



A MITEL
PRODUCT
GUIDE

MiVoice MX-ONE

Surveillance, Observation, and Monitoring Using SIP Proxy - Description

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Contents

- 1 General..... 1**
 - 1.1 Description..... 1
 - 1.2 Acronyms..... 1

- 2 Facilities..... 2**
 - 2.1 General..... 2
 - 2.2 Metadata..... 2
 - 2.3 Speech Monitoring..... 3
 - 2.4 External Recording Server..... 3

This chapter contains the following sections:

- [Description](#)
- [Acronyms](#)

1.1 Description

The Legacy SOM functionality has some hardware dependency and will not work in a software-only deployment. It is also limited to specific objects that are selected for monitoring and does not provide the facility to monitor all the calls in the system. The new SOM feature, which uses SIP Proxy will address these limitations.

SOM using SIP Proxy provides a function for monitoring of traffic in the system from a central monitoring server. The monitoring server is added as a free server in the MiVoice MX-ONE system without making it a LIM. The SOM functionality using SIP proxy is software based unlike the legacy SOM, which works on TDM Trunk Lines.

The monitoring server will have a media server instance and a stateless SIP proxy running on it. The media server will handle the redirection of RTP streams between endpoints. The stateless SIP proxy is used to receive metadata information of the calls so that this information can be mapped to the media streams.

After the function is enabled, the MiVoice MX-ONE will treat all the calls in the system as gateway calls and redirect the media traffic through the media server. The MiVoice MX-ONE also generates a SIP call through the SIP proxy running on the monitoring server. This SIP call contains the metadata required to capture the call and generate call data records. The metadata contains the call information along with the port numbers on media server where the RTP streams will be captured.

The administration of the surveillance function, that is, enabling, disabling, and configuring the proxy address, is done using a UNIX-based command.

The SOM using SIP proxy functionality is license-controlled. A port-level license is required to enable the "SOM using SIP proxy" feature. A license tag is used for each configured LIM.

1.2 Acronyms

Table 1: Acronyms

RTP	Real Time Protocol (IP media protocol)
SIP	Session Initiation Protocol, IETF standards for IP signaling

This chapter contains the following sections:

- [General](#)
- [Metadata](#)
- [Speech Monitoring](#)
- [External Recording Server](#)

2.1 General

The SOM functionality using SIP proxy makes all the traffic in the system go through a dedicated server. This is done by making the calls act as gateway calls through the media server instance. Because of this, the media streams can be monitored and captured on the monitoring server. To provide the metadata information for each call, the MiVoice MX-ONE system will generate a SIP call with all the information through the SIP proxy.

For small deployments, one central monitoring server must be idle, which will make mirroring the streams to external server easy from single location. In this scenario, all the LIMs in the system must be configured with the same SIP proxy address that runs on the central monitoring server.

But in case of large deployments spanned over multiple locations, there can be multiple monitoring servers installed based on the network setup. The media traffic will be redirected to the media server selected by the LIM. In this case, the LIMs must be configured with the address of the SIP proxy on the monitoring server near to the LIM. The mirroring of the SIP and RTP streams must also be planned properly in the network so that all the traffic from monitoring servers will reach the external server for recording.

2.2 Metadata

The metadata of the calls will be exposed by making a SIP call through the proxy running on the monitoring server. This is done for all types of calls whether IP or non-IP.

The SIP signalling will include the A-party and B-party information along with the media information such as codec, address, and port on media server where the RTP streams can be captured. The SIP INVITE message contains the CSTA Connection Call ID. This can be used to map the proxy call to the original call.

The proxy SIP call is initiated by sending INVITE to the SIP proxy after the actual call is answered and enters speech state. The INVITE will have A-Party and B-Party information in the From and To SIP Headers respectively. The INVITE SDP will have the media details for one RTP stream direction. The SIP proxy will redirect the INVITE back to the same LIM. The LIM will then reply with 200 OK for the INVITE. The 200 OK SDP will have the Media Details for the other RTP stream direction.

In case of conference call, separate proxy call is initiated for each party in the conference. The call will contain the specific party information along with media details for that party.

The SOM using SIP Proxy feature supports monitoring of traffic when the call is in speech state. Therefore, when the media path of the original call is disconnected because of call hold, the proxy SIP call will be terminated by sending BYE and when the media path re-establishes the proxy SIP call is sent again.

2.3 Speech Monitoring

The media streams for each call will be redirected through the media server instance. Because of this, there will be in total 4 RTP streams (2-pairs, one coming from endpoints to the media server, and the other going from the media server to endpoints) visible on the media server for each original call. The SIP signalling generated for metadata will provide the RTP ports to capture one pair of the RTP streams. The other pair of streams can be ignored.

For conference calls, the media streams of each party will be available separately. The port numbers of each stream will be sent in separate SIP calls to the proxy.

2.4 External Recording Server

The SIP and RTP streams available on the monitoring servers can be mirrored in the network and sent to an external server for storage and future reference. This external server can host any third-party application that will capture the mirror streams redirected from the network. The internal implementation of the third-party recording application is out of scope of this document.

