



A MITEL
PRODUCT
GUIDE

MiVoice MX-ONE

Voice Recording

Release 7.7
60/1551-ANF 901 14 Uen E

May 2024

Notices

The information contained in this document is believed to be accurate in all respects but is not warranted by **Mitel Networks Corporation (MITEL®)**. The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes. No part of this document can be reproduced or transmitted in any form or by any means - electronic or mechanical - for any purpose without written permission from Mitel Networks Corporation.

Trademarks

The trademarks, service marks, logos and graphics (collectively "Trademarks") appearing on Mitel's Internet sites or in its publications are registered and unregistered trademarks of Mitel Networks Corporation (MNC), its affiliates, parents, or subsidiaries (collectively "Mitel") or others. Use of the Trademarks is prohibited without the express consent from Mitel. Please contact our legal department at legal@mitel.com for additional information. For a list of the worldwide Mitel Networks Corporation registered trademarks, please refer to the website: <http://www.mitel.com/trademarks>.

®,™ Trademark of Mitel Networks Corporation

© Copyright 2024, Mitel Networks Corporation

All rights reserved

Contents

1 GENERAL.....	1
2 TRADITIONAL RECORDING.....	2
2.1 WHERE IT BEGAN.....	2
2.2 TRUNK SIDE RECORDING.....	2
3 PASSIVE IP RECORDING.....	3
4 ACTIVE IP RECORDING.....	5
5 GENERAL MX-ONE FUNCTIONALITY FOR VOICE RECORDING.....	6
5.1 GENERAL.....	6
5.2 ANALOG TELEPHONES.....	6
5.3 CAS EXTENSIONS.....	6
5.4 DIGITAL PHONES.....	7
5.5 INTEGRATED DECT.....	7
5.6 ISDN EXTENSIONS.....	7
5.7 MOBILE EXTENSIONS.....	7
6 MX-ONE FUNCTIONALITY FOR ACTIVE IP VOICE RECORDING.....	8
6.1 IP TELEPHONE FUNCTIONALITY.....	8
6.2 MX-ONE SERVICE NODE FUNCTIONALITY.....	9
6.3 MX-ONE SUPPORT FOR RECORDING OF ENCRYPTED CALLS.....	9
7 LIMITATION.....	10
8 LICENSES.....	11

Voice recording is important for many customers; it may be required by legislation or due to self-interest for business purposes where deals or agreements via the phone may have significant economic value.

Traditionally, Voice Recording equipment was added to a PBX by a third-party recording vendor, and there was no integration between the two systems. With the introduction of IP and direct media calls that may be required to be recorded, good integration between PBX and voice recording equipment must provide working and cost-effective solutions.

There are many recording equipment vendors in the market, and their capabilities and levels of integration with MX-ONE are very different. This document does not describe or name any specific recording vendor.

This chapter contains the following sections:

- [WHERE IT BEGAN](#)
- [TRUNK SIDE RECORDING](#)

Traditional, i.e., recording of TDM-type users, are available from most major vendors of recording systems without any need for integration with the PBX.

2.1 WHERE IT BEGAN

The first recording equipment was connected to the MDF, where a physical tap on the target user's telephone line was made. Recording vendors developed specific hardware, often by using reverse engineering, to be able to record conversations made with, e.g., proprietary digital telephones.

This type of recording is still used.

2.2 TRUNK SIDE RECORDING

Trunk side recording is accomplished by tapping incoming/outgoing trunks in the MDF by using dedicated hardware. This type of recording allows the recording of external calls, which are normally the most important.

Trunk side recording is available in two different versions. The integrated version is where CTI monitoring of end users is used to get a correct connection between the recorded user and the trunk identity over which his call is made. This is the recommended version.

If no integration is used, the recorder will listen to the D-channel signaling to connect the calling or called party with the actual recording. Note that this method does not guarantee a correct result. The recorder only gets the originally called/calling party number. The user actually talking over that trunk individual may be anyone in the system after any type of call forwarding, call pick-up, rerouting, call to hunt group, etc. Connected Line Identity Presentation, if supported by the network, will give sufficient information to the recorder about changes in the connected party unless features where the actually connected party number is hidden or replaced by, e.g., the pilot number of a hunt group.

The introduction of IP telephony required new solutions for voice recording. Passive recording has been available for several years, and despite certain limitations, it is a useful solution in many cases, and no integration with the PBX is required.

Passive IP recording uses the capability of a LAN switch to make a copy of all data traffic to/ from a user and, via the SPAN port, send this information to, e.g., a voice recorder. The voice recorder has the intelligence to listen to the signaling exchanged between PBX and telephone and retrieves information about called, calling, and connected party.

Note that LAN switches normally do not have sufficient intelligence to select and send only voice-related data to the central recorder.

The following shows the architecture and also the obvious limitations. For a remote IP extension, in particular, if the local office has direct access to PSTN, a vast amount of data needs to be sent over the WAN.

If only a few users are recorded and PSTN access is only available at the headquarters, the SPAN ports of LAN switches connecting the PSTN gateways can be used. If direct media calls shall be recorded, a forced gateway Class of Service for applicable users is needed.

Furthermore, if the customer requires that IP calls should be encrypted, there is no possibility to record and associate calls to any specific user, as the recording system does not have access to the keys used for the TLS session nor to the keys used to encrypt the media stream.

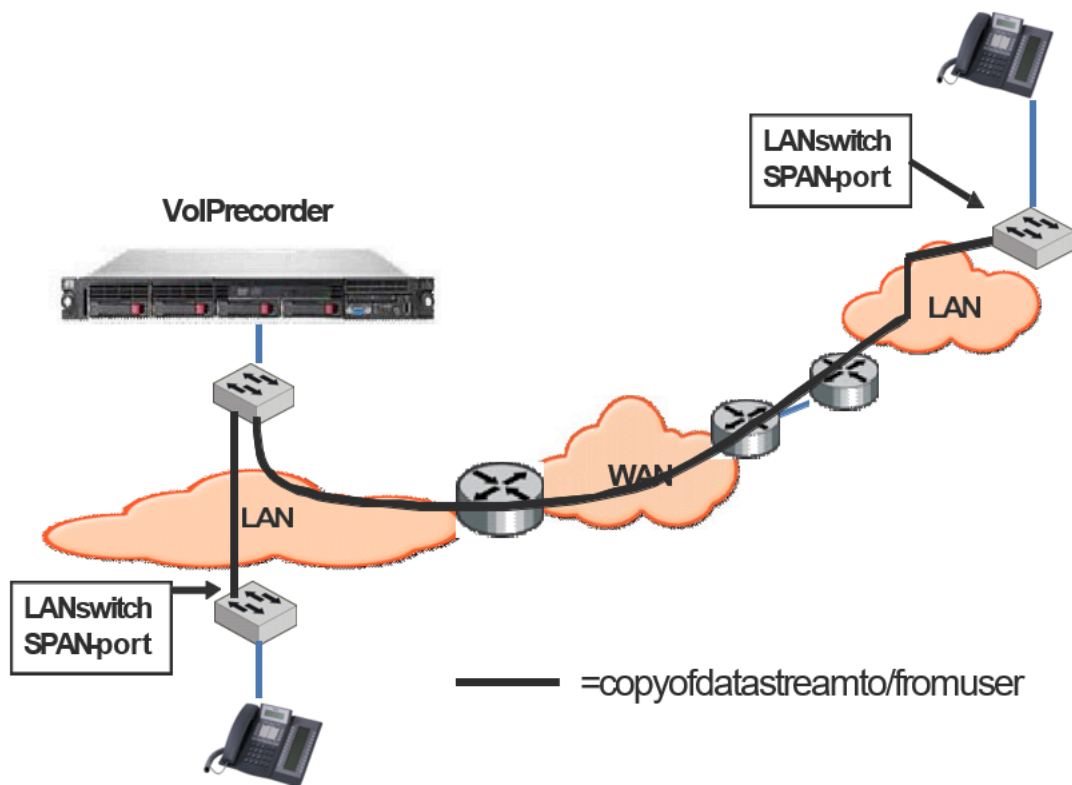


Figure 1: Passive IP recording

ACTIVE IP RECORDING

4

Active VoIP recording solves the problems/limitations found with passive recording.

With active VoIP recording, the recording system will order the terminal to make a copy of received and sent RTP or SRTP packets and forward those to the recorder, preferably as two logical streams where received and sent packets are separated. Certain features of advanced VoIP recording systems are dependent on the receipt of two separate voice streams.

Recording policies are defined in the recording systems, i.e., record all calls (Total recording), selective recording based on, e.g., call origin, and record on demand based on user input.

Active VoIP recording relies on integration with the PBX, as the recording system must know the actual IP address of the user's terminal when ordering a terminal to start recording. Furthermore, if encryption is used, the recording system must get the encryption keys. The integration normally uses SIP between the recorder and the terminal and CTI between PBX and the recorder.

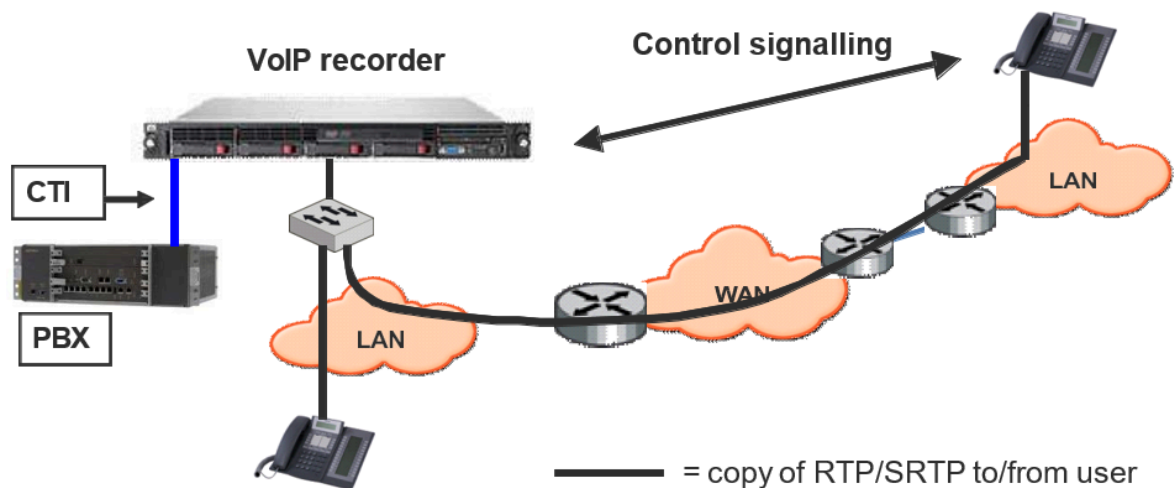


Figure 2: Active IP recording

GENERAL MX-ONE FUNCTIONALITY FOR VOICE RECORDING

5

This chapter contains the following sections:

- [GENERAL](#)
- [ANALOG TELEPHONES](#)
- [CAS EXTENSIONS](#)
- [DIGITAL PHONES](#)
- [INTEGRATED DECT](#)
- [ISDN EXTENSIONS](#)
- [MOBILE EXTENSIONS](#)

5.1 GENERAL

The MX-ONE provides a number of features and interfaces that can be used, fully or partly, by recording vendors to provide customer solutions for voice recording.

This section describes on a high level which support MX-ONE provides per type of terminal without any requirements for tight integration. For trunk-side recording, which is the most commonly used solution, required Trunk Identities are included by MX-ONE in relevant events over CSTA 3 for PSTN and tie-line calls.

5.2 ANALOG TELEPHONES

No specific support.

To record all calls, physical tapping of the user's line in the MDF or complex configuration where calls to/from a user will "hot-line" through a looped and recorded trunk is required. Standard event reporting with CSTA 3 allows the use of trunk-side recording for external calls.

5.3 CAS EXTENSIONS

No specific support.

To record all calls, the E1/T1 connecting these users to the MX-ONE can be tapped with the trunk side recording technique. Standard event reporting with CSTA 3 allows the use of trunk-side recording for external calls.

5.4 DIGITAL PHONES

No specific support.

To record all calls, physical tapping of the user's line in the MDF or complex configuration where calls to/from a user will "hot-line" through a looped and recorded trunk is required. Standard event reporting with CSTA 3 allows the use of trunk-side recording for external calls.

5.5 INTEGRATED DECT

No specific support.

To record all calls, a complex configuration where calls to/from a user will "hot-line" through a looped and recorded trunk is required. Standard event reporting with CSTA 3 allows the use of trunk-side recording for external calls.

5.6 ISDN EXTENSIONS

No specific support.

To record all calls, the ISDN interface connecting these users to the MX-ONE can be tapped. Standard event reporting with CSTA 3 allows the use of trunk-side recording for external calls.

5.7 MOBILE EXTENSIONS

Standard event reporting with CSTA 3 allows the use of trunk-side recording for external calls.



Note:

This is the only recording option for this type of user.

MX-ONE FUNCTIONALITY FOR ACTIVE IP VOICE RECORDING

6

This chapter contains the following sections:

- [IP TELEPHONE FUNCTIONALITY](#)
- [MX-ONE SERVICE NODE FUNCTIONALITY](#)
- [MX-ONE SUPPORT FOR RECORDING OF ENCRYPTED CALLS](#)

6.1 IP TELEPHONE FUNCTIONALITY

The MiVoice 4400, Mitel 7400, Mitel 6900/6800/6700 ranges of telephones have full support for active recording.

The base control protocol between telephone and recorder for total, selective, and threat recording is based on SIP.

When ordered by the recorder, the telephone creates copies of received and sent RTP or SRTP packets and forwards these as two logically separate streams to the recorder. The telephone will always send the same type of packets (RTP or SRTP) as is used for the "real" media stream.

The telephone will continue to send the copies until the user hangs up or when the recorder orders it to stop.

The telephone must get the order to copy packets for each call that should be recorded,

i.e. integration with CTI is a must.

The telephone will screen SIP invites received and will only act on Invites received from a trusted source. Trusted sources, a maximum of six, are configured in the telephone configuration file.

The SIP invite contains the IP address of the server to which the copies of the media stream shall be sent. This allows the recording system to provide load balancing for capacity and redundancy purposes.

Recording on demand is supported by the telephones. When this option is used, a shortcut (function) key on the telephone can be used to start the recording of a specific call. When the predefined recording key on the terminal is pressed, a start recording request message based on https/XML/SOAP is sent to the recorder. The recording stops when the call is cleared. Depending on the recording system, the recording can also be stopped when the key is pressed a second time.

The IP address to which the record on-demand request shall be sent and additional data to enable the functionality is configured in the telephone's configuration file. For Mitel 6900/6800/6700 phones, this data can be configured via an MX-ONE command.

Most manufacturers provide an alternative record-on-demand feature where a client application on the user's PC is used to start recording and provide a GUI to search, retrieve, and listen to recordings.

6.2 MX-ONE SERVICE NODE FUNCTIONALITY

The functionality of the IP telephones is enough to provide solutions for the active recording of unencrypted voice calls. Supported by CSTA 3 integration, the VoIP recorder will know the IP address of the target users' active telephone, and this is sufficient for basic recording needs.

Roaming, i.e. when the user logs on/off, is reported with an event named **CXN/IP extension location registration**. The IP address of the telephone used for logging on is included in the data. Furthermore, the application can request the IP address of any user by the **Verification Request** function. If the directory number requested to be verified is a registered IP extension, the IP address is included in the response.

For recording on demand or threat call recording, MX-ONE can reserve a key on the IP telephones used by targeted users. MX-ONE will only reserve the key and prevent that it is used for other purposes.

The key is part of the user's personal instrument layout which is loaded when the user logon to an IP phone.

6.3 MX-ONE SUPPORT FOR RECORDING OF ENCRYPTED CALLS

Recording of encrypted media streams, using SRTP, is supported by MX-ONE, but not necessarily by all recording applications.

The encryption keys for the call, both for the original basic call, but also in situations where the encryption keys are re-negotiated, as for hold, inquiry, transfer and similar services, are forwarded, for CSTA phase 3 in the event "**PrivateEventCSTAIII**", and in CSTA phase 1 in the event "**Additional information**" with function code **#B8**.

The sending of encryption keys is controlled by a setting in the "**csta-serv/SERV**" parameter of the "**csta -i/ CSTLI**" commands.

Call Admission Control

CAC does not consider if active voice recording is used for a specific call or a specific user. The available bandwidth for a domain must be adjusted to take VoIP recording into account.

CSTA

The sending of encryption keys in private events is supported by CSTA 3. See “Compatibility Matrix, MiContact Center Enterprise” for which versions.

Directory number length

Application Link and OAS can handle up to 5-digit directory numbers. CSTA 3 can handle up to 10-digit directory numbers.

Type of Forwarding Devices

IP operators (Operator Media Devices), IP soft clients, IP trunks (both H.323 and SIP), and media gateways do not support VoIP recording.

Codecs

The telephone copies the rtp stream to the recording logger using the same codec as for the call. Some recording equipment does not support all the codecs that the telephone supports, which means that the recording will not work if an unsupported codec is used.

Multiple Terminal Service

This feature is used when a directory number is associated with more than one phone or terminal. Recording can be combined with multiple terminal services, but the recording application must then support terminal identity and multiple endpoints.

Otherwise, there may be limitations, e.g., that only one of the terminals can be recorded.

Hold

Depending on the recording system, the recording might stop when the user presses hold. The recording continues in another file when the call is un-parked.

In the case of a record on demand, the user might have to press the recording key again to continue the recording.

There is one per-user license for the record on-demand key (per IP extension with that type of key).

Depending on which recording application is used, there may be other licenses required, but those are handled outside the MX-ONE system.

