications, such as the Encore system, from using the RTP packets for recording.

multicall groups

See key system/multicall groups.

other party name

Name of the other party on the line with the person being recorded; may be blank if this is an external call.

other party number

Number of the other party on the line with the person being recorded; if external and incoming call, this is an ANI.

outbound

Calls which are initialed/placed by a recorded party.

pause/resume on hold

A method that pauses the recording of audio and screen when a call is placed on hold, and resumes recording when the hold is taken off.

PBX (PABX)

Private (Automated) Branch Exchange. The phone system to which the office phones are connected.

recorded party name

Name of person being recorded.

recorded party number

Number of person being recorded.

recording

The audio recording, screen recording, and database record associated with a single phone call or conversation.

Related ID (RID)

This unique ID is created by CT Gateway and is associated with all related call segments so you can view the customer's entire interaction with the contact center, including the number of people the customer spoke to, how many times the customer was transferred, etc. As an example, if an agent is on a call and then places a consultation call, the first call is put on hold and the recording is suspended. Assuming the called party is also using a recorded phone, the consultation call is recorded as two separate recordings – one for each extension. When the agent hangs up the consultation call and retrieves the caller, the two recordings end and the first recording resumes; the second portion of the recording is appended to the first portion. All three recordings share the same RID.

Segment ID (SID)

This unique ID is generated by CT Gateway and assigned to each recording. For instance, if an agent is on a call and then places a consultation call, the first call is put on hold and the recording is suspended. Assuming the called party is also using a recorded phone, the consultation call is recorded as two separate recordings – one for each extension. When the agent hangs up the consultation call and retrieves the caller, the two recordings end and the first recording resumes; the second portion of the recording is appended to the first portion. All three recordings have different SIDs.

station

A phone connected to the PBX.

TAPI

Telephony Application Programming Interface. A telephony software interface included in Microsoft Windows operating system that supports the incorporation of telephony control by other applications.

TCP/IP

Transmission Control Protocol/Internet Protocol.The Internet Protocol Suite is the set of communications protocol used for the Internet and other similar networks.

trunk

The connection between the phone company and the PBX that carries incoming calls.

trunk ID

The trunk of a call. If the call is an internal call, then this field is blank. Captured for all inbound/outbound calls.

work group name

If the call is a work group call, then the name of the work group is stored in this database field. If the call is not in a work group, then the field is blank.

work group number

If the call is a work group call, then the number of the work group is stored in this database field. If the call is not in a work group, then the field is blank.