

# Remote Extension over SIP

OPERATIONAL DIRECTIONS



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## 1

## 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, *-register-type* 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.

## 1.1

## 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.

## 1.2

## TARGET GROUPS

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

## 1.3

## GLOSSARY

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

## 2

## 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.

## **3 TOOLS**

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

## 4

## WORKFLOW

### 4.1

### INITIATING SIP REMOTE R1/R3 EXTENSIONS

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

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.

1. Initiate a SIP route, see 5.1 Route and Trunk on page 9.
2. If needed, initiate the R1/R3 access numbers, see 5.2.1 Initiating Access Numbers in the Number Series on page 10.
3. Initiate a remote extension, see 5.3.1 Initiating a Remote Extension on page 11.
4. Initiate number conversion data, see 5.4.1 Initiate Number Conversion Data on page 11
5. If desired, name the remote extension, see 5.5.1 Initiating a Name Identity for the Remote Extension on page 13.
6. Initiate the personal number, see 5.6 Personal Number on page 13.
7. Initiate the Recorded Voice Announcements (RVAs), see 5.7.1 Initiating a Recorded Voice Announcement on page 14.
8. Increase the callback time, see 5.9 Callback / Recall Time on page 14.

### 4.2

### REMOVING SIP REMOTE R1/R3 EXTENSIONS

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

1. Remove number conversion data, see 5.4.2 Removing Number Conversion Data on page 12
2. Remove the name for the remote extension, see 5.5.2 Removing a Name Identity for the Remote Extension on page 13
3. Remove the personal number, see 5.6 Personal Number on page 13
4. Remove the remote extension, see 5.3.2 Removing a Remote Extension on page 11

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 5.2.2 Removing a Number From a Series on page 10
2. Remove the Recorded Voice Announcements, see 5.7.2 Removing a Recorded Voice Announcement on page 14
3. Remove the trunks, see 5.1.2 Removing a Route or Trunk on page 9
4. Decrease the call back time, see 5.9 Callback / Recall Time on page 14

## 4.3

## INITIATING SIP REMOTE R2 EXTENSIONS

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

1. Initiate a SIP route, see 5.1 Route and Trunk on page 9
2. If needed, initiate the R2 access numbers, see 5.2.1 Initiating Access Numbers in the Number Series on page 10.
3. Initiate a remote extension, see 5.3.1 Initiating a Remote Extension on page 11.
4. Initiate Authorization data, see 5.4.3 Initiate a Individual Authorization Code on page 12.
5. If desired, name the remote extension, see 5.5.1 Initiating a Name Identity for the Remote Extension on page 13.
6. Initiate the personal number, see 5.6 Personal Number on page 13
7. Initiate the Recorded Voice Announcements (RVAs), see 5.7.1 Initiating a Recorded Voice Announcement on page 14.
8. Increase the callback time, see 5.9 Callback / Recall Time on page 14.

## 4.4

## REMOVING SIP REMOTE R2 EXTENSIONS

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

1. Remove Authorization data, see 5.4.4 Remove the Individual Authorization Code on page 13.
2. Remove the name for the remote extension, see 5.5.2 Removing a Name Identity for the Remote Extension on page 13
3. Remove the personal number, see 5.6 Personal Number on page 13
4. Remove the remote extension, see 5.3.2 Removing a Remote Extension on page 11

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

1. If used, remove the R2 access numbers, see 5.2.2 Removing a Number From a Series on page 10
2. Remove the Recorded Voice Announcements, see 5.7.2 Removing a Recorded Voice Announcement on page 14
3. Remove the trunks, see 5.1.2 Removing a Route or Trunk on page 9
4. Decrease the call back time, see 5.9 Callback / Recall Time on page 14



## 5 PROCEDURES

### 5.1 ROUTE AND TRUNK

#### 5.1.1 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.

#### 5.1.2 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.

## 5.2 NUMBER SERIES

### 5.2.1 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.

### 5.2.2 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.

## 5.3 REMOTE EXTENSION

### 5.3.1 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 route numbers and the remote number  
"--uri rou:xx&yy;remote-number=".

### 5.3.2 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.

## 5.4 NUMBER CONVERSION DATA

### 5.4.1 INITIATE 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.  
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*.
4. Enter the command *number\_conversion\_print* to verify the result.

### 5.4.2

#### 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.

### 5.4.3

#### INITIATE A INDIVIDUAL AUTHORIZATION CODE

Use the command *auth\_code -i*, specify the parameter *-dir*.

Use the command *auth\_code -p* to verify that the function has been initiated.

#### 5.4.4 REMOVE 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.

### 5.5 NAME IDENTITY FOR THE REMOTE EXTENSION

#### 5.5.1 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

#### 5.5.2 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.

### 5.6 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*.

## 5.7 RECORDED VOICE ANNOUNCEMENTS

### 5.7.1 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.  
See the parameter description for *RECORDED VOICE ANNOUNCEMENT, RA*.
3. Enter the command *RACEP* to verify the result.

### 5.7.2 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*.

## 5.8 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.

## 5.9 CALLBACK / 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)*.

## 5.10 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*.

## 5.11

### 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).

## 5.12

### STATUS PRINTOUT

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

## 6

## EXAMPLES

## 6.1

## 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 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>	Route number: 77 Redundant route number 88 Remote number: 719001234567
<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 the route**

Initiate the route for outgoing traffic.

```

sip_route -set -route 77 -protocol tcp -password hidden
-realm abcip.ip.oper.se -uristring0 sip:?@c-253.btrunk.se\
-uristring1 sip:+?@c-253.btrunk.se \
-fromuri0 "sip:!?@c-253.btrunk.se"

```

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 for incoming traffic.

```

sip_route -set -route 77 -password hidden -realm abcip.ip.oper.se \-accept ALL
-trusted yes

```

Register to a Broadworks SIP trunk.



```

sip_route -set -route 77 -register Broadworks -registerstring
"sip:4685630975001@c-253.btrunk.se" -authname 4685630975001

```

- **Initiate the route parameters**

Set the route categories.

```

ROCAI:ROU=77,SEL=7110000010000010,SIG=5111100000A0,
TRAF=03151515,TRM=5,SERV=3100030001,BCAP=111111;

```

```

RODAI:ROU=77,TYPE=TL66,VARC=00000001,VARI=00000000,
VARO=00000020;

```

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**

Initiate the route for outgoing traffic.

```

sip_route -set -route 88 -protocol tcp -password hidden\
-realm abcip.ip.oper.se -uristring0 sip:?@c-288.btrunk.se\
-uristring1 sip:+?@c-288.btrunk.se \
-fromuri0 "sip:!?@c-288.btrunk.se"

```

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

```

sip_route -set -route 88 -rexstring "666"

```

Initiate the route for incoming traffic.

```

sip_route -set -route 88 -password hidden -realm abcip.ip.oper.se\
-accept ALL -trusted yes

```

Register to a Broadworks SIP trunk.

```

sip_route -set -route 88 -register Broadworks -registerstring
"sip:4685630975001@c-288.btrunk.se" -authname
4685630975001

```

- **Initiate the redundancy route parameters**

Set the route categories.

```

ROCAI:ROU=88,SEL=7110000010000010,SIG=5111100000A0,TRAF=03151
515,TRM=5,SERV=3100030001,BCAP=111111;

```

```

RODAI:ROU=88,TYPE=TL66,VARC=00000001,VARI=00000000,VARO=0000
0020;

```

Finalize the initiation. Do not use registration for this route.

```

ROEQI:ROU=88,TRU=3-1&&3-10,INDDAT=000000000000;

```

Initiate the destination.

```

RODDI:ROU=88,DEST=88,SRT=3;

```

- **Initiate the extension categories**

```

extension_profile -i --csp 1 --ext-traf 0103151515 --ext-serv 2071511000300
--ext-cdiv 10810000 --ext-roc 000001 --ext-npres 0011

```

- **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 "rou:77;remote-number=719001234567"`
- **Initiate the number conversion data**  
`number_conversion_initiate -entry 00 719001234567`  
`-conversiontype 6, -truncate 14 , -pre 479`

**Note:** *remote-number* should be the same as a number received in the incoming call.

## 6.2

### 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`  
`"ROU:4;remote-number=01421163"`  
`ip_extension -i -d 68005 --terminal-identity "sip:68005@192.168. 32.13" --uri`  
`"ROU:4;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`

## 6.3

### 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`  
`"ROU:4;remote-number=01421163"`  
`ip_extension -i -d 68005 --terminal-identity "sip:68005@192.168. 32.13" --uri`  
`"dest!4@192.168.21.12;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
```

## 7

**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*.