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Ingate SIParator-Firewall verification test

Contents

1 Document revision information

Version	Revision information	Resp.	Date
Pa1	First draft	ROKE	2007-01-18
Pa2	Some changes, mostly cosmetic, no content change. Figure 1 and 2 need to be replaced	CSP	2007-01-19
Pa3	Figure 1 and 2 are replaced	CSP	2007-01-19

2 Introduction

This document describes the verification test and results of two products from Ingate Systems, the Ingate SIParator and Ingate Firewall. The Ingate firewall makes it possible to access public SIP trunks outside the firewall even when the respective ports are closed. The firewall will be opened only when verified requests are received. The SIParator will be used when a customer already has a firewall installed that does not support this functionality. The SIParator will communicate with that firewall and configure it during SIP trunk access the same way as using an Ingate firewall.

2.1 Purpose

This verification test describes a number of tests to proof that the Ericsson MX-ONE™ can use a SIP-Trunk via the Ingate SIParator or an Ingate Firewall. Passing all tests is mandatory for the products to be recommended for type of deployments.

2.2 Overview

The test covers end-to-end verification of voice functionality between MX-ONE™ users located on the local network and users on the standard PSTN network, a SIP-Trunk is used for communication between PSTN and MX-ONE.

The full test will be done using the Ingate SIParator and a limited number of tests will be tested on the Ingate Firewall to verify the same functionality as with the Ingate SIParator. As the Firewall and SIParator use the same code it is only required to run fewer tests for verification and it can then be assumed that the same functionality is offered by the Firewall and SIParator.

2.3 References

Refer to http://www.ingate.com for more information about the verified products.

3 Requirements

The following hardware and software version(s) have been used for the test.

3.1 Hardware requirements

- One Ingate SIParator 20 running software version 4.5.0.
- One Ingate Firewall 1400 running software version 4.5.0.
- One SIP Trunking Service Provider account with two DID numbers and permission to make outbound calls to the PSTN phones.

- Two PSTN phones with DID's reachable from the SIP-Trunks and with permission to place calls to the SIP-trunk DID.
- Access to Internet with one public IP address for the SIParator
- MX-ONE™ V3 LSV23+. This version is not released to the market but has been approved by product management to be run for this test.
- 1 LIM system consisting of 1 Media Gateway
- 2 IP phones DBC42502, firmware P4A4
- 1 analogue fax

4 Configuration of used components

4.1 Test setup

The following test scenarios have been used during the test.

The first one covers the Ingate SIParator where a different brand of firewall has been used, A Juniper Netscreen NS50

The second one included an Ingate firewall the FW 1400.

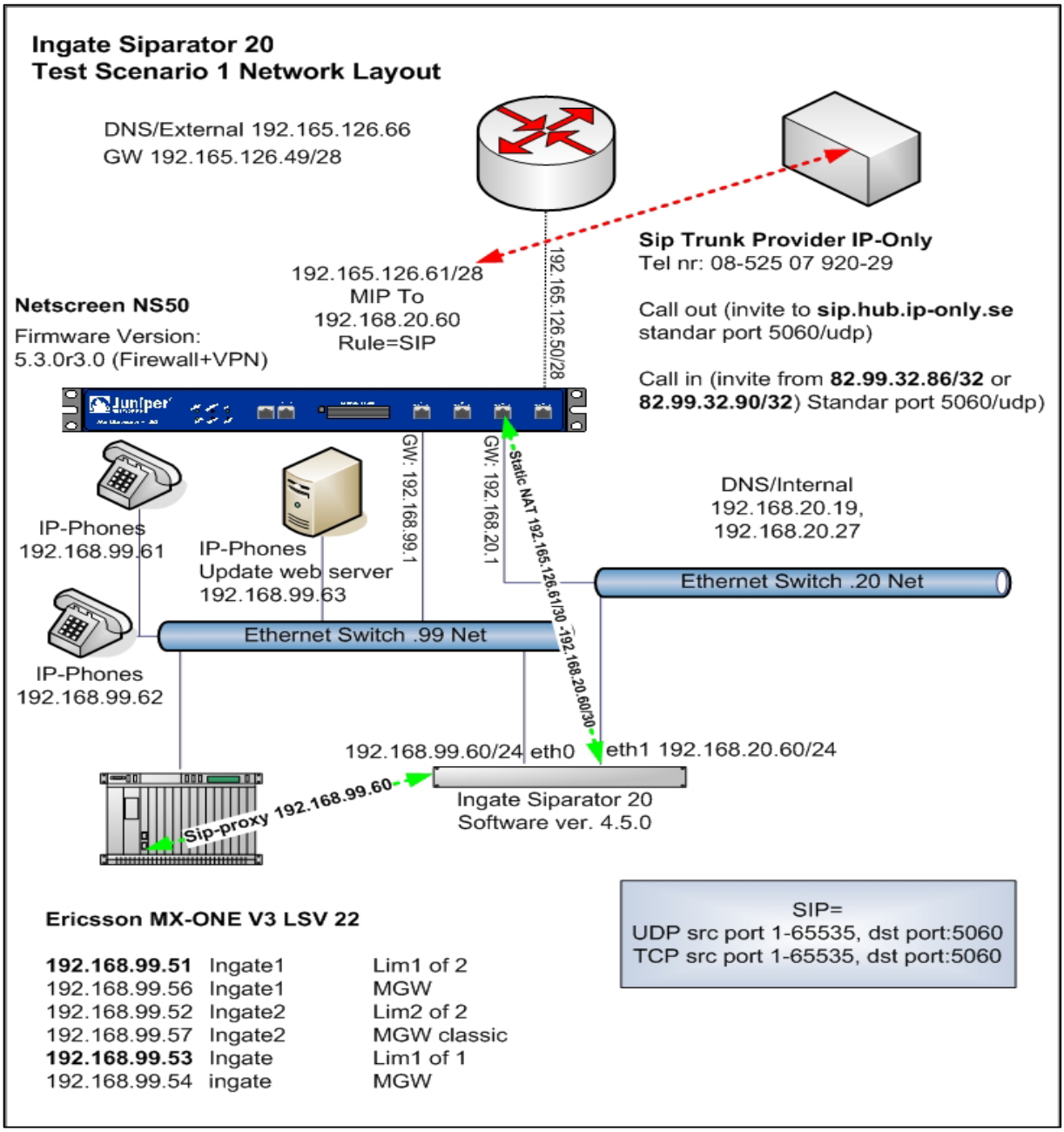

Figure 1 Test scenario 1: SIParator 20

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EBC/Z/DL Rob Kemmeren		EBC-07:000125 Uen		
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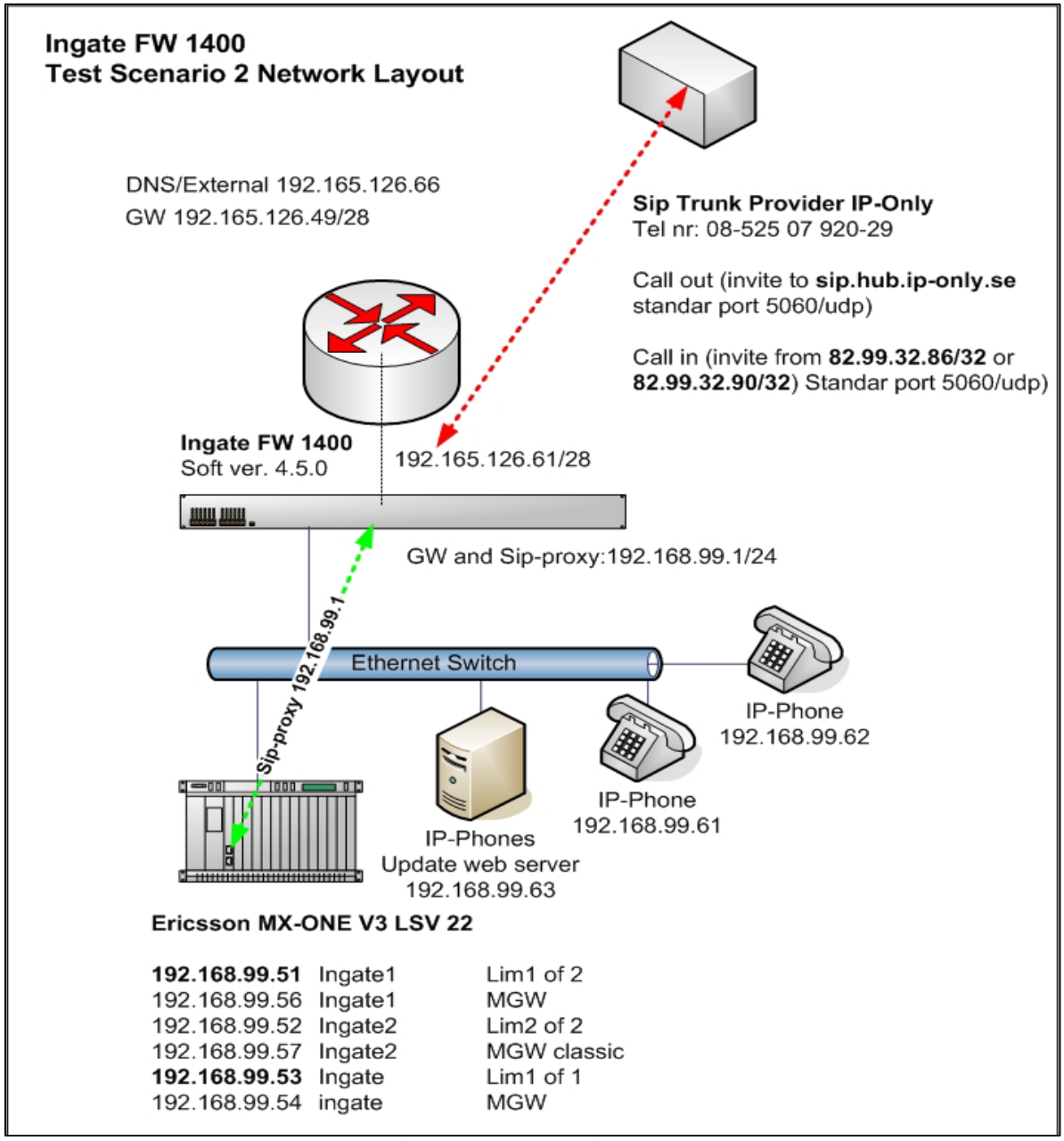

Figure 2: Test scenario 2, FW 1400

4.1.1 Ingate SIParator configuration

Basic setup

Basic setup and network configuration:
Assign IP addresses to two interfaces
- Assign a default gateway
- Assign a DNS server address
- Define IP subnet allowed to configure and interfaces to use for configuration

NETWORK – NETWORK AND COMPUTERS
- Add a network for the Service provider (IP-Only)
- Add a network for the internal LAN (inside)

SIP configuration

Make the SIP configuration in the according to the following. The same configuration is used for both the Firewall and SIParator unless other are specified.

BASIC CONFIGURATION – SIParator TYPE (SIPARATOR ONLY)
- DMZ/LAN

SIP SERVICE – BASIC
- SIP module: On
- Log class for SIP debug messages: Local

SIP SERVICE – INTEROPERABILITY
- Public IP address for NATed SIParator: 192.165.126.61 (SIPARATOR ONLY)

SIP TRAFFIC – Filtering
- Default policy for SIP requests: Process all
- Add a row to the Content Types table:
    - Content type: */*
    - Allow: On

SIP TRAFFIC – DIAL PLAN
- Use Dial Plan: On
- Matching From line 1
    - Name: IP-Only (just a friendly name)
    - Username: *
    - Domain: *
    - Reg. Exp: (empty)
    - Transport: UDP
    - Network: IP-only
- Matching From line 2
    - Name: MX-ONE (just a friendly name)
    - Username: *
    - Domain: *
    - Reg. Exp: (empty)
    - Transport: UDP
    - Network: internal
- Matching Request-URI 1
    - Name: Any_number (just a friendly name)
    - Prefix: (empty)
    - Head: (empty)
    - Tail: “0-9, +, -, #, *”
    - Min. Tail: (empty)
    - Domain: *
    - Reg Exp: (empty)
- Matching Request-URI 2
    - Name: Inbound (just a friendly name)
    - Prefix: (empty)
    - Head: 085250792
    - Tail: “0-9, +, -, #, *”
    - Min. Tail: (empty)
    - Domain: 192.165.126.61
    - Reg Exp: (empty)

- Forward To 1
  - Name: IP_Only (just a friendly name)
  - Subno.: 1
  - Use This Account: -
  - Replacement URI: sip.hub.ip-only.se
  - Port: 5060
  - Transport: UDP
  - Reg exp: (empty)
- Forward To 2
  - Name: MX-ONE (just a friendly name)
  - Subno.: 1
  - Use This Account: -
  - Replacement URI: 192.168.99.51
  - Port: 5060
  - Transport: UDP
  - Reg exp: (empty)
- Dial Plan 1
  - No: 1
  - From Header. IP-Only
  - Request-URI: Inbound
  - Action: Forward
  - Forward To: MX-ONE
  - Forward: (empty)
  - ENUM: (empty)
  - ENUM root: -
  - Comment: (optional)
- Dial Plan 2
  - No: 2
  - From Header. MX-ONE
  - Request-URI: Any_number
  - Action: Forward
  - Forward To: IP-Only
  - Forward: (empty)
  - ENUM: (empty)
  - ENUM root: -
  - Comment: (optional)

SIP TRAFFIC – ROUTING

- Always handle REFER locally.

4.1.2 MX-ONE Sip-Trunk configuration

```
sip_route -set -route 75 -uristring sip:?@192.168.99.1 -proxyip 192.168.99.1 -
proxyport 5060

sip_route -set -route 75 -accept REMOTE_IP -match 192.168.99.1

ROCAI:ROU=75,SEL=7110000000000010,SIG=0111111000A0,TRAF=03151515,TRM=4,SERV=311
0000001,BCAP=000100;

RODAI:ROU=75,TYPE=TL66,VARI=00000011,VARC=00000001,VARO=00000010;

RODDI:rou=75,dest=00,srt=3;

ROEQI:ROU=75,TRU=1-1;
```

5 Test cases

5.1 Test cases – Ingate SIParator

These test cases verify that any clients connected to the MX-ONE can make calls to and receive calls from PSTN using the SIP-Trunk.

5.1.1 Basic call

Basic normal calls between IP-PBX and PSTN
- Make a call from A to B
- Make sure you have a ringing signal in both ends.
- Look in the Ingate at SIP TRAFFIC – SIP STATUS and make sure there is a call in status “calling”
- Answer the call and verify media in both directions
- Look in the Ingate at SIP TRAFFIC – SIP STATUS and make sure there is one active call.
- A clears. Verify that the call is terminated on B and that the call disappears on the Ingate Status page.

	Incoming Call	Outgoing Call
A clears	OK	OK
B clears	OK	OK

5.1.2 Call Cancellation

Test termination for the call by the caller before answer.

- Make a call from A to B
- Make sure you have a ringing signal in both ends.
- A clears
- Verify that B stops ringing.

The call will be listed in the Ingate status page for 30s after call cancellation.

Incoming Call	Outgoing Call
OK	OK

5.1.3 Call no answer

Test behavior when no answer when calling to the PBX.

- Make a call from A to B
- Make sure you have a ringing signal in both ends

Incoming Call	Outgoing Call
OK	OK

5.1.4 Call to busy

- Make a call from A to B
- Make sure you have a ringing signal in both ends.

Incoming Call	Outgoing Call
OK	OK

5.1.5 DTMF tones

- DTMF signaling between PBX and PSTN

Outgoing Call to Voice System
OK

5.1.6 Fax

- Send/receive a fax

Incoming Fax	Outgoing Fax
OK	OK

5.1.7 Diverted calls

For this test we used IP extensions with a personal list initiated and direct diversion.

	Incoming call follow me	Incoming call div. no answer	Incoming call div. on busy
A clears	OK	OK	OK
B clears	OK	OK	OK

External follow me

A calls B who has ECF activated toward C

	Internal call	External call
A clears	OK	OK
C clears	OK	OK

5.1.8 Call Transfer before answer

Test the behavior in unattended call transfer scenarios.

NOTICE: The PBX phone should initiate the transfer. The PSTN phone will never initiate transfer

- Make a call from A to B
- Make sure you have a ringing signal in both ends
- Answer the call and verify media in both directions
- Make an unattended transfer from the PBX phone to C
- Make sure it rings on C and that the transferred party has the expected signal.
- Answer the call and verify media in both directions.
- Terminate the call
- Verify that the call is terminated from the Ingate SIP Status

A-party	B-party	Transfer done by:	C-party	result
Internal	Internal	A	External	OK
Internal	Internal	B	External	OK
External	Internal	B	External	OK
External	Internal	B	Internal	OK
Internal	External	A	Internal	OK
Internal	External	A	External	OK

5.1.9 Call transfer after answer

Test the behavior in attended call transfer scenarios.

NOTICE: The PBX phone should initiate the transfer. The PSTN phone will never initiate transfer.

- Make a call from A to B
- Make sure you have a ringing signal in both ends
- Answer the call and verify media in both directions
- Make an attended transfer from the PBX phone to C
- Make sure it rings on C and answer. Verify media in both directions.
- Verify expected signal for the party to be transferred

- Initiate the transfer and verify media in both directions after transfer.
- Terminate the call
- Verify that the call is terminated from the Ingate SIP Status page

A-party	B-party	Transfer done by:	C-party	result
Internal	Internal	A	External	OK
Internal	Internal	B	External	OK
External	Internal	B	External	OK
External	Internal	B	Internal	OK
Internal	External	A	Internal	OK
Internal	External	A	External	OK

5.1.10 Conference

Test 3-party conference scenarios.

NOTICE: The PBX phone should initiate the 3-party conference.

- Make a call from A to B
- Make sure you have a ringing signal in both ends
- Answer the call and verify media in both directions
- Initiate a conference call from the PBX party to C.
- Make sure it rings on C and that the waiting party have expected signal
- C answers and verify media in both directions
- Conference all three parties into a conference and verify media in all directions.
- Terminate the call
- Verify that the call is terminated from the Ingate SIP Status page

A-party	B-party	Transfer done by:	C-party	result
Internal	Internal	A	External	OK
Internal	Internal	B	External	OK
External	Internal	B	External	OK
External	Internal	B	Internal	OK
Internal	External	A	Internal	OK
Internal	External	A	External	OK

5.1.11 Emergency Call Logged off IP extension

A is logged of, but is able to dial the emergency number

Outgoing Call to Emergency nr
OK

5.2 Test cases – Ingate Firewall

Before starting these tests make sure the setup and configuration are changed according to the section Configuration of the Ingate Firewall.

5.2.1 Call Cancellation

Test termination for the call by the caller before answer.

- Make a call from A to B
- Make sure you have a ringing signal in both ends.
- A clears
- Verify that B stops ringing.

The call will be listed in the Ingate status page for 30s after call cancellation.

Incoming Call	Outgoing Call
OK	OK

5.2.2 Call Transfer before answer

Test the behavior in unattended call transfer scenarios.

NOTICE: The PBX phone should initiate the transfer. The PSTN phone will never initiate transfer

- Make a call from A to B
- Make sure you have a ringing signal in both ends
- Answer the call and verify media in both directions
- Make an unattended transfer from the PBX phone to C
- Make sure it rings on C and that the transferred party has the expected signal
- Answer the call and verify media in both directions.
- Terminate the call
- Verify that the call is terminated from the Ingate SIP Status

A-party	B-party	Transfer done by:	C-party	result
Internal	Internal	A	External	OK
Internal	Internal	B	External	OK
External	Internal	B	External	OK
External	Internal	B	Internal	OK
Internal	External	A	Internal	OK
Internal	External	A	External	OK

5.2.3 Call transfer after answer

Test the behavior in attended call transfer scenarios.

NOTICE: The PBX phone should initiate the transfer. The PSTN phone will never initiate transfer.

- Make a call from A to B
- Make sure you have a ringing signal in both ends
- Answer the call and verify media in both directions
- Make an attended transfer from the PBX phone to C
- Make sure it rings on C and answer. Verify media in both directions.
- Verify expected signal for the party to be transferred
- Initiate the transfer and verify media in both directions after transfer.
- Terminate the call
- Verify that the call is terminated from the Ingate SIP Status page

A-party	B-party	Transfer done by:	C-party	result
Internal	Internal	A	External	OK
Internal	Internal	B	External	OK
External	Internal	B	External	OK
External	Internal	B	Internal	OK
Internal	External	A	Internal	OK
Internal	External	A	External	OK

5.2.4 Conference

Test 3-party conference scenarios.

NOTICE: The PBX phone should initiate the 3-party conference.

- Make a call from A to B
- Make sure you have a ringing signal in both ends
- Answer the call and verify media in both directions
- Initiate a conference call from the PBX party to C.
- Make sure it rings on C and that the waiting party have expected signal
- C answers and verify media in both directions
- Conference all three parties into a conference and verify media in all directions.
- Terminate the call
- Verify that the call is terminated from the Ingate SIP Status page

A-party	B-party	Transfer done by:	C-party	result
Internal	Internal	A	External	OK
Internal	Internal	B	External	OK
External	Internal	B	External	OK
External	Internal	B	Internal	OK
Internal	External	A	Internal	OK
Internal	External	A	External	OK

5.2.5 Emergency Call Logged off IP extension

A is logged of, but is able to dial the emergency number

Outgoing Call to Emergency nr
OK

6 Conclusion

In all above test cases the Ingate products, Ingate SIParator and Ingate Firewall, performed according to expectations in every aspect.

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