VoIP Recording Interface

INTERWORKING DESCRIPTION

Mitel®

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1 SCOPE

1.1 IN SCOPE

This is a description of the interface between the IP telephone and the recording system.

1.2 NOT IN SCOPE

For a description of the complete recording solution with all involved devices, see description of *VOICE RECORDING* (60/1551-ANF 90114).

1.3 FUNCTION

The Voice Recording function for voice media 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 economical 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 in particular direct media calls that may be required to be recorded, a good integration between PBX and voice recording equipment is a must in order to provide working and cost effective solutions.

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

The recording function is supported for SIP and H.323 terminals, and requires CSTA support, especially if voice media encryption is used.

2 ACRONYMS AND GLOSSARY

2.1 ACRONYMS

CSTA	Computer Supported Telecommunications Applications
H.323	IP Standards from ITU-T
LAN	Local Area Network
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SOAP	Simple Object Access Protocol
VoIP	Voice over the Internet Protocol
XML	eXtended Markup Language

3 DESCRIPTION

Recording is supported in the following IP terminals; MiVoice 442x (DBC 42x), Mitel 743x (DBC 43x), Mitel 744x (DBC 44x) Mitel 6700/6800/6900.

There is support for the following options:

- Active VoIP Recording
- Record on Demand

3.1 ACTIVE IP VOICE RECORDING

Active VoIP Recording means that the calls to and from the predefined and monitored extensions are recorded. The VoIP logger is connected to the LAN. The telephones will duplicate the RTP streams and send them to the VoIP logger. The end-user cannot control if a certain call shall be recorded or not.

The CTI interface enables the recording system to collect information in real time. The extensions that shall have the voice recording possibilities must be monitored via the CSTA interface. In this interface the PBX sends events and status about the calls.

There are two types of recording sessions:

- **Dynamic mapping**. The recording session is established and kept for each call.
- **Static mapping**. The recording session is established when the terminal has registered to the PBX and when the first call is started. The session is kept until the terminal is logged off.

Which method to be used is configured in the recording system, and depends on the recording needs and prerequisites of the customer. See the recording system provider’s documentation.

3.2 SIP INTERFACE

The RTP forwarding signaling interface between the telephone and the recording system is based on SIP.

When dynamic mapping is used and if the recording policy states that a call shall be recorded, the recording system sends an INVITE to the telephone. This message contains information about where the RTP packets shall be sent. The telephone answers with a SIP 200 OK. A SIP ACK message orders the telephone to start sending the packets to the logger. The telephone continues to send the packets until a SIP BYE message is received from the recording equipment.

When static mapping is used, the recording system sends the INVITE message when the terminal is registered and BYE at log off.

As a keep alive check, the recording system sends a SIP UPDATE message.

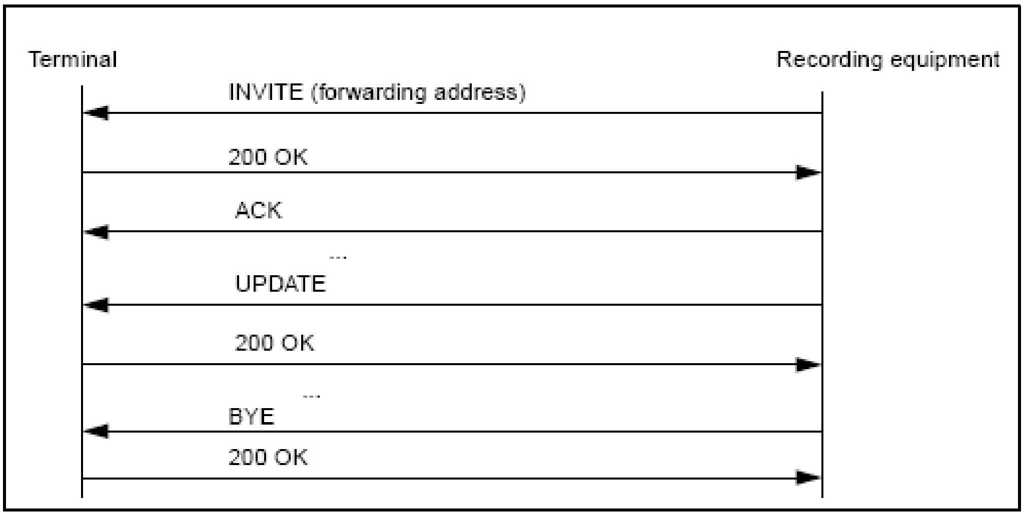


Figure 1: Message flow in the forwarding media session

The telephone checks if the IP address from which the INVITE/ACK/UPDATE/BYE messages are sent, correspond to the allowed IP addresses defined in the telephone configuration file.

To increase robustness, the following additions are made:

- The telephone sends BYE to the logger when the telephone makes an un-registration (if an INVITE but no BYE has been received before).
- The phone ignores INVITE when it is logged off.

In the attached LAN trace a working recording session is shown. It is very important that the SIP message header, message body and especially SDP are followed as shown in the LAN trace.

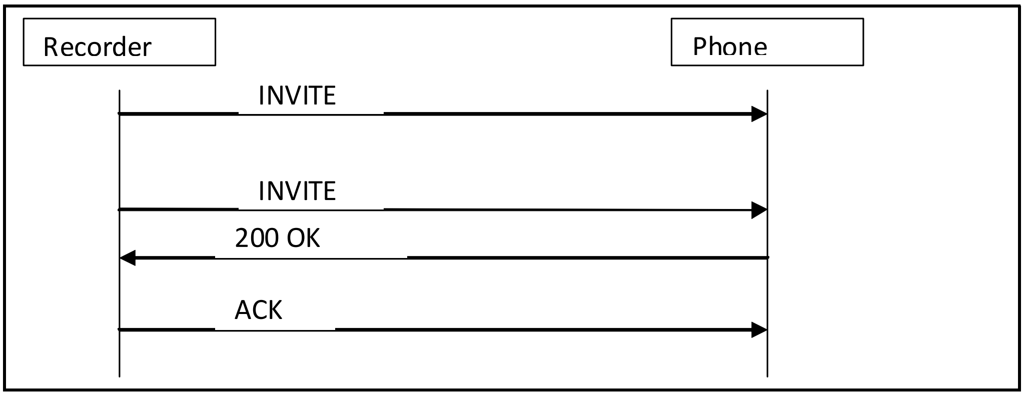
Note: Copy the 156_start_end_recording .pcap file attached to the document to a folder and open it in Wireshark packet analyzer.

For Mitel 7400 and MiVoice 442x: the SIP stack in the telephone listens on port 7300. The port number is fixed.

For Mitel 6700: the port number is possible to set via the configuration file.

3.2.1 MULTIPLE INVITE MESSAGES

When the recorder sends an INVITE but does not get an answer within a certain time, the recorder sends another INVITE with the same session id. The telephone sends 200 OK on the first INVITE and receives an ACK from the recorder. The phone checks if the session id in the ACK message is associated to the first INVITE message. In this case the recording will start.

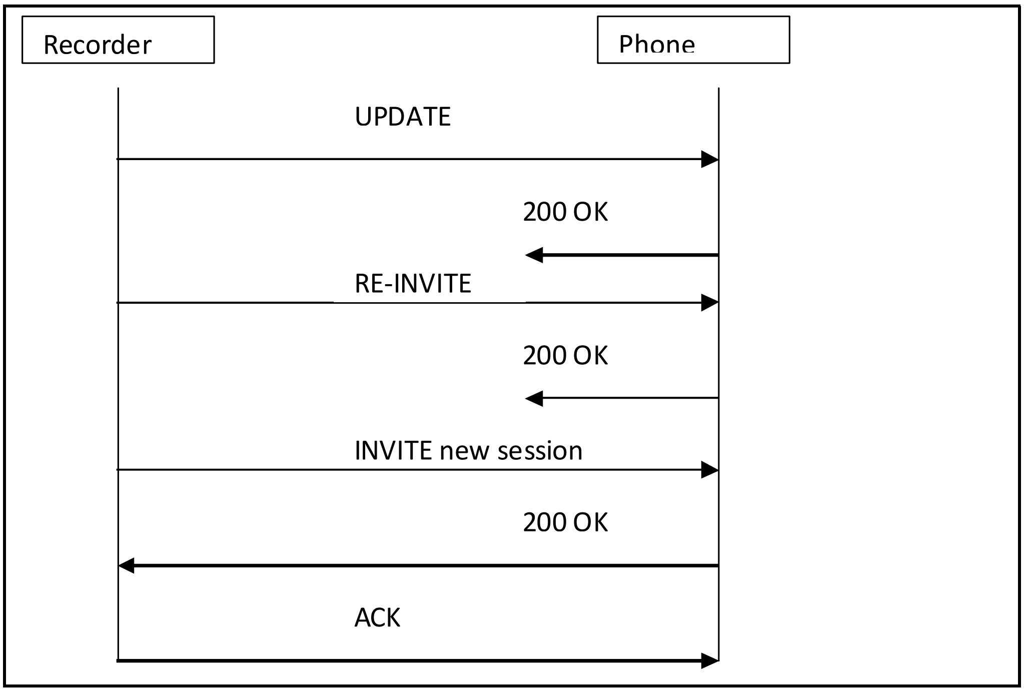


3.2.2 KEEP ALIVE CHECK

The recorder sends UPDATE messages as a keep alive check and the

phone answers 200 OK. If the recorder does not get the 200 OK within a certain time, the recorder will send a RE-INVITE as a session refresher. If there is no response on the RE-INVITE, the recovery mechanism will start working and the recorder sends a new INVITE and a new session will be opened.

The telephone does not send any BYE when closing the first session.



3.2.3 FORCED LOG OFF

If a telephone A is registered with an extension number 1000 and another telephone B registers with the same number, telephone A will be logged off automatically. The PBX will inform the recording system via the CSTA interface that extension 1000 has the IP address to phone B instead of A.

When telephone A is forced to unregister, it sends a BYE message to the recorder.

The recording system will send the next INVITE to phone B.

3.2.4 POWER IS SWITCHED OFF

When the power to the telephone is switched off, the phone cannot send any BYE message to the recording system. When the phone registers towards the PBX, information about the IP address and extension number is sent out on the CSTA interface to the recording system.

The recording system will send an INVITE message to start the recording.

3.2.5 UPDATE CONFIGURATION AND FIRMWARE IN THE TELEPHONE

If the firmware shall be updated in the phones, there is a command in the PBX to order the phones to unregister and read the configuration file. The phone will send a BYE message to the recording system. If the firmware shall be updated or if there is a major change in the configuration file, the phone will restart. When the phone registers again towards the PBX, information about the IP address and extension number is sent out via the CSTA interface to the recording system.

The recording system will send an INVITE message to start the recording.

3.2.6 RESTART IN THE MX-ONE SYSTEM

When a restart happens in the server in which the telephone is registered, the keep alive check in the phone towards the gatekeeper will discover that the gatekeeper does not answer. The telephone will try to unregister and it will send a BYE message to the recording system if there is an active SIP session. Then the telephone will try to register and when the registration is successful, the PBX will send information about the IP address and extension number via the CSTA interface to the recording system.

The recording system will send an INVITE message to start the recording.

3.3 RECORD ON DEMAND

When this option is used, a function key on the telephone can be used to start the recording of each call.

There are two scenarios:

- When the user presses the recording key on the terminal, the telephone sends a http/https request. The recording system sends INVITE- UPDATE - BYE to the telephone, the same sequence as when active recording is used. The recording is stopped when the call is cleared.
- The calls are recorded from the start. When the user presses the recording key, the http/https request is an order to the recording system to save the recorded call.

When the predefined recording key on the terminal is pressed, a request message based on https/XML/SOAP or http is sent to the recording server. The data to be included in this request is defined in the configuration file for the telephone.