Dialogic® Brooktrout® SR140 Fax Software with Aastra MX-ONE™

Installation and Configuration Integration Note

IMPORTANT NOTE

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1. Scope

This document is intended as a general guide for configuring a basic installation of the ***Aastra MX-ONE™*** for use with Dialogic® Brooktrout® SR140 Fax over IP (FoIP) software platform. The interoperability includes ***SIP*** call control and T.38/T.30 media.

This document is not intended to be comprehensive and thus should not and does not replace the manufacturer's detailed configuration documentation. Users of this document should already have a general knowledge of how to install and configure the ***Aastra MX-ONE™.***

The sample configuration shown and/or referred in the subsequent sections was used for lab validation testing by Dialogic. Therefore, it is quite possible that the sample configuration will not match an exact configuration or versions that would be present in a deployed environment. However, the sample configuration does provide a possible starting point to work with the equipment vendor for configuring your device. Please consult the appropriate manufacturer's documentation for details on setting up your specific end user configuration.

For ease of reference, the Dialogic® Brooktrout® SR140 Fax Software and Dialogic® Brooktrout® TR1034 Fax Boards will sometimes be denoted herein, respectively, as SR140 and TR1034. All references to the SDK herein refer to the Dialogic® Brooktrout® Fax Products SDK. The Aastra MX-ONE will be denoted herein as Aastra or MX-ONE, or some other form thereof. Some of the screen shots in this document reference Ericsson MX-ONE which was the previous product name for the Aastra MX-ONE™ system.

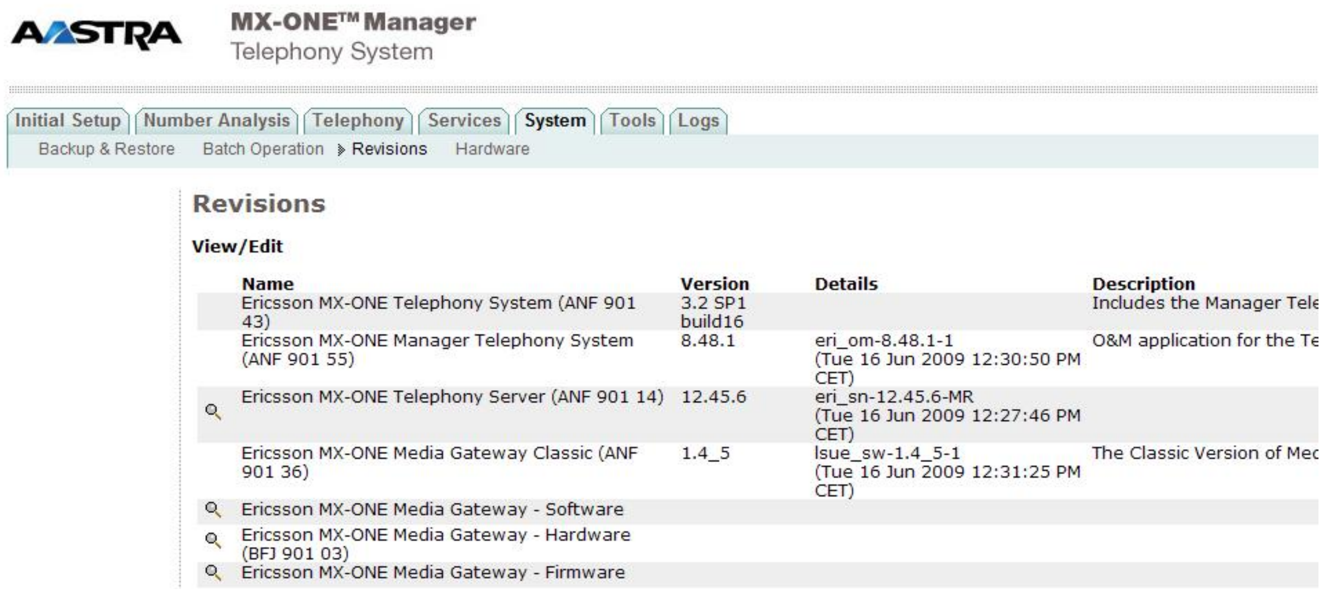
2. Configuration Details

The following systems were used for the sample configuration described in the document.

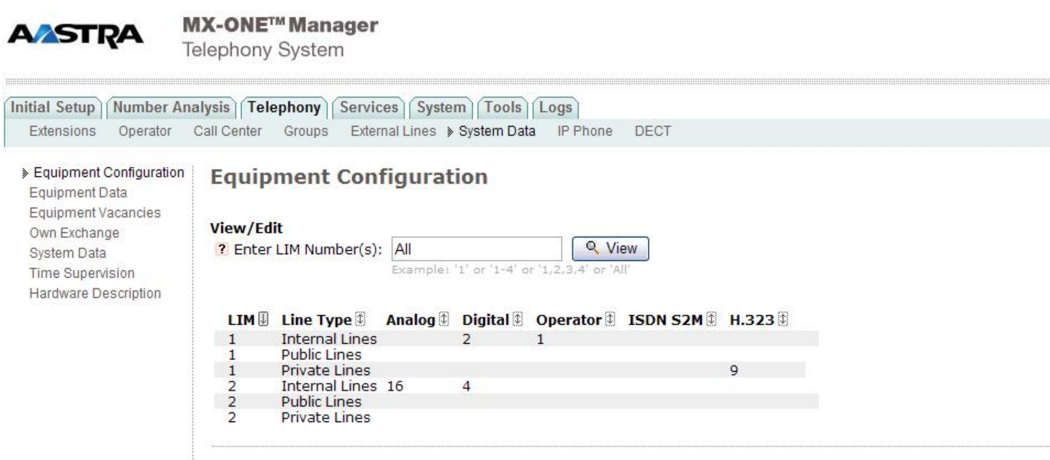
2.1 Gateway Aastra MX-ONE™ system

Vendor	***Aastra (previously Ericsson)***
Model(s)	***MX-ONE™***
Software Version(s)	***Aastra MX-ONE™ Telephony System (ANF 901 43) version 3.2 SP1 build 16*** ***Aastra MX-ONE™ Manager Telephony System (ANF 901 55) version 8.48.1*** ***Aastra MX-ONE™ Telephony Server (ANF 901 14) version 12.45.6*** ***Aastra MX-ONE™ Media Gateway Classic (ANF 901 36) version 1.4_5***
Hardware Version	***IPLU (line unit) board with minimum revision R6A in order to support T.38***
PSTN Device	***Dialogic® Brooktrout® TR1034 Analog LoopStart and*** ***Analog Nashuatec DSm622 multifunctional device***
Protocol to PSTN Device	***Internal Analog line on the same PBX***
IP Device	***Dialogic® Brooktrout® SR140*** ***Aastra MX-ONE™ call server***
Configuration	***Nothing to configure for T.38***
Additional Notes	***To enable T.38 on the Aastra MX-ONE, T.38 license is required.*** ***The Aastra MX-ONE has a call server that manages the call control and a DSP board that handles the T.38 Fax media.***

Shown below is a screenshot of the Aastra MX-ONE system configuration that was tested.



The following is the equipment configuration from the MX-ONE:



2.2 Dialogic® Brooktrout® SR140 Fax Software

Vendor	***Dialogic***
Model	***Dialogic® Brooktrout® SR140 Fax Software***
Software Version	***SDK 6.0.3 – used to test basic call functionality*** ***SDK 6.1.1 – used for the interop test suite***
Protocol to Gateway or Call Manager	***SIP***
callctrl.cfg file	***Default callctrl.cfg file included in SDK 6.1.1***

2.3 Dialogic® Brooktrout® TR1034 Fax Board

Vendor	***Dialogic***
PSTN Device	***Dialogic® Brooktrout® TR1034 Fax Board***
Software Version	***SDK 6.1.1***
Protocol to PSTN Device	***Analog Loop Start***
callctrl.cfg file	***All defaults***

2.4 Network System Configuration

The diagram below details the sample configuration used in connection with this document.

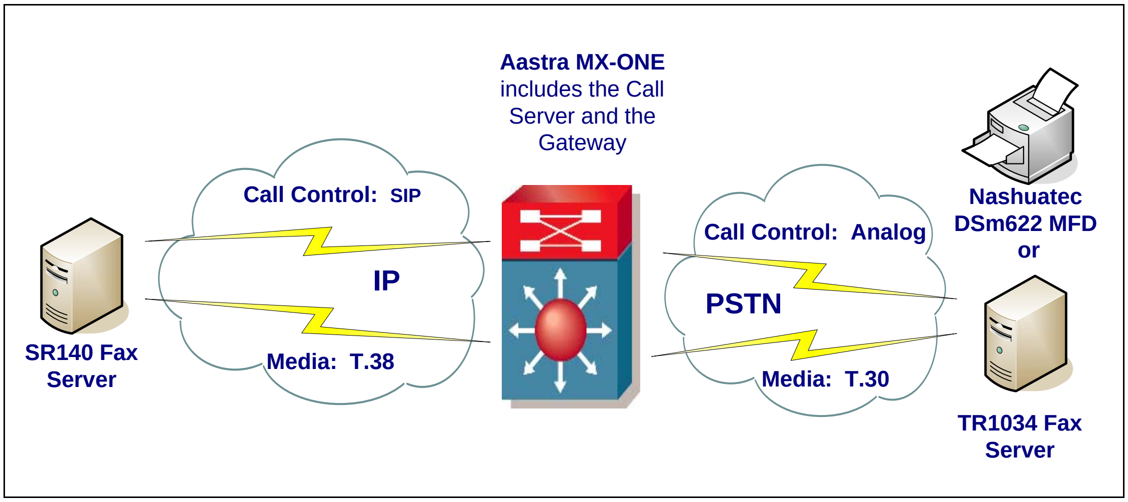


Diagram Notes:

- SR140 Fax Server = Fax Server including Dialogic® Brooktrout® SR140 Fax Software and third party fax application
- TR1034 Fax Server = Fax Server including Dialogic® Brooktrout® TR1034 Fax Board and third party fax application

3. Prerequisites

MX-ONE:

- IPLU (line unit) board with minimum revision R6A in order to support T.38
- A T.38 license is required to enable T.38 fax

SR140:

- SDK 6.0.X starting with SDK 6.0.3
- SDK 6.1.X starting with SDK 6.1.1

4. Summary of Limitations/Features

MX-ONE:

- T.38 is not configurable on the MX-ONE.
- The MX-ONE™ does Error Correction Mode (ECM) and V.17 14400 baud.
- The MX-ONE does not support V.34.

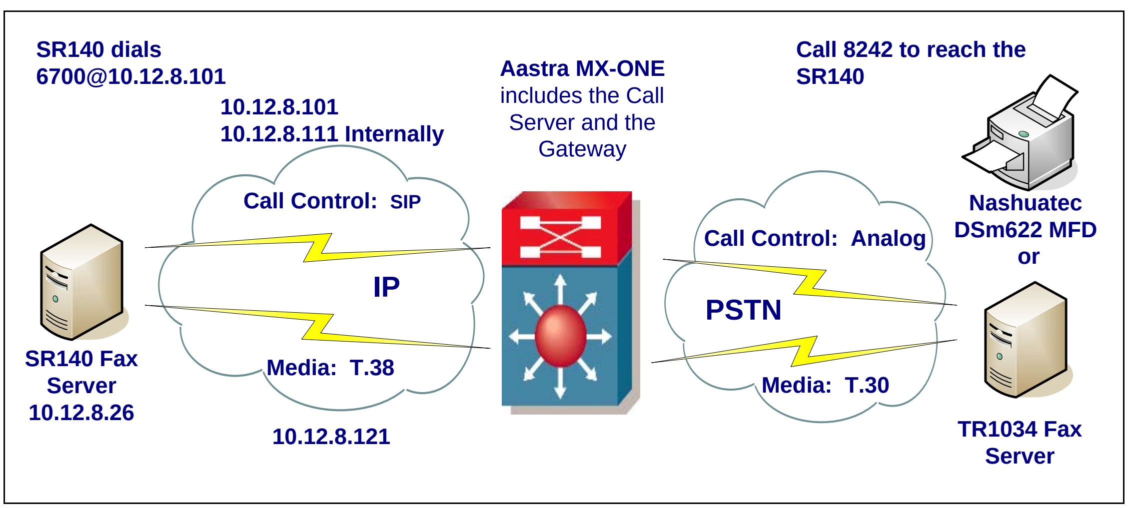
5. Deployment Details

5.1 Network Addresses

Device #	Device Make, Model, and Description	Device IP Address
1	SR140	10.12.8.26
2	MX-ONE	10.12.8.101

There were two Call Servers in the MX-ONE™ for failover. The external IP address, 10.12.8.101, was used to connect to the SR140. There were three gateways in the MX-ONE. The gateway with IP address, 10.12.8.121, was used in the interop tests.

5.2 Dialing Plan Overview



6. Dialogic® Brooktrout® SR140 Fax Software Setup Notes

The Installation and Configuration Guides for SDK 5.2.x, SDK 6.0.x and SDK 6.1.x are available from the site:

http://www.dialogic.com/manuals/brooktrout/default.htm

For the sample test configuration, the SR140 was configured using the default values from SDK 6.1.1 and is shown below for reference.

```
api_trace=none
host_module_trace=none
internal_trace=none
ip_stack_trace=none
l3l4_trace=none
l4l3_trace=none
max_trace_files=1
max_trace_file_size=10
trace_file=
[host_module.1]
module_library=brktsip.dll
enabled=true
[host_module.1/t38parameters]
t38_fax_rate_management=transferredTCF
fax_transport_protocol=t38_only
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
t38_fax_version=0
media_renegotiate_delay_inbound=1000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_type_of_service=0
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2
[host_module.1/rtp]
rtp_frame_duration=20
rtp_jitter_buffer_depth=100
rtp_codec=pcmu pcma
rtp_silence_control=inband
rtp_type_of_service=0
rtp_voice_frame_replacement=0
[host_module.1/parameters]
sip_max_sessions=256
sip_default_gateway=0.0.0.0:0
sip_proxy_server1=
sip_proxy_server2=
sip_proxy_server3=
sip_proxy_server4=
sip_registration_server1=
sip_registration_server1_aor=
sip_registration_server1_username=
sip_registration_server1_password=
sip_registration_server1_expires=3600
sip_registration_server2=
sip_registration_server2_aor=
sip_registration_server2_username=
```

```
sip_registration_server2_password=
sip_registration_server2_expires=3600
sip_registration_server3=
sip_registration_server3_aor=
sip_registration_server3_username=
sip_registration_server3_password=
sip_registration_server3_expires=3600
sip_registration_server4=
sip_registration_server4_aor=
sip_registration_server4_username=
sip_registration_server4_password=
sip_registration_server4_expires=3600
sip_registration_interval=60
sip_Max-Forwards=70
sip_From=Anonymous <sip:no_from_info@anonymous.invalid>
sip_Contact=0.0.0.0:0
sip_username=-
sip_session_name=no_session_name
sip_session_description=
sip_description_URI=
sip_email=
sip_phone=
sip_Route=
sip_session_timer_session_expires=0
sip_session_timer_minse=-1
sip_session_timer_refresh_method=0
sip_ip_interface=
sip_ip_interface_port=5060
sip_redirect_as_calling_party=0
sip_redirect_as_called_party=0
[module.41]
model=SR140
virtual=1
exists=1
vb_firm=C:\interop kit SDK611 v1.2\fdtool-6.1.1\bin\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={933ECC8B-7B1C-49D1-A036-33B1FFF17F9A}:0
f  media_port_min=56000
media_port_max=57000
[module.41/host_cc.1]
host_module=1
number_of_channels=60
```

No sip_default_gateway was filled in since the IP address of the gateway was specified in the dial string in the application. The following dial string was used for the outbound calls: 6700@10.12.8.101. However, when the application does not allow specifying the gateway's IP address, make sure to fill in the IP address in the sip_default_gateway field. In our test scenario, this would be: sip_default_gateway=10.12.8.101:5060

7. Dialogic® Brooktrout® TR1034 Fax Board Setup Notes

In the test configuration, the SR140 sent faxes to a MFP device and received faxes from the TR1034 Analog Loopstart board. For the sample test configuration, the following callctrl.cfg was used, however, note the default callctrl.cfg included in SDK 6.1.1 works fine as well.

```
l3l4_trace=none
l4l3_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=
max_trace_files=1
max_trace_file_size=10
[module.2]
model=TR1034+P8V8F-8L
exists=1
cc_type=0
channels=8
set_api=bfv
pcm_law=alaw
static_ring_detect_enable=true
[module.2/port.1]
port_config=analog
missing_wait=100
flash_hook_duration=50
input_gain=0
output_gain=0
transfer_variant=hookflash
protocol_file="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\analog_loopstart_us.lec"
country="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\us600.qslac"
did_offset=0
caller_id=disabled
num_rings=1
loop_reversal_for_connect=disabled
loop_reversal_for_disconnect=disabled
[module.2/port.2]
port_config=analog
missing_wait=100
flash_hook_duration=50
input_gain=0
output_gain=0
transfer_variant=hookflash
protocol_file="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\analog_loopstart_us.lec"
country="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\us600.qslac"
did_offset=0
caller_id=disabled
num_rings=1
loop_reversal_for_connect=disabled
loop_reversal_for_disconnect=disabled
[module.2/port.3]
port_config=analog
missing_wait=100
flash_hook_duration=50
input_gain=0
output_gain=0
transfer_variant=hookflash
```

```
protocol_file="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\analog_loopstart_us.lec"
country="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\us600.qslac"
did_offset=0
caller_id=disabled
num_rings=1
loop_reversal_for_connect=disabled
loop_reversal_for_disconnect=disabled
[module.2/port.4]
port_config=analog
missing_wait=100
flash_hook_duration=50
input_gain=0
output_gain=0
transfer_variant=hookflash
protocol_file="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\analog_loopstart_us.lec"
country="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\us600.qslac"
did_offset=0
caller_id=disabled
num_rings=1
loop_reversal_for_connect=disabled
loop_reversal_for_disconnect=disabled
[module.2/port.5]
port_config=analog
missing_wait=100
flash_hook_duration=50
input_gain=0
output_gain=0
transfer_variant=hookflash
protocol_file="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\analog_loopstart_us.lec"
country="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\us600.qslac"
did_offset=0
caller_id=disabled
num_rings=1
loop_reversal_for_connect=disabled
loop_reversal_for_disconnect=disabled
[module.2/port.6]
port_config=analog
missing_wait=100
flash_hook_duration=50
input_gain=0
output_gain=0
transfer_variant=hookflash
protocol_file="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\analog_loopstart_us.lec"
country="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\us600.qslac"
did_offset=0
caller_id=disabled
num_rings=1
loop_reversal_for_connect=disabled
loop_reversal_for_disconnect=disabled
[module.2/port.7]
port_config=analog
missing_wait=100
flash_hook_duration=50
input_gain=0
output_gain=0
transfer_variant=hookflash
protocol_file="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\analog_loopstart_us.lec"
country="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\us600.qslac"
did_offset=0
caller_id=disabled
```

```
num_rings=1
loop_reversal_for_connect=disabled
loop_reversal_for_disconnect=disabled
[module.2/port.8]
port_config=analog
missing_wait=100
flash_hook_duration=50
input_gain=0
output_gain=0
transfer_variant=hookflash
protocol_file="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\analog_loopstart_us.lec"
country="C:\interop kit SDK611 v1.2\fdtool-6.1.1\config\us600.qslac"
did_offset=0
caller_id=disabled
num_rings=1
loop_reversal_for_connect=disabled
loop_reversal_for_disconnect=disabled
# here followed the configuration parameters for the SR140 which was in the same server
```

All default values from the btcall.cfg configuration file were used, except for the country_code (European Community 0190).

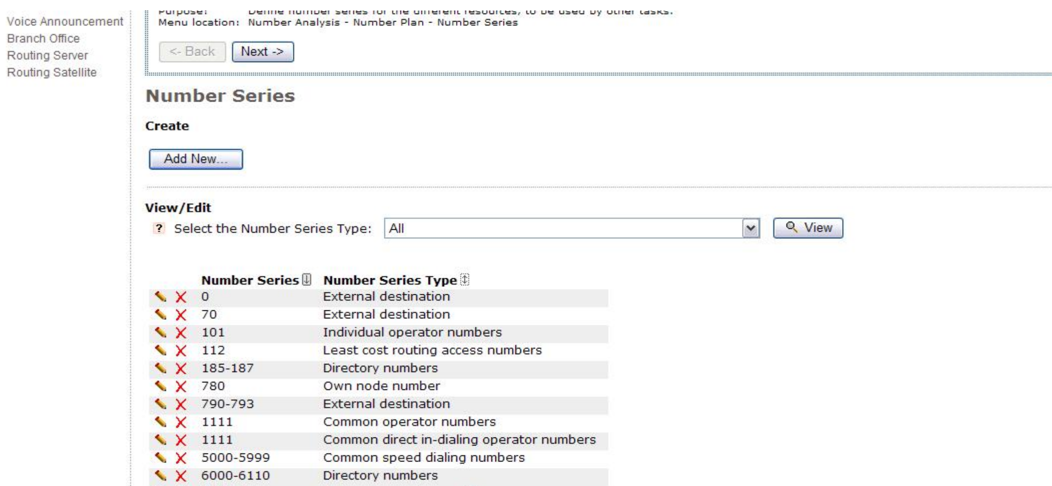
8. Aastra MX-ONE Gateway Setup Notes

To configure the Aastra MX-ONE, you can go to the respective menus or follow the MX-ONE Walkthroughs. The full MX-ONE setup walkthrough includes 28 steps. For the sample configuration, a SIP trunk was added to the SR140 fax server within the existing network by stepping through the walkthrough of the Route and the walkthrough of the Routing Server.

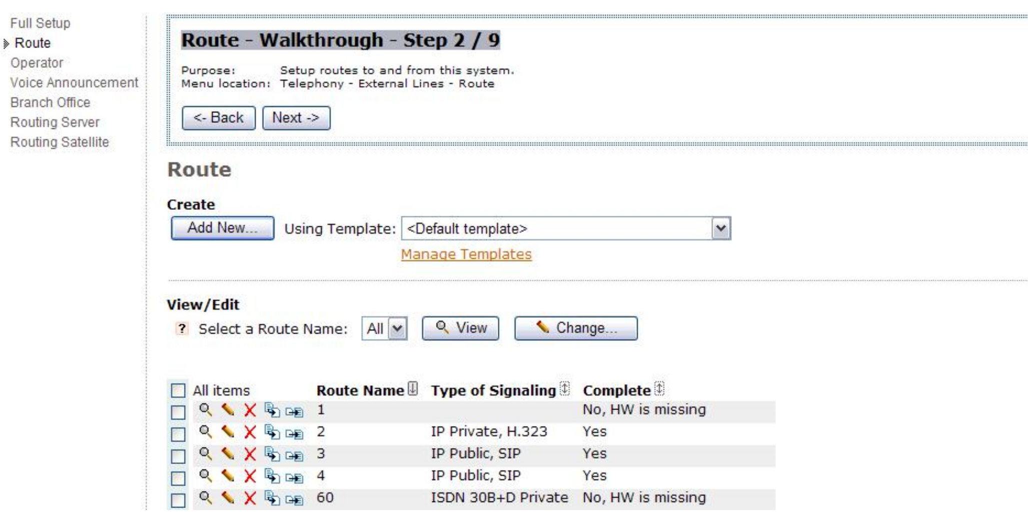
8.1 Walkthrough for the Route

The following screenshots capture the steps 1 – 4 used to configure the MX-ONE Route for the sample configuration.

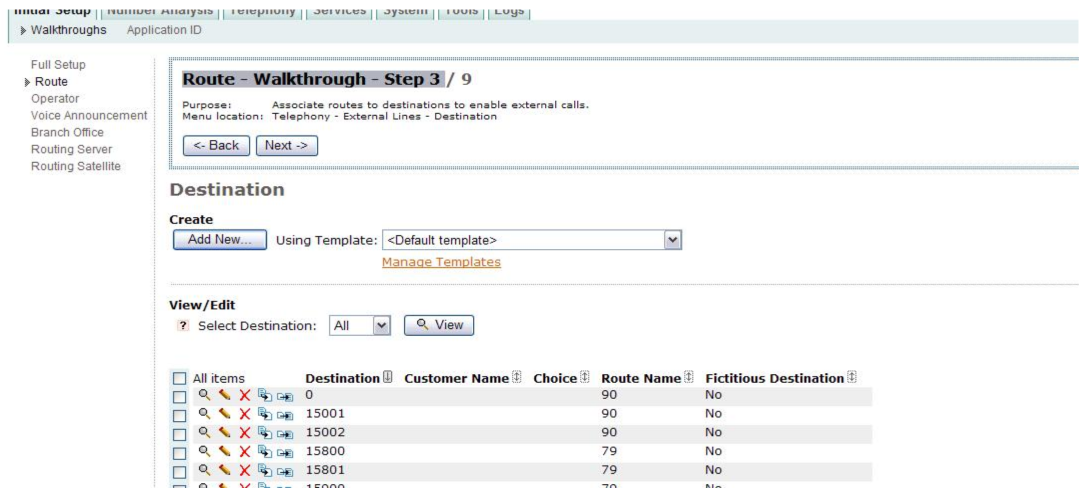
Step 1:



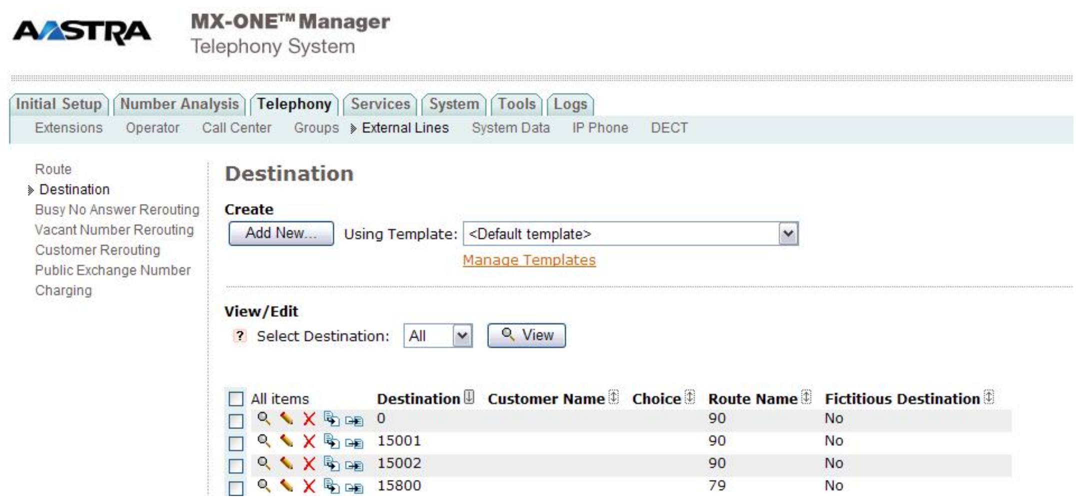
Step 2:



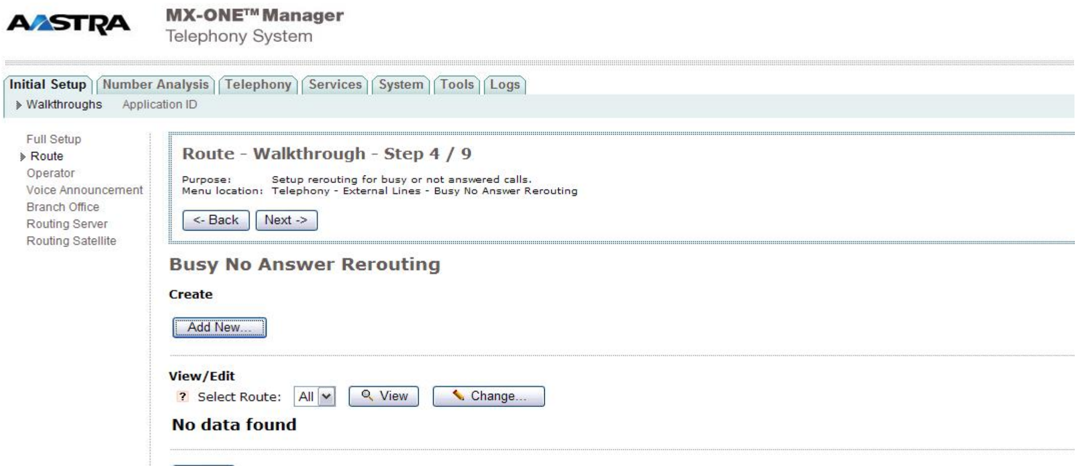
Step 3:



As the Menu location indicates, you can also find this same page back in the menu as follows:



Step 4:

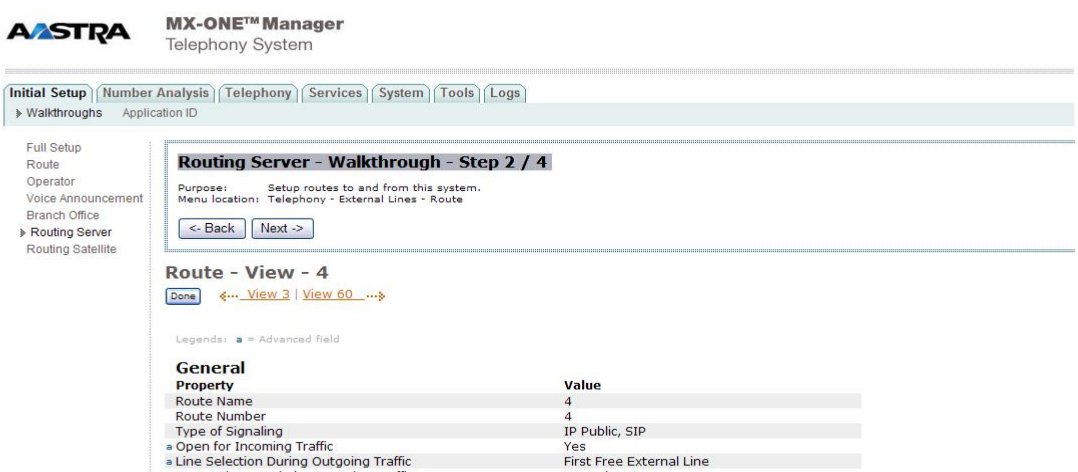


Steps 5 – 9 were skipped as they were not required for this sample configuration. After completing step 4 of the Route Walkthrough, proceed to the Walkthrough for the Routing Server.

8.2 Walkthrough for the Routing Server

The following screenshots capture steps 2 and 3 of the Routing Server Walkthrough used to set up the SIP trunk to the SR140 for this sample configuration.

Step 2:



The lower section of the Step 2 screenshot is shown below:

Route Characteristics Outgoing Traffic	Normal route
Allow Number Conversion	Yes
Dial Tone Characteristics after External Line Seizure	No monitoring path established
User of Digit Transmission for Transit Exchange	No
Ringing Tone Transmission for Outgoing Traffic	Ringing tone is generated in own exchange

Services

Property	Value
Rerouting on Congestion	Yes
Rerouting on Busy	No
Rerouting on no Answer	Yes
Allow Initiation of Call Waiting Tone Transmission	No
Allow Reception of Call Waiting Tone and Intrusion	Yes
Call Discrimination Group Night for Incoming External Lines	Fully Open
Call Discrimination Group Day for Incoming External Lines	Fully Open
Traffic Connection Class	15
Allow Alternative Route Selection	Yes
Restricted Presentation of Calling / Connected Number	Controlled by the extension
Abbreviated Dialing Traffic Class	3

Hardware

LIM-Trunk Index
1-1

IP Public, SIP

Property	Value
Protocol to Use When Calling	udp
Remote Port	5060
Invite URIString for Unknown Public Number	sip:8242@10.12.8.26
Type of Accepted Calls	All
Trusted Privacy Domain	No
Priority for Incomming Calls	255

Step 3:

Full Setup
Route
Operator
Voice Announcement
Branch Office
Routing Server
Routing Satellite

Routing Server - Walkthrough - Step 3 / 4

Purpose: Associate routes to destinations to enable external calls.
Menu location: Telephony - External Lines - Destination

<- Back | Next ->

Destination - View - 8242

Done ← View 8241 | View 9110 →

Property	Value
Destination	8242
Route Name	4
Start Position For Digit Transmission	1
Type of Called Number	Unknown Public
Type Of Calling Public Number	Unknown Public
Type Of Calling Private Number	Unknown Private
Truncated Digits in Dialed Number	0
Type of Signal Seizure	Terminating seizure
B-Answer Signal Available	Allowed
Allow to send Traveling Class Mark	Not Allowed
Maximum Number of Transit Exchanges	25
PNR Number Translation Information	No Translation
Supplementary Services Using User to User Interface	Not Allowed
Use Least Cost Routing for All Calls	No
Allow Sending of Expensive Route Warning Tone	Allowed
Type of Protocol to use for Supplementary Service Call Offer	User to user Interface(UUI)
Type of Protocol for Call Back/Call Completion	User to user Interface(UUI)
Show Original A Number	No

The screenshot below captures the full configuration for the SIP trunk to Destination 8242, the SR140 fax server, used in the sample configuration:

Print Close

AASTRA MX-ONE™ Manager
Telephony System

Logged in as: Ericsson

Date:Jul 9, 2009

Destination - 8242

Property	Value
Destination	8242
Route Name	4
Start Position For Digit Transmission	1
Type of Called Number	Unknown Public
Type Of Calling Public Number	Unknown Public
Type Of Calling Private Number	Unknown Private
Truncated Digits in Dialed Number	0
Type of Signal Seizure	Terminating seizure
B-Answer Signal Available	Allowed
Allow to send Traveling Class Mark	Not Allowed
Maximum Number of Transit Exchanges	25
PNR Number Translation Information	No Translation
Supplementary Services Using User to User Interface	Not Allowed
Use Least Cost Routing for All Calls	No
Allow Sending of Expensive Route Warning Tone	Allowed
Type of Protocol to use for Supplementary Service Call Offer	User to user Interface (UUI)
Type of Protocol for Call Back/Call Completion	User to user Interface (UUI)
Show Original A-Number	No
Use Original A-Numbers Type Of Number	No
Enable Enhanced Sent A-Number Conversion	Not Allowed
Use as Emergency Destination	No
Use ETSI Diversion supplementary service	No

9. Frequently Asked Questions

- ”*I’m configured as near as possible to this the sample configuration described in this document, but calls are still not successful; what is my next step?”*
  - ➔ Provide this document to your gateway support.
  - ➔ Ensure T.38 is enabled on the gateway.
  - ➔ Confirm that basic network access is possible by pinging the gateway.

- ”*How do I obtain Wireshark traces?*”
  - ➔ The traces can be viewed using the Wireshark network analyzer program, which can be freely downloaded from *http://www.wireshark.org*.
  - ➔ To view the call flow in Wireshark, open the desired network trace file and select “Statistics->VoIP Calls” from the drop down menu. Then highlight the call and click on the “Graph” button.