MiVoice MX-ONE

Remote Extension over SIP - Operational Directions

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Introduction

The function Remote Extension over SIP makes it possible to have public terminals as extensions in the MX-ONE Service Node. The feature is implemented as generic extensions in the exchange and can be adapted to the legacy mobile extension protocol which uses R1/R3 numbers. The Remote extension can be any public subscriber number.

A Remote Extension is an IP Extension, identified by the MX-ONE directory number, DIR, which is tagged with the remote number to the external communication system (handled by *ip_extension --uri tel:*). The remote number is the phone number on the other end of the SIP route (handled by *ip_extension --uri rou*). The destination to DIR is the SIP route access point and the number sent to this access is the remote-number.

The call to and from the communication system will use the protocol dictated by the type of SIP trunk, defined by the command `sip_route`, where the parameters, *-registertype* and *-trusted* differentiate the trunk type. (See *RFC 3324*).

This document contains operational directions for remote extensions, that is, the procedures for initiating and removing remote extensions over a SIP trunk. For reference, see the description for *REMOTE EXTENSION*, which describes similar operations over ISDN.

Limitations

The Mobile and Remote extension over SIP is handled like an extension in the system, and does not support trunk related telephony traffic services, like alternative routing.

Target Groups

This document is intended for personnel performing initiation and removal of mobile and fixed remote extensions.

Glossary

For a complete list of abbreviations and glossary, see the description for Acronyms, Abbreviations, and Glossary.

Prerequisites

Verify with the command `license_status` that the MX-ONE system has a valid licenses for using a SIP trunk. The following licenses are needed: AUTOMATIC-REGISTRATION, EXTERNAL-LINE-SIP, and IP-EXTENSION.

If a license is missing or needs to be upgraded, contact the purchase office where the MX-ONE system was bought. When a new license file is received, see the operational directions for *ADMINISTRATOR USER'S GUIDE* section LICENSE HANDLING.

A SIP trunk that supports SIP remote extensions should be installed for connection to the PSTN or PLMN.

Tools

An I/O terminal is used to enter the commands.

Workflow

Initiating SIP Remote R1/R3 Extensions

Use R1 number when dial tone is wanted, normal remote extension. Use R3 number when no dial tone is wanted; used when CSTA application shall control further actions; for example, to make a call request.

The following steps describe the workflow for initiating a remote extension:

1. Initiate a SIP route, see *Route and Trunk*.
2. If needed, initiate the R1/R3 access numbers, see *Initiating Access Numbers in the Number Series*.
3. Initiate a remote extension, see *Initiating a Remote Extension*.
4. Initiate number conversion data, see *Initiating Number Conversion Data*.
5. If desired, name the remote extension, see *Initiating a Name Identity for the Remote Extension*.
6. Initiate the personal number, see *Personal Number*.
7. Initiate the Recorded Voice Announcements (RVAs), see *Initiating a Recorded Voice Announcement*.
8. Increase the callback time, see *Callback or Recall Time*.

Removing SIP Remote R1/R3 Extensions

The following steps describe the workflow for removing a remote extension:

1. Remove number conversion data, see *Removing Number Conversion Data*.
2. Remove the name for the remote extension, see *Removing_a_Name_Identity_for_the_Remote_Extension*.
3. Remove the personal number, see *Personal Number*.
4. Remove the remote extension, see *Removing_a_Remote_Extension*.

The following steps only apply when no more remote extensions are initiated in the system:

1. If used, remove the R1/R3 access numbers, see *Removing a Number from a Series*.
2. Remove the Recorded Voice Announcements, see *Removing a Recorded Voice Announcement*.
3. Remove the trunks, see *Removing a Route or Trunk*.
4. Decrease the callback time, see *Callback or Recall Time*.

Initiating SIP Remote R2 Extensions

The following steps describe the workflow for initiating a remote extension:

1. Initiate a SIP route, see *Route and Trunk*.
2. If needed, initiate the R2 access numbers, see *Initiating Access Numbers in the Number Series*.
3. Initiate a remote extension, see *Initiating a Remote Extension*.

4. Initiate Authorization data, see *Initiating an Individual Authorization Code*.
5. If desired, name the remote extension, see *Initiating a Name Identity for the Remote Extension*.
6. Initiate the personal number, see *Personal Number*.
7. Initiate the Recorded Voice Announcements (RVAs), see *Initiating a Recorded Voice Announcement*
8. Increase the callback time, see *Callback or Recall Time*.

Removing SIP Remote R2 Extensions

The following steps describe the workflow for removing a remote extension:

1. Remove Authorization data, see *Removing_the_Individual_Authorization_Code*.
2. Remove the name for the remote extension, see *Removing a Name Identity for the Remote Extension*.
3. Remove the personal number, see *Personal Number*.
4. Remove the remote extension, see *Removing a Remote Extension*.

The following steps only apply when no more remote extensions are initiated in the system:

1. If used, remove the R2 access numbers, see *Removing a Number from a Series*.
2. Remove the Recorded Voice Announcements, see *Removing a Recorded Voice Announcement*.
3. Remove the trunks, see *Removing a Route or Trunk*.
4. Decrease the call back time, see *Callback or Recall Time*.

Procedures

Route and Trunk

Initiating SIP Route

General

A direct route (mobile direct access) between the PLMN and the PBX offers possibilities for the public terminal (remote extension) to request PBX services in the same way as for other generic extensions.

Execution

1. Initiate the SIP route, see the command description for `sip_route`.
2. Initiate the route parameters. Use the command `ROCAI`, followed by the command `RODAI`.
3. Enter the commands `ROCAP` and `RODAP` to verify the result.
4. Enter the command `ROEQI` to finalize the route initiation.
5. In *ROEQI* specify the same LIM number (X in TRU=X-1) as used when initiating the directory number in command extension `-i --lim`. Calls for remote extensions will always be sent from the `LIM` set in command extension. TRUs may also be set for other LIMs.

For reference, see the operational directions for *ROUTE DATA, RO*.

Removing a Route or Trunk

1. Remove the trunk lines by entering the command `ROEQE`.
2. Remove the route by entering the command `sip_route -remove`.

For reference, see the operational directions for *ROUTE DATA, RO*.

Number Series

Initiating Access Numbers in the Number Series

General

Only set R1/R3 if this is used for Mobile Extensions (not the generic Remote Extension) and if the ISP requires this.

The R1/R3 access number, which is used when the A-number is sent to the exchange, should be initiated for calls from a remote extension. The R1/R3 access number is optional, though. The access number should be within the direct in-dialing numbering plan that allow full access to the MX-ONE Service Node functionality.

Execution

1. Verify that the R1/R2/R3 number is not used by entering the command `number_print`.
2. Initiate an access number with command `number_initiate -numbertype` (r1 or r3)/r2.

   This number must be included in the direct in-dialing numbering plan.
   As the remote number is exactly what is sent to the Communication System hosting this directory number, the R1/R2/R3 number must be part of the remote-number if it is defined.

Removing a Number from a Series

1. Enter the command `number_end` to remove the R1/R2/R3 number from the number series.
2. Enter the command `number_print` to verify the result.

Remote Extension

Initiating a Remote Extension

General

A generic extension directory number is initiated as a remote extension by the command `ip_extension`.

More than one SIP trunk can be stated per Remote Extension to get redundant targets towards the ISP.

Prerequisites

The directory number must be present in the extension number series (command `number_initiate`) and initiated as a generic extension (command `extension`) with an appropriate Common Service Profile (command `extension_profile`). See the operational directions for *GENERIC EXTENSION*.

Execution

1. Enter the command `extension -d -p` to verify the directory number data.
2. Enter the command `ip_extension -d --terminal-identity --uri`, where terminal-identity has the form ¡§user@host¡¨ and uri specifies the destination number with a fictive ip-address, and the remote number Example *--uri dest:xx@y.y.y.y;remote-number=zzzz¡*.

Removing a Remote Extension

General

A generic extension directory number is ended as a SIP remote extension by the command `ip_extension -e`. If the remote extension has an ongoing call, the removal will be denied. Once the remote extension has been removed, the directory number is ready to be assigned again to any type of generic extension.

Prerequisites

If the remote extension has an ongoing call, the call must be ended.

Execution

1. Enter the command `ip_extension -p` to check that the directory number has been initiated as an IP terminal
2. Enter the command `ip_extension -e` to terminate the directory number as a remote extension.
3. Enter the command `ip_extension -p` to verify the result

Number Conversion Data

Initiating Number Conversion Data

General

The received calling party number should be validated and transformed to the remote extension number by means of the number conversion function. If multiple terminals are associated to the remote extension, a number conversion must be initiated for each terminal.

Execution

1. Key the command `ip_extension -p` to verify that it is a remote extension. For example, that URI is set.
2. Does the remote extension already have number conversion data? If YES, go to End.
3. Enter the command `number_conversion_initiate -conversiontype` 6 to initiate number conversion for the remote extension.
4. The parameter *-pre* must be equal to the remote extension number. Parameter *-truncate* must be equal to the number of digits in the parameter *-entry*.
5. Enter the command `number_conversion_print` to verify the result.

Removing Number Conversion Data

1. Enter the command `number_conversion_end` to remove the number conversion data for the relevant entry. Only the conversions with relevant entry and with conversion type 6 should be removed.
2. Enter the command `number_conversion_print` to verify the result.

Initiating an Individual Authorization Code

1. Use the command `auth_code -i`, specify the parameter *-dir*.
2. Use the command `auth_code -p` to verify that the function has been initiated.

Removing the Individual Authorization Code

Use the command `auth_code -e`.

NOTE: *The system administrator has to ensure that RAC is not erased for a secure extension, with SECEXC = NO.*

Use the command `auth_code -p` to verify the function.

Name Identity for the Remote Extension

Initiating a Name Identity for the Remote Extension

General

The command `name -i` is used to associate a name with an individual, and it is specified using the *--name1* and *--name2* parameters. See the command description for *NAME IDENTITY*.

Execution

1. Enter the command `name -p` to obtain a printout.
2. Initiate a name identity for the remote extension. Enter the command `name -i`. See the command description for *name*.
3. Enter the command `name -p` to verify the result.

Removing a Name Identity for the Remote Extension

1. Enter the command `name -e` to remove the name for the remote extension.
2. Enter the command `name -p` to verify the result.

Personal Number

The remote extension number should be the personal number. This number should be the first choice in the first list. This is done in order to minimize the extending time for the operator when extending to a remote extension (to the public network). For information about this procedure, see the operational directions for *PERSONAL NUMBER*.

Recorded Voice Announcements

Initiating a Recorded Voice Announcement

General

A welcome announcement can be provided for individual calls to the calling party when the call is made to a remote extension. An RVA announcement is activated by using the command RACEI.

Execution

For information on RVA commands and parameters, see the operational directions for *RECORDED VOICE ANNOUNCEMENT, RA*.

1. Is an RVA going to be initiated to prompt the user to enter PIN code? If NO, go to End.
2. Enter the command `RACEI` to state the vocal guidance traffic case and the vocal guidance announcement number.
3. See the parameter description for *RECORDED VOICE ANNOUNCEMENT, RA*.
4. Enter the command `RACEP` to verify the result.

Removing a Recorded Voice Announcement

When a remote extension is removed, it is not necessary to remove the RVA, but it is recommended if all remote extensions are removed. See the operational directions for *RECORDED VOICE ANNOUNCEMENT, RA*.

Original A-Number

If it is possible to receive the original calling party number (A-number), the remote extension could show the calling party's number. The Original A-number feature gives the possibility to transfer the original calling party number to an external party. There are no configurations steps needed, the feature is always enabled, as the SIP remote extension is treated as an internal user. Limitations at the SIP trunk provider may apply.

Callback or Recall Time

The standard time to answer a callback recall in MX-ONE Service Node is eight seconds. As the time to set up the connection to the remote extension can be longer than the standard time, the callback time should be increased. Use the command `ASPAC`*(PARNUM=29)*.

Transfer Service

The transfer service can be invoked by the user going on-hook or pressing a suffix digit. The transfer service is requested (on-hook or suffix digits) by the command `ASPAC`*(PARNUM=217)*. The suffix digit is also selected with *PARNUM = 217*.

Inquiry

The Inquiry service can be invoked by the user by pressing a suffix digit procedure. The suffix digit procedure is set by the command `ASPAC` *(PARNUM = 124)*.

Status Printout

To get a status printout, see the operational directions for *SYSTEM USER INFORMATION*.

Examples

Example 1

This section includes an example showing the steps to initiate a remote extension over a SIP trunk.

R1 and R3 number provide the same service, except that R1 provides dial tone and R3 does not.

In following examples, the number type R1 stated 9999. When 9999 number is dialed, then a dial tone is provided.

When no dial tone shall be provided after completion, use number type R3 instead.

In commands below use *-numbertype* r3 instead of *-numbertype* r1.

Table 6.1: Remote extension over SIP trunk, example

Variable	Example Value
<R1_number>	9999(optional)
<Remote_extension_number> (the MX-ONE directory number)	479
<Terminal ID>	479@c-253.btrunk.se(host part received from network operator)
<URI>	dest:00@1.1.1.1;remote-number= +468719001234567 **NOTE:** Ip-number is not used, but is needed for syntax check.
<Route_access_code>	00(used for normal traffic)
<External_number_length>	14(route access code length + remote number length)
<LIM> (where the SIP route is located)	3
Security domain	abcip.ip.oper.se(received from network operator)

- **Initiate Configure SIP settings for the route**

sip_route -set -route 77 -profile 'TeliaEntry' -uristring0 'sip:!?@10.105.64.41' -uristring1 'sip:1000@10.105.64.43' -fromuri0 'sip:?@10.105.64.43' -fromuri1 'sip:+?@10.105.64.43'

If needed, add an optional string in fromuri0 (to be placed after sip: to replace!).

sip_route -set -route 77 -rexstring "666”

- **Initiate the route parameters**

Set the route categories.

ROCAI:ROU=77,SEL=7110000010000010, SIG=0111100000A0,TRAF=03151515,TRM=4, SERV=3100030001,BCAP=111111;

RODAI:ROU=77,TYPE=TL66,VARC=00000000,VARI=00000000, VARO=00000000;

Finalize the initiation. Set SIP registration for this LIM.

ROEQI:ROU=77,TRU=3-1,INDDAT=000000000001;

Initiate the destination.

RODDI:ROU=77,DEST=00,SRT=3;

- **Initiate the redundancy route**

If needed add more routes to be used for alternative routing. (see *Route Data, RO - Operational Directions, 49/154 31-ANF 901 14,*

section: Initiate Alternative Route Choices to an External Destination

- **Initiate the extension categories**

extension_profile -i --csp 1 --ext-traf 0103151515 --ext-serv 212132110001000000000000000000

--ext -cdiv 111000001110000 --ext-roc 000001 --ext-npres 0011000

- **If needed, initiate the R1 access number for the remote extension**

number_initiate -numbertype r1 -number 9999

- **Initiate the directory number for the remote extension**

number_initiate -numbertype ex -number 479

- **Initiate the remote extension**

extension -i -d 479 --lim 3 --csp 1 ip_extension -i -d 479 -terminal_identity 479@c-253.btrunk.se --uri "dest:00@1.1.1.1;remote-number=+468719001234567"

- **Initiate the number conversion data**

number_conversion_initiate -entry 468 -conversiontype 2 -numbertype 1 -newtype 5 -truncate 3 -pre 00

number_conversion_initiate -entry 00719001234567 -conversiontype 6, -truncate 14, -pre 479

NOTE: In this, the remote-number is an example, the number the ISP is expecting in e164 format when calling the remote extension. The number received at incoming call may use a different type of number, so normally the incoming number conversion (conversiontype 2) transforms this number to the internal dial plan (TON 5) before doing the -conversiontype 6 check.

Example 2

Advanced configuration with multiple remote extensions over SIP trunk.

1. Initiate R1 number
   -numbertype R1 -number 23000
2. Initiate 2 SIP-remote Terminals (SIP-route, CSP and Number data must be configured in advance)

   extension -i -d 68005 --csp 2 -l 1 --max-terminals 2
   ip_extension -i -d 68005 --terminal-identity "sip:68005@192.168.32.12" --uri "dest:00@192.16832.12;remote-number=01421163"
   ip_extension -i -d 68005 --terminal-identity "sip:68005@192.168.32.13" --uri "dest:00@192.168.32.13;remote-number=01421161"
   number_conversion_initiate -conversiontype 6 -pre 68005 -truncate 8 -entry 01421163
   number_conversion_initiate -conversiontype 6 -pre 68005 -truncate 8 -entry 01421161
3. Enable Forking

   parallel_ringing -i -d 68005

Example 3

Advanced configuration with multiple remote extensions over SIP trunk.

1. Initiate R3 number
   -numbertype R3 -number 23000
2. Initiate 2 SIP-remote Terminals (SIP-route, CSP and Number data must be configured in advance)

   extension -i -d 68005 --csp 2 -l 1 --max-terminals 2
   ip_extension -i -d 68005 --terminal-identity "sip:68005@192.168.32.12" --uri "dest:00o@192.168.32.12;remote-number=01421163"
   ip_extension -i -d 68005 --terminal-identity "sip:68005@192.168. 32.13" --uri "dest:04@192.168.21.13;remote-number=01421163"
   number_conversion_initiate -conversiontype 6 -pre 68005 -truncate 8 -entry 01421163
   number_conversion_initiate -conversiontype 6 -pre 68005 -truncate 8 -entry 01421161
3. Enable Forking

   parallel_ringing -i -d 68005

Termination

Inform the department or person responsible for telephony matters if any alteration is made.

If any exchange data have been changed, a dump to backup media must be performed, see the operational directions for *ADMINISTRATOR USER'S GUIDE*.

Powering connections

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