A MITEL
PRODUCT
GUIDE

MiVoice MX-ONE

SIP Extension - Description

Release 7.6 SP1
73/1551-ANF90114 Uen A

February 2024

Mitel
Powering connections

Notices

Trademarks

Contents

General 1

This chapter contains the following sections:

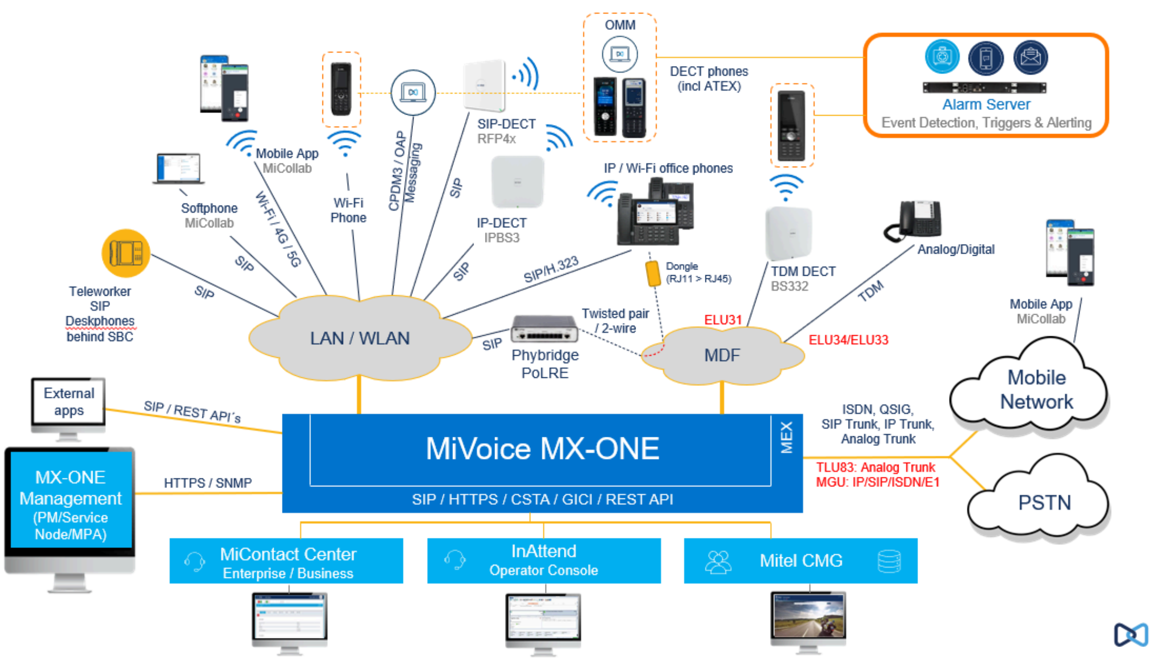
- Glossary

This document describes the functionality of the SIP extension interfaces in the ASP 113 system, i.e. in the MiVoice MX-ONE. It describes the basic functionality of the SIP extension interface, but just mentions some examples of end user services for the SIP extensions.

SIP (Session Initiation Protocol) extension is a facility in the MiVoice MX-ONE Service Node that allows the data network to transmit voice communication. Using TCP/IP the MX-ONE converts voice into packets before transmitting them over the IP network, using the SIP protocol, where they are unpacked at the other end.

For H.323 extensions, see the description *H.323 Extension*.

Figure 1: MX-ONE High-Level Architecture



Users at remote small offices can connect their terminals to the system. The MX-ONE Service Node grants access to and provide services for these terminals.

1.1 Glossary

For a complete list of abbreviations and glossary, see the description for *ACRONYMS, ABBREVIATIONS AND GLOSSARY*.

Integration in the MX-ONE Service Node

2

This chapter contains the following sections:

The SIP extension is fully integrated in the MX-ONE Service Node architecture. This feature allows the SIP terminals to call and be called, as any other type of extension in the MX-ONE Service Node.

For Non-Gateway (NGW) calls between two SIP terminals, the media are directly transmitted from one endpoint to the other.

For Gateway (GW) calls to and from a SIP terminal, the media is transmitted through the MX-ONE Media Gateway. A call between an H.323 and SIP extensions is a GW call.

A detailed explanation of every component needed to make the SIP extension work, is given in the following subsections.

2.1 SIP Extension

The SIP extension feature allows terminals that are compliant with SIP standards, IETF RFCs, to connect to the MX-ONE Service Node. These standards give recommendations for multimedia communications over IP networks. The term “IP extension” includes both H.323 and SIP extensions.

The SIP extension is implemented as a generic extension.

The SIP extension can be either single line access (have only one active call), or multi-line access (have several active calls). Which access type is valid depends on the terminal brand/model, and the configuration.

2.2 SIP Terminals

The name *SIP terminal* is used basically to refer to any SIP-compliant terminal attached to the IP network. It can be either an IP telephone or softphone (PC or mobile client).

The SIP terminals that are compliant to SIP, and additionally support the proprietary protocol to communicate with the MX-ONE Service Node, are called MX-ONE SIP terminals.

The SIP terminal makes use of the mobility concept. That is, the user can log on to the MX-ONE Service Node using any compliant terminal and will get the capability profile defined for this specific user. Once the user has logged on to the system, the terminal becomes a SIP terminal.

The way to log on is by providing the extension number and, optionally, an authorization code (password), which is validated against the one stored in the MX-ONE Server Node. The end user can change the authorization code.

2.3 IP Network

The term IP network here embraces any kind of data network with the TCP/IP protocol, regardless of the underlying type of network. Most enterprises are comprised of a Local Area Network (LAN) for connecting IP devices, such as PCs, servers and printers. The MX-ONE Service Nodes, Media Gateways, and IP terminals are connected to the LAN using 10/100/1000 Mbit Ethernet interfaces. That is, other types of data networks (over FDDI, Token Ring, ATM, and so on) cannot be used to connect directly to these devices.

A corporate IP network can be comprised of several LAN segments, each with its own address range, which can reside on the same physical site or be spread over several locations. Several LAN networks spread over multiple sites can be inter-connected via a Wide Area Network (WAN). This is usually accomplished using routers communicating via service provider connections between the sites. The underlying protocol (xDSL, ATM, MPLS, ISDN, etc...) is irrelevant as long as the routing protocol is IP.

In case a Dynamic Host Configuration Protocol (DHCP) server is set up correctly in the IP network, the server can provide an IP terminal with a network domain name.

2.4 MX-ONE Media Gateways

The different MX-ONE Media Gateways provide media transmission for GW calls (RTP and RTCP handling).

For an overview of the different Media Gateways, see the *MX-ONE SYSTEM DESCRIPTION*.

Functional Overview 3

To completely integrate the IP extension in the MX-ONE Service Node architecture, two functions are needed:

- The **Registrar/Proxy** function.

  The SIP Proxy Server, which handles user registration and location, user capabilities, session setup and session management.
- The **Media Gateway** function.

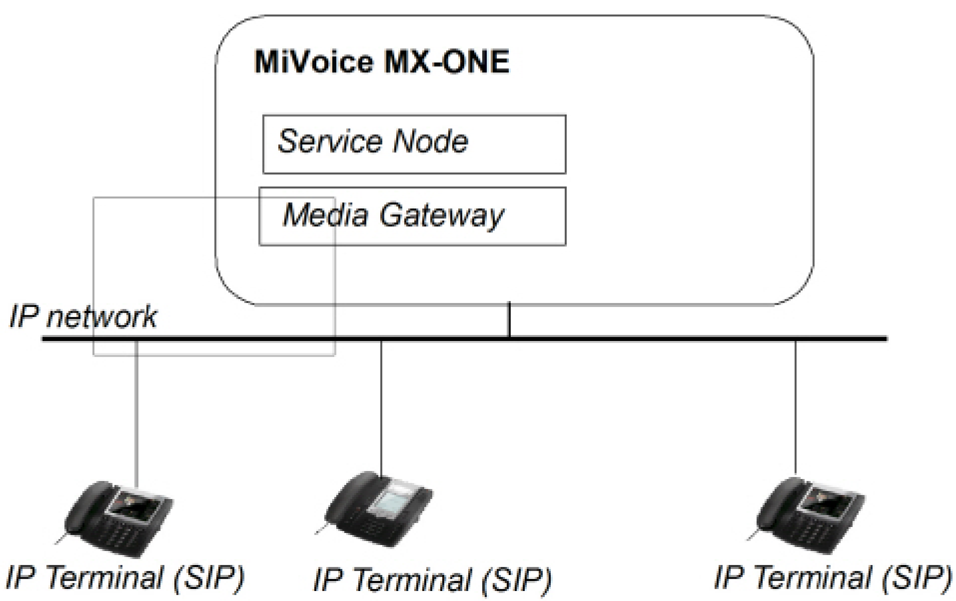
  Provides real-time, two-way communication between SIP terminals on the IP network and other TDM devices and interfaces, or IP terminals when the call is GW.

In the MiVoice MX-ONE the gatekeeper (H.323) and the proxy/registrar (SIP) functions are handled in the same way.

Media (audio or data) transmission in GW calls between a SIP terminal and any other type of extension or trunk is made through the MX-ONE Media Gateway. Media transmission between two SIP terminals can pass directly through the IP network.

The gateway entity is composed either of software and hardware, or if prerequisites allow (i.e. when no TDM lines of types not supported by the Media Gateway are needed), only software, see

Figure 2: System Architecture, with SIP extensions



IP Network Configurations 4

See the description for *MX-ONE SYSTEM PLANNING*, for example in the chapter IP Network Structure.

Note:
The user phones and terminals should normally be located on a subnet, separate from the server and media gateway network or networks.

Facilities, SIP extension 5

This chapter contains the following sections:

- Traffic
- Services

5.1 Traffic

Before a SIP terminal can make a call or receive a call (except for emergency calls), a registration procedure has to be executed by the user. For more information, see Registration on page 9.

If a SIP terminal is logged out or switched off, any call towards that terminal will not progress (the terminal is marked as unavailable in the MX-ONE Service Node).

For GW calls between a SIP terminal and other types of extensions, the speech and 3.1 kHzb audio bearer services are supported (3.1 kHz Audio bearer service is only supported for calls received by the SIP terminal). This means that GW calls between a SIP terminal and other types of extensions can only exchange voice.

For direct media calls between SIP terminals, there are no limitations regarding the bearer services (unless any terminal sets any restriction). The reason is that the media is transmitted directly between the SIP endpoints. This means that calls between SIP terminals within the same IP network can exchange voice, video, and data. However, calls between SIP terminals can be forced to be GW. In such cases, the bearer services are limited by the media gateway. Instant messaging calls (using MSRP) are supported for SIP terminals/clients.

When a SIP terminal receives a call, the MX-ONE Service Node sends to the terminal the own extension directory number and the calling party directory number. The MX-ONE SIP terminals receive additional information from the MX-ONE Service Node that can become visual and audible messages (information related to numbers and names, call progress, service execution). How these messages are handled by the terminal is dependent on its type.

When a SIP terminal initiates a call, the MX-ONE Service Node only informs the terminal about the call progress (call establishment, call release, and so on). As above, the MX-ONE SIP terminal receives from the MX-ONE Service Node additional information that can become visual and audible messages.

5.1.1 Call to a not Available SIP Extension

If a SIP extension has been called, but it is not registered (logged off), switched off or the network link between the extension and the ASP 113 01 is down, the call towards the SIP extension will not progress. If the caller is a SIP extension, it will get a 4XX message unless some re-direction service is activated or suffix services are enabled (in this case, the caller receives a 200 OK message).

5.2 Services

Most of the common telephony services, such as the following are supported for Mitel 6900/6800/6700 SIP extensions. Some of the services are also supported for third party SIP extensions:

- Number and Name Identity Services
- Do Not Disturb (group or individual)
- Extra Directory Numbers (EDNs)
- Forwarding/Personal Number List
- Instant Messaging
- Intercom
- Callback
- Call Parking Pool
- Call Pickup
- Group Memberships
- Pickup Group Monitoring
- Intrusion
- MNS Multiple Representation
- Multiple Terminal Support
- Multiplicity
- Parking
- Shared Call Appearance (SCA), with or without bridging
- Terminal Selection Service
- Transfer, and so on.

For terminals that support proprietary XML, provisioning of the Diversion Monitoring (DMN) feature is also supported. This feature is similar to MNS, with the exception that it will only work on calls which have previously been diverted or deflected from the terminal which has the DMN key to the supervised party (the alerting diversion destination).

See the description for *MX-ONE FEATURE MATRIX* for the services that are available for SIP terminals.

For information about a service available for the SIP extension, see the appropriate description of the service, and also the User Guides for the terminal type.

A list of supported SIP RFCs for SIP extension can be found in the MX-ONE System Description.

Multiplicity

More than one public number can be associated within the MX-ONE Service Node with a SIP remote extension: mobile phone, home, and summer cottage. When using Multiplicity, only one SIP remote extension can be defined with the command `ip_extension`, with many entries defined in the number conversion table for that extension.

All these terminals share the same category profile. Only one of the public terminals is associated with the remote extension that can receive calls at a time. This means that whenever one of the terminals is involved in a call, the rest of the terminals sharing the same directory number cannot receive calls through the PBX.

If there are several terminals associated with one MX-ONE Service Node remote extension directory number, the answering position can be changed through a procedure or CSTA request, that is, calls can be addressed to a terminal, which is different from the one stated in `ip_extension` command.

In this way, the remote extension with multiple terminals associated with the same directory number can decide at any moment to set answering position of the terminal to choose the place where the user is situated. The selected or active answering position is stored in semi-permanent data.

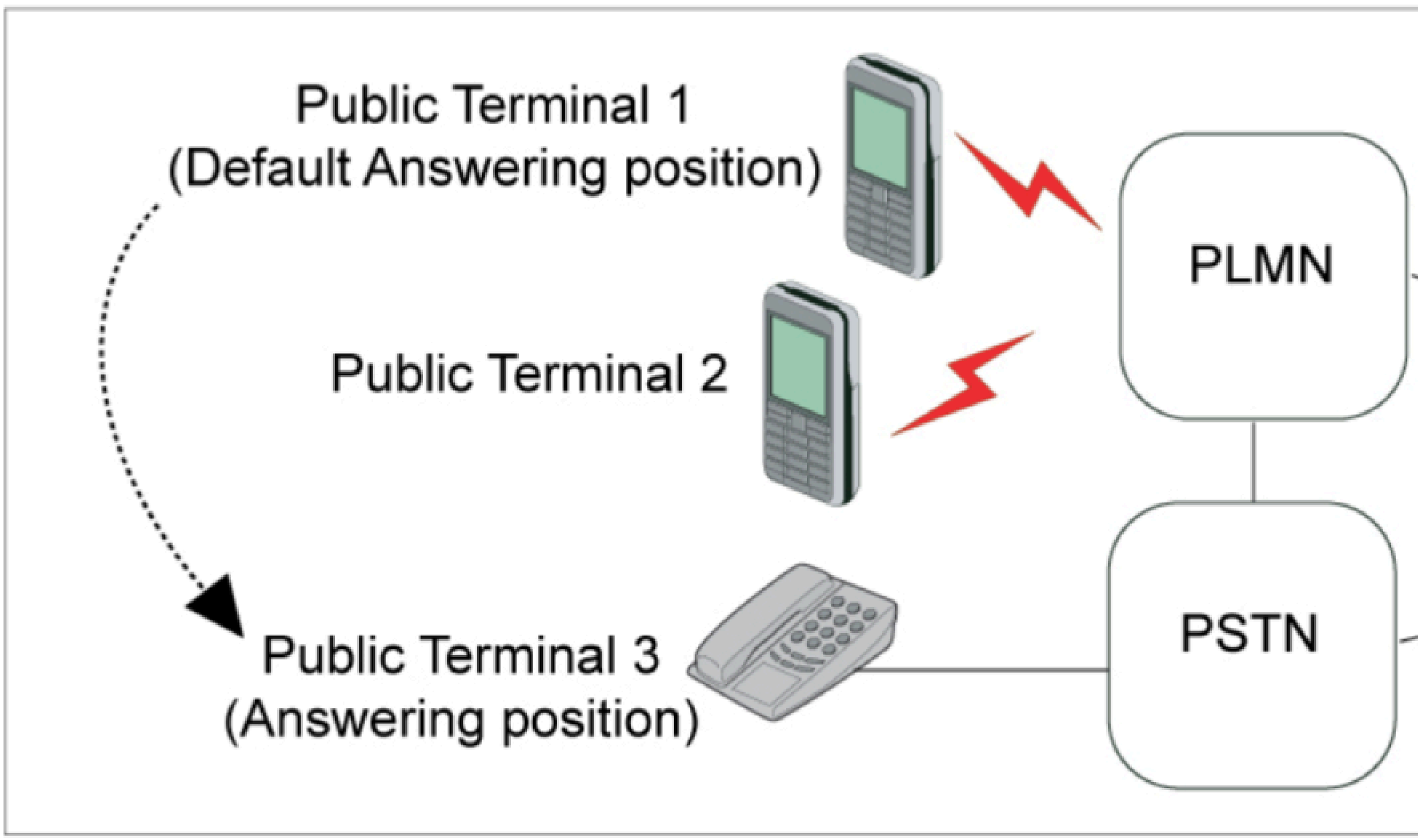
Selected remote extension number does not affect command initiated exchange data.

With a procedure it is only possible to change the answering position if the public calling party number is known. Thus, this facility is only available if the R1/R3 access number is used.

The procedure *FC# (where FC is the Function Code for the application system) is used to set the active answering position to one of the listed public terminals in the list. This procedure is requested from the public terminal which is intended to become the new answering position.

The procedure #FC# is used for the setting of the number defined with `ip_extension` command as answering position for remote extensions with multiple terminals (the default one becomes active). This procedure could be entered from any of the associated public terminals.

Figure 3: Multiplicity - Moving the Answering Position



Procedures, SIP extension 6

This chapter contains the following sections:

Registrar Addressing

The terminal needs to know the server IP address. The server address can either be provided in configuration files or can be discovered by the SIP terminals using DNS_SRV lookup.

6.1 Registration

Before a SIP terminal can make or receive a call, the following steps must be taken:

- The terminal must be initiated in the MX-ONE Service Node as a SIP extension.
- The terminal must know its own IP address. This address may be manually entered or may be acquired through a DHCP server. The DHCP server can also provide information about the name of the network domain the terminal belongs to. The MX-ONE SIP terminals are able to understand this domain name information.
- The terminal must know the IP address of the MX-ONE Service Node where to request access to the system. This is handled by the Home Location Register (HLR) mechanism. See Initiation and Registration Distribution on page 13.
- The terminal must be manually set up with its associated directory number. For Mitel 6900/6800/6700 terminals, there can be several associated directory numbers, Extra Directory Numbers (EDNs). Registration of the EDNs will be done automatically when the own directory number is registered.

  Finally, the SIP terminal sends its PIN code to the exchange to verify its identity, allowing its entry into the system under the user’s request (see Authentication and Authorization on page 10). The access to the system will be granted by the SIP registrar.
- For SIP terminals/clients, it is allowed to register several terminals to the same directory number. When the maximum allowed number is reached, a 'push-out' (de-registration) will be done, primarily of a

terminal of the same type if the terminal supports 'push-out'; otherwise, the registration request will be rejected. See the description for MULTIPLE TERMINAL SERVICE for details.

6.2 Authentication and Authorization

Authenticating a SIP user means verifying its identity if the user is initiated into the system. Authorizing a SIP user means verifying its rights if the user is allowed to access the system. Both processes are always invoked in connection with the registration procedure.

When the SIP user is initiated in the MX-ONE Service Node, the system administrator has to configure an Individual Authorization Code (IAC) for the SIP user directory number, which will be used as the PIN code/ Password. The user of the SIP terminal needs to enter this PIN code to get registered in the system.

The PIN code entered is compared to the IAC stored in the MX-ONE. The registration procedure is successful if the PIN code matches.

Note:
If the SIP user initiated in the MX-ONE Service Node has no IAC assigned to it, then there is no request/check for the PIN code. From a security point of view, it is strongly recommended always to assign a PIN code to the user.

If the authentication and authorization procedures are successful, the user gets the traffic categories associated with the SIP user directory number.

For Mitel 6xxx terminals, at registration, the MX-ONE Service Node will provide the SIP terminal with its directory number and name and information related to services that either are active or can be invoked from the terminal (URI links).

EDNs can have separate passwords (Authorization codes), which differ from the SIP user's password. See the description for the AUTHORIZATION CODE FOR EXTENSION.

6.3 Login Methods for Mitel 6xxx SIP Terminals

Following are the three ways to configure a Mitel 6xxx SIP Terminal by using a username and password for the registration:

- Local configuration through Web GUI or <*mac*>*.cfg*. See Local Configuration on page 11.
- Login using XML on page 11.
- Login using VDP on page 12.

Note:
It is recommended to use the VDP login method for configuring the Mitel 6xxx SIP Terminal.

Local Configuration

This method provides the user number, password, and SIP registrar through device configuration files. The following figure depicts the sample SIP extension registration that uses the credentials from configuration files.

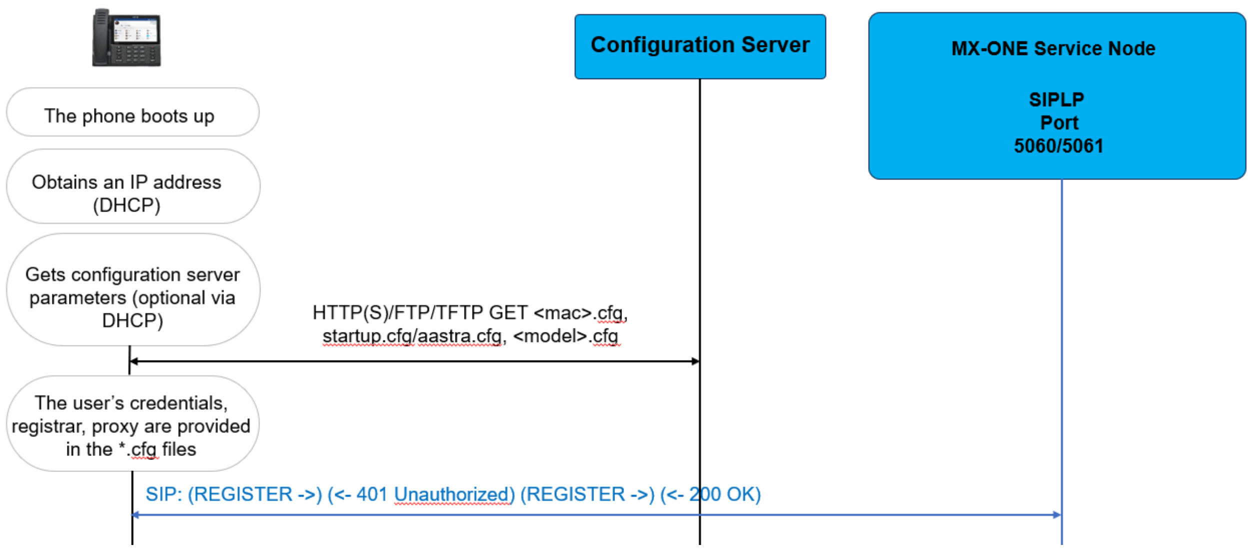


Figure 4: Local Configuration Scenario

Login using XML

This method uses XML API to control the Mitel 6xxx phone's configuration. A login key is configured in the *<model>.cfg*, *startup.cfg*, or *<mac>.cfg* files.

For example,

```
#Classic XML logon
softkey5 label:"LogOn"
softkey5 type:xml
softkey5 value:http://$$ACTIVEPROXY$$:22222/Logon
#when TLS is used:
#softkey5 value:https://$$ACTIVEPROXY$$:22223/Logon
softkey5 states:idle
softkey5 line:1
```

In above example, the key selected is model dependent, in this example 6940 is used.

When pressing the 'LogOn' key, the device calls a script in the service node, and a login dialog is presented, where the user has to enter the user number and password. When the authentication and authorization are successful, the service node pushes via XML payload in HTTP(S) 200 OK message the configuration data to the terminal (SIP Registrar, SIP authentication name, and SIP password). Subsequently, the terminal registers to the registrar with the user and password. After successful registration, the service node pushes via XML payload in SIP NOTIFY message other configuration data like user name and keys.

There is an alternative way to call the initial script for the login dialog, that is configuring the user action at terminal boot up. The following action can be configured in *<model>.cfg*, *startup.cfg*, or *<mac>.cfg* files:

```
action uri startup:http://$$ACTIVEPROXY$$:22222/Startup?user=$$SIPUSERNAME$$
```

When TLS is used:

```
action uri startup:https://$$ACTIVEPROXY$$:22223/Startup?user=$$SIPUSERNAME$
$
```

The following figure depicts the sample SIP extension registration that uses the XML method.

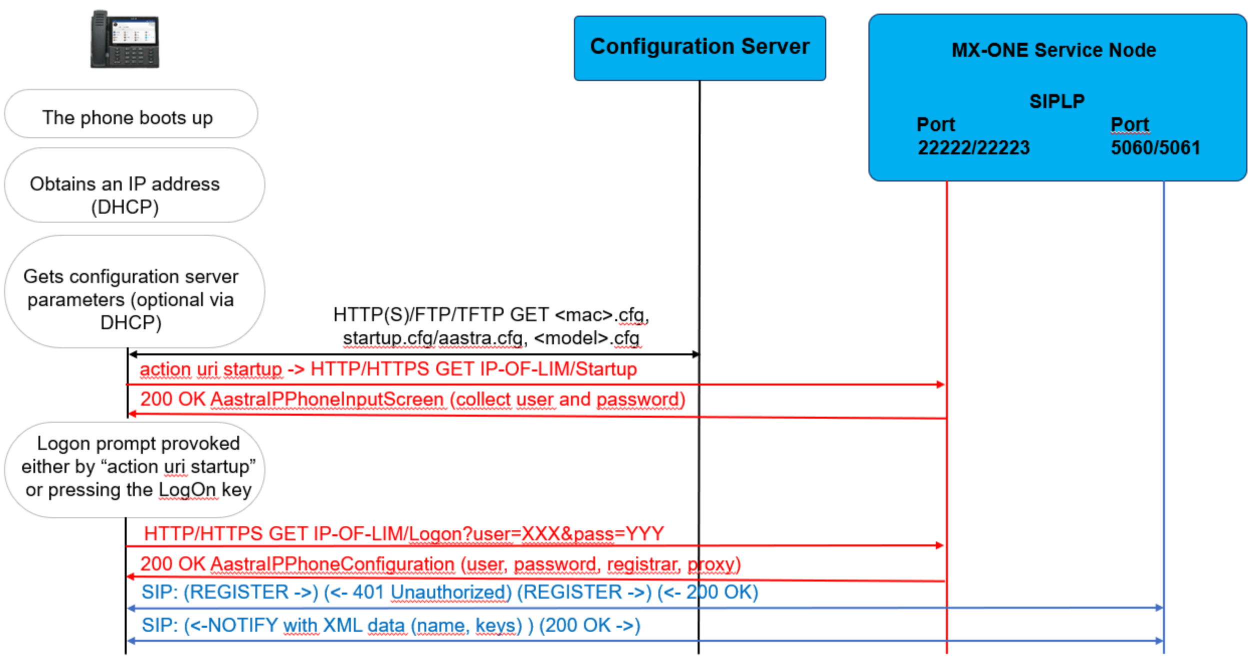


Figure 5: XML Login Scenario

Login using VDP

This method is based on the Visitor Desk Phone (VDP) feature; refer to the *Mitel 6xxx Series SIP Phones Administrator Guide* for basic information. At VDP Logon flow, the phone contacts a configuration server (defined by the *user config url* parameter) and downloads the subscriber's profile contained in the *<user>.cfg* and *<user>_local.cfg* files. To use this method, configure a login key in the *<model>.cfg*, *startup.cfg*, or *<mac>.cfg* files.

For example,

```
#Native login
softkey5 type:hotdesklogin
softkey5 states:idle
```

In addition, activate the VDP settings in one of the **.cfg* file as follows:

```
user config url:http://<lim IP>/limX.<mxone-domain>:22225/vdp
```

```
# when TLS is used, execute the following command:
user config url:https://<lim IP>/limX.<mxone-domain>:22226/vdp
```

In above example, the key selected is model dependent, in this example 6940 is used.

There are also other settings related to VDP; refer to the verified templates provided with the service node (current location: */etc/opt/eri_sn/mitelSIPphones/)* or *Mitel 6xxx Series SIP Phones Administrator Guide*.

When VDP login is configured, XML logon or Startup URIs to ports 22222/22223 must not be used in any configuration file.

When pressing the 'LogIn' key, a login dialog is presented, where the user must enter the user number and password. The device contacts the service node, authenticates, and authorizes. The device gets <user>.cfg file from the service node, and this file has the configuration data needed for registration (SIP Registrar, SIP authentication name, SIP password, and SIP proxy). Subsequently, the terminal registers to the registrar with the user and password. After successful registration, the service node pushes via XML payload in a SIP NOTIFY message other configuration data like user name and keys.

The following figure depicts the sample SIP extension registration using the VDP.

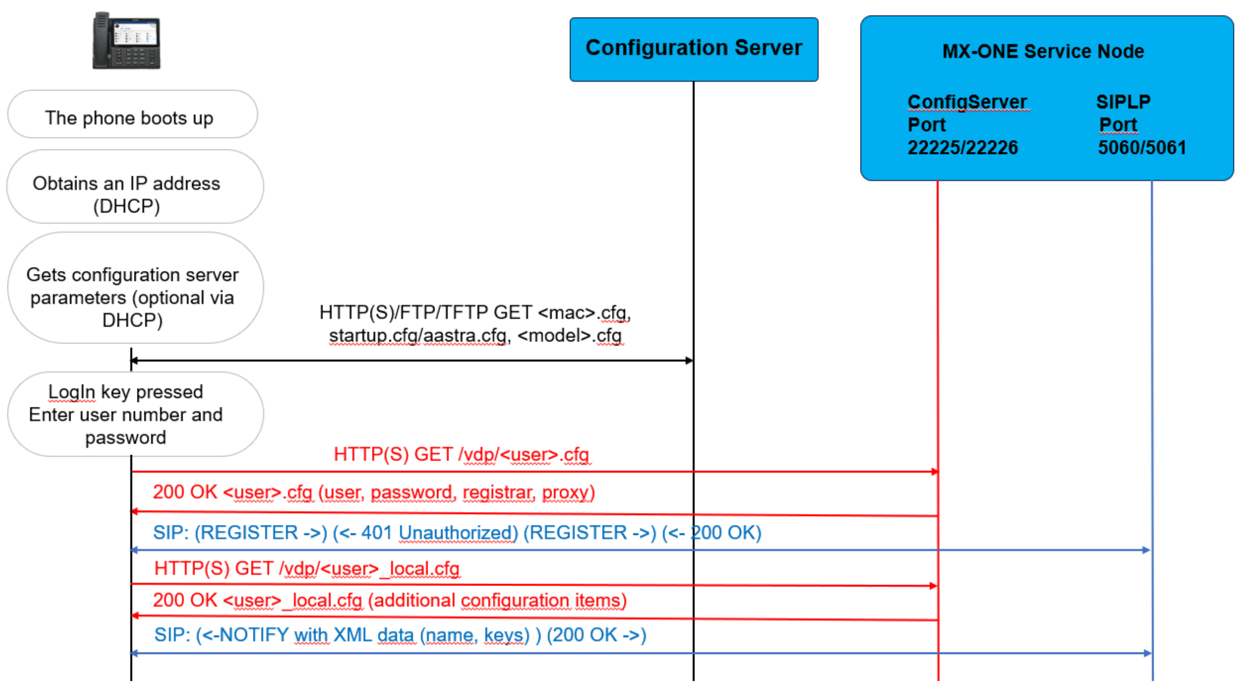


Figure 6: VDP Login Scenario

6.4 Initiation and Registration Distribution

General

The initiation and registration distribution function applies to all generic extensions and thus also here for SIP extensions.

The function has two parts, initiation and registration (logon).

At initiation the HLR can be set to a selected server or, by using the initiation distribution function to let the system decide in which server of a domain it shall be located. At registration the first choice of where to place the User Location Register (ULR) is always in the same server as the HLR. If the optional registration distribution is selected the ULR can be created in an other server of the domain, if the HLR server can take no more registrations. If the servers of the domain are full a server in an other domain, where there is registration capacity, will be selected by the system for creating the ULR.

It is mandatory to use an IP domain when using initiation/registration distribution, even if only one domain is used. It is important to define the IP networks that are part of the domain. There is a default domain for the whole system.

The initiation/registration distribution facility also requires terminals that support the load distribution mechanism. The following models support this type of distribution: the Mitel 6900/6800/6700 SIP telephones.

For initiation/registration distribution to work, the MiVoice MX-ONE should either contain domains with local servers or a centralized server farm where domains do not have any servers locally.

Note:
That the load distribution mechanism described here is quite different from the one used in MX-ONE Version 3.2. Effective distribution requires proper planning.

Initiation

When a generic extension is initiated an HLR is created for this extension in a MX-ONE Service Node. If the initiation/registration distribution function is used, the server selected is based on the capacity of the server and how many registrations that have already been done towards it. The HLR holds the static data that is initiated in the command for the extension.

The SIP extensions are initiated to their geographical domains or explicitly to selected servers.

If there are Service Nodes in the SIP extension domains, they should be set to be part of the domain they reside in. When the SIP extensions are initiated in a domain that contains more than one MX-ONE Service Node, HLRs will be created in the different servers depending on the server capacities. If the maximum HLR capacity has been reached for the domain, a server in another initiated domain will be selected if it has HLR initiation capacity left. If no more HLRs can be initiated an error will be indicated.

In case of a server farm, where all Service Nodes are located in one place and the SIP terminals are located at different domains, there is no HLR capacity in the domains so all HLRs will be created at the server farm. The servers will belong to the default, system domain. The maximum number of terminals in a server can be set by command.

Registration

When an initiated IP extension (here SIP) is logged on to the system a ULR is created in the MX-ONE Service Node. The ULR keeps track of the dynamic data for the extension. The system will try to create the ULR in the same Server as the HLR to minimize the signaling load between these records. If the number of ULRs in a server has reached the maximum value and the initiation/registration distribution function is used, the ULR will be created in another server of the domain where there is free capacity. If no ULR capacity is found in the own domain, the registration will be attempted in an other domain. This will increase the possibility to make successful logons, but as the ULR is allocated in an other domain, the call performance will be degraded.

The ULR load distribution is disabled by default, but can be enabled by command.

Mitel 6900/6800/6700 terminals with Extra Directory Numbers (EDNs) will register all EDNs at the same time as the own directory number, that is, ULRs are created for all associated directory numbers.

Note:
The extension registration distribution for Mitel SIP phones work if the terminal is using either XML or VDP login methods. If terminal credentials (user name/password) are configured from *<mac>.cfg* file, the functionality does not apply.

6.5 Call Admission Control

The lowest common bandwidth between two domains can be set by command.

For a description of the Call Admission Control feature, see the operational directions for *CALL ADMISSION CONTROL*.

6.6 Security

Media encryption according to SRTP is supported for SIP extensions, both for GW and non-GW calls, provided all end-points support SRTP. Three different security policies are supported, and can be selected via O&M.

The supported crypto suites are the standard ones according to RFC 4568, plus a proprietary one corresponding to AES_CM_128 without HMAC_SHA1_80/32. If any other crypto suite is received, the call will be rejected.

Emergency calls will override security policy restrictions.

6.7 Redirect Calls to Marooned SIP Extensions

This procedure allows incoming calls to marooned SIP extensions to be re-directed to a back up answering position defined on a per-extension basis. For more information, see Figure 4.

The typical scenario where this function is useful is branch offices connected to the main PBX by means of an IP link.

If no special provisions are made, the SIP extensions in the branch office will be unable to make or receive calls if the IP link between the branch office and the main office goes down.

Nevertheless, it is possible to provide voice service to users in the branch office when the IP connection is down, according to the following criteria:

- A backup answering position number must be provided for SIP extensions located at the branch office. These backup positions must be public network numbers. These numbers must be handled by a branch node installed on the branch office premises.
- SIP extensions at the branch office must be configured to periodically re-register to the MX-ONE Service Node. If a SIP extension fails to re-register on time, the MX-ONE Service Node will consider it to be marooned and will redirect incoming calls to its backup answering position.
- SIP terminals (Mitel 6900/6800/6700) at the branch office, when getting no reply to the reregistration requests sent to the MX-ONE Service Node, will automatically register to the branch node. Other SIP terminals must be manually registered on the branch node, or accept calls directed to the SIP telephone from static configuration.
- The branch node must be configured to associate every SIP extension with its corresponding backup answering position as defined in the MX-ONE Service Node. The branch node must also be configured to have the same numbering plan as the MX-ONE Service Node so that users can dial the same directory number to reach a given destination regardless of whether they are registered in the MX-ONE Service Node or the branch node.

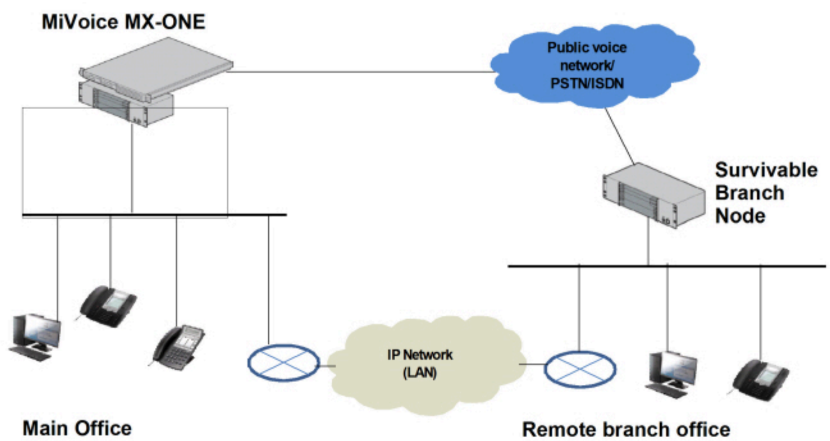


Figure 7: Branch Office with Marooned Extensions Scenario

Summarizing, the redirect calls to marooned SIP extensions functionality, combined with properly configured SIP terminals and a branch node on the branch office premises, make it possible for users of SIP extensions at the branch office to make calls and to receive them on their usual directory number when the IP connection between the branch office and the main office is down.

Note:

For a SIP extension to be tagged as marooned, it must fail to re-register within the agreed-upon time span. This means that, for a SIP extension to become marooned, it must be registered when the network connection goes down.

6.8 Area Code per Extension

It is possible to associate a home area code to a SIP extension during extension initiation. This facility allows, in conjunction with the Least Cost Routing functionality, modification of the number dialed by a SIP extension in order to route the call properly.

The typical scenario where this facility is useful is a branch office scenario, see Redirect Calls to Marooned SIP Extensions on page 15, where in addition it is required to route the public calls through the local branch node at each branch office, but dialed public numbers may contain no area code.

The home area code is added to the dialed public number by means of the LCR Number Length Table. For details on Least Cost Routing see the extra facility description for *LEAST COST ROUTING*. The home area code is, in this case, the area code associated to the calling IP extension. For the Branch Office scenario with area code, see Figure 5.

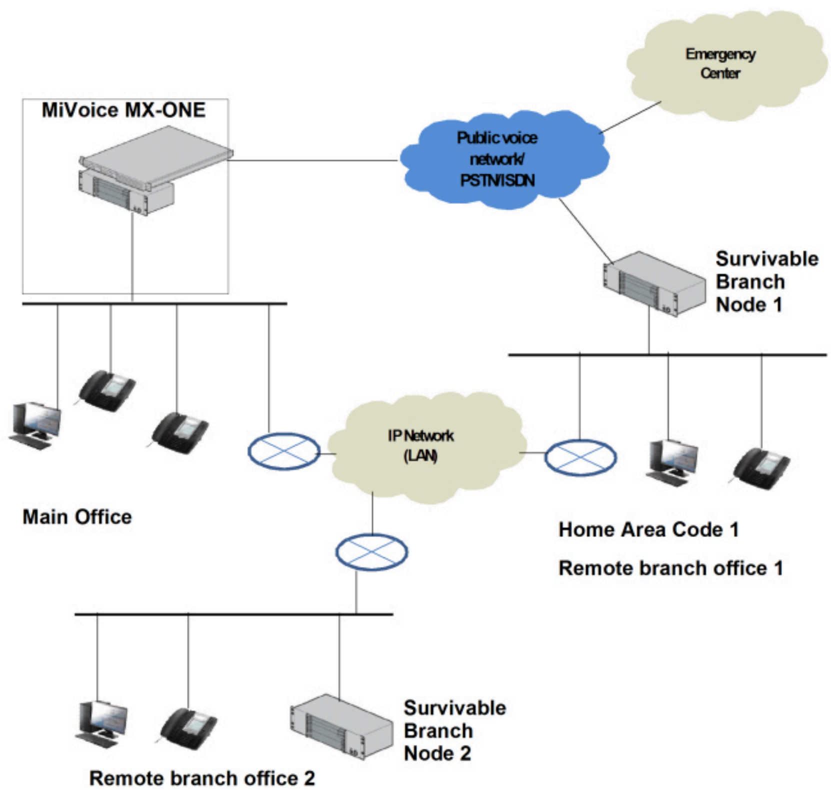


Figure 8: Branch Office scenario with Area Code

Having IP routes between the main office (where the calling SIP extension is registered) and the branch node at each branch office, and having different destinations related to the area codes, the call can be routed towards the branch node in the calling party's branch office due to the added area code. The branch node is set up so the call is routed to the PSTN.

The home area code only takes effect if the LCR access code (LAC) and the LCR tables are set correctly.

If the calling party is one of the following the LAC and LCR tables are set to translate public dialed numbers:

- a non-IP extension
- an IP extension that does not have an associated home area code

The area code that will be added is the one associated to the calling party LIM. See the description for *LEAST COST ROUTING*.

6.9 Area Code per Domain

It is possible to associate an area code to a network domain. This facility allows, in conjunction with the Least Cost Routing functionality, modification of the number dialed by an IP extension in order to route the call properly. The domain area code works similarly to the home area code with the exception that only domain area code is used for Emergency Calls.

6.10 Call Park Pool

For SIP extensions, the Call Park Pool feature is available, i.e. a call in speech can be parked ‘remotely’, by transferring it to a dedicated hunt group member, from which the call can be picked (answered) from any extension in the same system. A recall to the parking party can be done if the call is not picked within a few minutes.

See the operational directions for *CALL PARK POOL* for details.

6.11 Emergency Calls, SOS Calls

This facility enables any registered SIP terminal type to make emergency calls, SOS calls, to an emergency center. The Mitel 6900/6800/6700 SIP terminals are able to make emergency calls even when they are logged off from the exchange.

It is also possible for the emergency center to dial back to the SIP telephone which calls to the emergency center or to a pre-defined central answering position.

For further information, see operational directions for *EMERGENCY CALLS, SOS CALLS*.

6.12 Extra Directory Numbers

For SIP extensions (Mitel 6900/6800/6700 models), the EDN feature is available, that is, a number of extra directory numbers can be associated with one SIP phone. These EDNs are sort of extra ‘telephones’ in the terminal.

EDN is a directory number that is defined on a specific key on a SIP terminal. This key has, with a few exceptions, the same characteristics, features and Classes of service, as the terminal's own directory number and operates as a telephone in its own right.

As default, EDNs are not manually logged on/off. When the user logs on/off the Line1 (own directory number), all EDNs will also be logged on/off.

If a re-direction service is activated for Line1, then Do Not Disturb (with a possibility to configure a re-direction due to DND) is automatically activated for the EDNs.

The state of the ODN and other EDNs when the user has an ongoing conversation on one of his/her EDN lines, is that only that line is busy. The other lines are not regarded as busy.

You can associate an extra directory number to only one user number.

Note:
EDNs for SIP extensions are similar to the ADNs of DTS, but there are functional differences. For a multiple representation feature, the SCA feature is used instead of MDN. The EDNs do not support a 'Multi Member Busy' feature, i.e. the line states of the different EDNs and ODN are independent.

6.13 HLR Redundancy

HLR backup/HLR redundancy is a feature that provides back-up registration of certain IP extensions, when the ordinary HLR no longer can be accessed. After the process (called change-over) has taken part, a different (LIM) server hosts the temporary back-up HLR. To be able to create the backup HLR, an external system database (Cassandra) with replication functions is used. The ULR will register towards the backup HLR.

If the ordinary HLR recovers, a re-registration (change-back) towards this HLR will be performed. In order to not overload the server of the ordinary HLR, the re-registration will be done with a delay and distribution in time.

There is no need for the user to start the change-over or the change-back process, when HLR redundancy is activated. The system or the terminal will detect if the conditions for the change-over (or the change-back) process is fulfilled.

The conditions for change-over are the following:

- Ordinary HLR server (LIM) is out of order
- Isolation of ordinary HLR server (LIM) isolation
- Manual blocking of entire server (LIM)

Some services, like group functions and busy/queue functions are lost while registered to the backup HLR. Some services that depend on common/centralized resources may also be lost depending on configuration.

The HLR backup feature is activated/deactivated on system level, using commands. The default setting is inactive state.

For further information, see the description for *HOME LOCATION REGISTER REDUNDANCY*.

6.14 Intercom

For SIP extensions (Mitel 6900/6800/6700 models), an Intercom function is available. The function is configured using the EDN, Hotline and automatic answer services.

The Intercom feature can be assigned to an IP extension (SIP-compliant endpoints) that supports Intercom functionality. When the user presses a line key with the Intercom function assigned, a call will be set up directly to the pre-defined destination, which must be another SIP extension.

The call is indicated as an internal call, but with no display updates before the call is answered. If the called party is busy with another call, it will only receive one muted ring burst. The LED for the Intercom key will be lit and flash.

Automatic, immediate answer is available on Mitel 6900/6800/6700 SIP terminals, with appropriate configuration. If the called Intercom line has been programmed for immediate answer, the call will enter a speech/hands-free mode directly.

The caller will get one ordinary ring tone, and the called party a specific ‘automatic answer indication tone’ (instead of ring signal), and after the tone/signal, speech connection is established.

Note:
Also, DTS/ADN with a similar function can be used together with the SIP extensions, but the DTS will not have the muting function. For some third party SIP terminals Intercom may also be possible, if appropriate automatic answer and hotline functionality is supported.

6.15 Shared Call Appearance

For SIP extensions (Mitel 6900/6800/6700 models), the Shared Call Appearance or Shared Line service is available, that is, the directory numbers (own) can be represented and monitored in other SIP phones. See the description for *SHARED CALL APPEARANCE* for details.

Capacities and Limitations, SIP Extensions

7

This chapter contains the following sections:

Only one IP network, or two at redundancy, can be connected to a particular MX-ONE Service Node.

The SIP terminals can only be addressed by their directory number. Other types of addressing such as E-mail address and URL address are not allowed in MX-ONE Service Node for SIP terminals.

The SIP terminals cannot be included in an ACD group, but can be agent phones in a CTI group (when CSTA monitored by a Call Center application).

The Registrar/Proxy address must either be manually set in the IP terminal's configuration file, or found via configured DNS SRV records. This is true also when the SIP terminals support the associated protocol messages, for example, multi-cast. However, as load distribution procedures are supported, the SIP terminal may be registered in another LIM (Registrar/Proxy) than in the LIM whose address was manually set.

Load distribution requires that the different domains either contain terminals and servers or, for the server farm case, only terminals.

MNS keys are supported by the following MX-ONE SIP phones: Mitel 6900/6800/6700. Capacities and display capabilities vary for the different Mitel 6900/6800/6700 models.

Forking supports up to 8 SIP terminals registered to the same directory number.

7.1 User Agent Profiles

Some configuration items are User Agent (device) specific and present in user agent files in */etc/opt/eri_sn/<version>/sip_user_agents/*. If changes in those files are needed, those shall be made after consulting with Mitel Technical Support.