



Interactive Intelligence

CIC 2017 R2 Patch6

Configuration Guide

Performed By



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1 Introduction

This configuration guide describes how to configure an Interactive Intelligence (ININ) CIC 2017 R2 P4 SERVER with an Oracle Acme Packet SBC running ECZ740p2 release on a AP3900. The deployment model covered in this application note is an ININ CIC 2017 R2 SERVER with an Oracle Acme Packet SBC 3900 connected to the SIP trunk.

The configuration of the ININ CIC 2017 R2 SERVER detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between the Oracle Acme Packet SBC 3900 and SIP trunk.

Testing was performed by tekVizion Labs, an independent testing and certification facility, in accordance to ININ Certification Test Plan. Key features verified are:

- New services install processes
- Call capabilities (various features, basic call operations, local calling, domestic long distance calling and international calling)
- FAX
- Out-of-band DTMF
- Conference Calls

The configuration described in this document details the critical commands to have enabled for interoperability to be successful.

This Application Note details the configuration used for connectivity to the tekVizion PSTN network. This document serves as guidance for the integration, but does not guarantee interoperability for every use case or release combinations.

2 Validation Environment

2.1 Network Topology

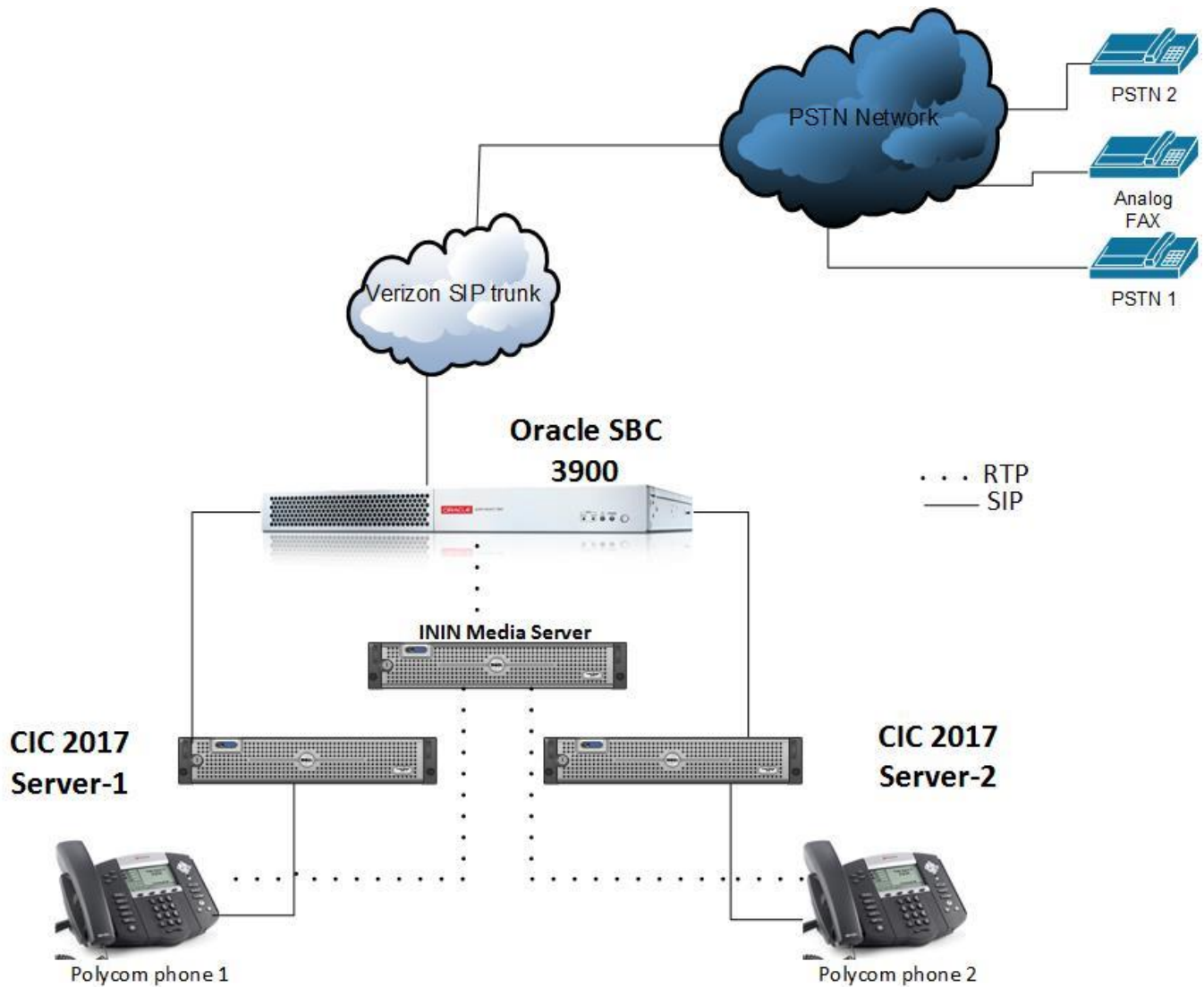


Figure 1: Network Topology

2.2 Hardware Components

- ININ CIC Media Server
- ININ CIC 2017 Servers
- Polycom IP 550 and IP 650 Phones
- Oracle SBC 3900

2.3 Software Requirements

- ININ CIC 2017 R2 Patch6
- Acme Packet 3900 ECZ7.4 Patch 2

2.4 Features

2.4.1 Features Supported

- Voice calls using G.711 codec
- RFC 3261 support
- Calling number presentation / restriction
- Call conferencing
- Call transfer (attended, unattended & blind transfer)
- Call hold and resume
- Call forwarding
- CIC Server Switchover

2.4.2 Features Not Supported

Summary of Test Results

2.5 Declaration

tekVizion has tested the ININ CIC 2017 R2 Patch6 with the Oracle “Acme Packet SBC” and has validated its use with the Service Provider.

2.5.1 Purpose

This document serves to provide other areas of ININ CIC 2017 R2 Patch6 Certification test review of the Oracle Acme Packet SBC 3900. Uses of this product outside that of public voice call termination/origination to the Oracle Acme Packet SBC 3900 host falls outside the scope of this validation.

2.6 Passed

All test cases that were successfully executed have been marked as ‘Pass’

2.7 Not Tested / Not Supported

- None identified

2.8 Known Limitations and Restrictions

- None identified

2.9 Test Plan

- **Device Under Test (DUT)** – Oracle Acme Packet SBC 3900
- **DUT - Customer Interface Type** – SBC
- **WAN Transport** – Direct Internet connection

#	TC#	Title	Results	
1	4906	Configure SIP Carrier to Route Calls to IC	Pass	
2	11403	Configure SIP Carrier Line	Pass	
3	11402	Configure & Provision Managed IP Phones	Pass	
4	11406	Configure Dial Plan for Outgoing Calls	Pass	
5	11411	Out-of-band DTMF	Pass	
6	11416	G.711 Through Carrier	Pass	
7	4898	Inbound call with ANI	Pass	
8	4899	Inbound Call without ANI	Pass	
9	4900	Outbound Call with ANI	Pass	Call works only if the ANI information is valid DID for ITSP
10	8700	Early Media	Pass	
11	5553	Transfer - Blind (Internal)	Pass	
12	4877	Transfer – Consult (Internal)	Pass	
13	4878	Transfer – Blind (External)	Pass	
14	4879	Transfer – Consult (External)	Pass	
#	TC#	Title	Results	
15	4884	Hold Support – Re-Invite to 0.0.0.0	Pass	
16	4885	Forward Incoming Call - from Carrier to Carrier	Pass	
17	4886	3 Party Conference	Pass	
18	4887	Fax - G.711 Pass-through	Pass	10 out of 10 inbound and outbound FAX calls are successful
19	14094	Always In Audio	Pass	
20	4908	Service Unavailable – 503 response from carrier	Pass	ITSP is sending announcement saying “The number is no longer is in service”.
21	4915	Power Failure	Pass	
22	14127	Switchover Support	Pass	
23	14128	WAN Failure	Pass	

24	14311	Latency SIP Outbound	Pass	Latency is measured as 10ms. Call was loop backed at DUT
25	14150	Outbound Call with RESTRICTED ANI	Pass	
26	4880	Putback (Release Link) Transfer (Blind)	Pass	Completed REFER based test cases using different ITSP as we had issues with Verizon SIP trunk
27	4881	Putback (Release Link) Transfer (Consult)	Pass	Completed REFER based test cases using different ITSP as we had issues with Verizon SIP trunk
28	4888	Fax - T.38 (Inbound)	Pass	10 out of 10 FAX calls are successful
29	4889	Fax - T.38 (Outbound)	Pass	10 out of 10 FAX calls are successful
30	4890	Dynamic Audio	Pass	
#	TC#	Title	Results	
32	6001	QoS tagging of RTP is preserved	Pass	
33	6002	QoS tagging of SIP Signaling is preserved	Pass	
Optional Test Cases –FS 1259				
34	6113	Multiple Ports	Pass	
35	6114	SIP to SIP call support	Pass	
36	6115	Support for TCP SIP Messaging	Pass	
37	6655	Support for sRTP	Pass	
38	6722	Support for TLS	Pass	
WAN Managed Phone Test Cases – FS 1422				
#	TC#	Title	Results	Notes
40	14179	Provision WAN Managed Phone	Pass	
41	14183	WAN Phone Intercom Calls	Pass	
42	14184	WAN Phone External Calls	Pass	
43	14185	WAN Phone Transfer (Blind)	Pass	
44	14186	WAN Phone Transfer (Consult)	Pass	
45	14187	WAN Phone Conference	Pass	
46	14188	WAN Phone Switchover	Pass	No issues with SBC functionality. Normal switchover scenarios were executed successfully. While trying to execute WAN phone based switchover scenarios, ININ CIC server is not processing the calls as expected. Support ticket opened with ININ Incident NO: 927664

3 CIC Server Trunk Configuration

3.1 Trunk (Line) Configuration

3.1.1 Set Server IP Address

1. Navigate **Start > Control Panel > Network and Internet > View Network Status and Tasks > Change Adapter Settings > Ethernet > Ethernet Status > Properties > Internet Protocol Version 4(TCP/IP4) > Properties**
2. **Use the Following IP Address:** Selected
3. Set **IP Address:** Enter the IP address assigned for the ININ CIC 2017 server
4. Set **Subnet Mask:** 255.255.0.0 is used for this example
5. Set **Default Gateway:** 10.64.1.1 is used for this example
6. Set **Preferred DNS Server:** 10.64.3.39 is used for this example
7. Set **Alternate DNS Server:** Enter the DNS server address for Enterprise network..
8. Click **OK**

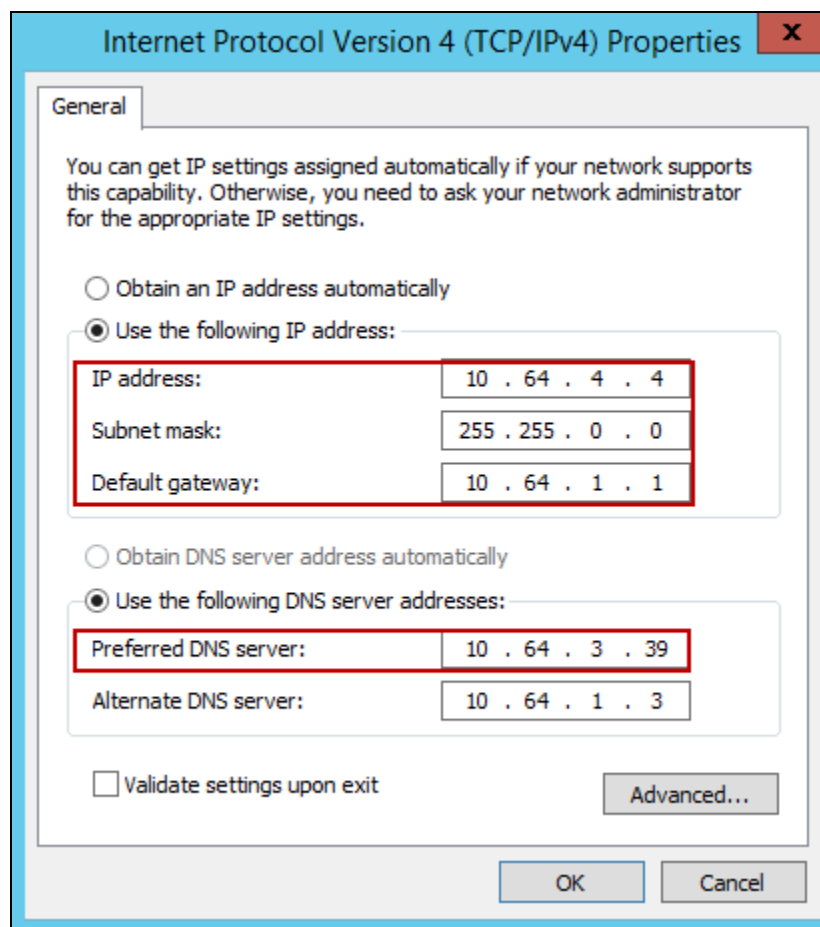


Figure 2: Server IP Address

3.1.2 SIP Line Configuration

1. In **Interaction Administrator**: Navigate to CICSERVER1(Name of IC Server) – 2017 > R2 Lines
2. Right-click on **Lines** and select **New**
3. Set **Line Name**: Stations-UDP is used for this example

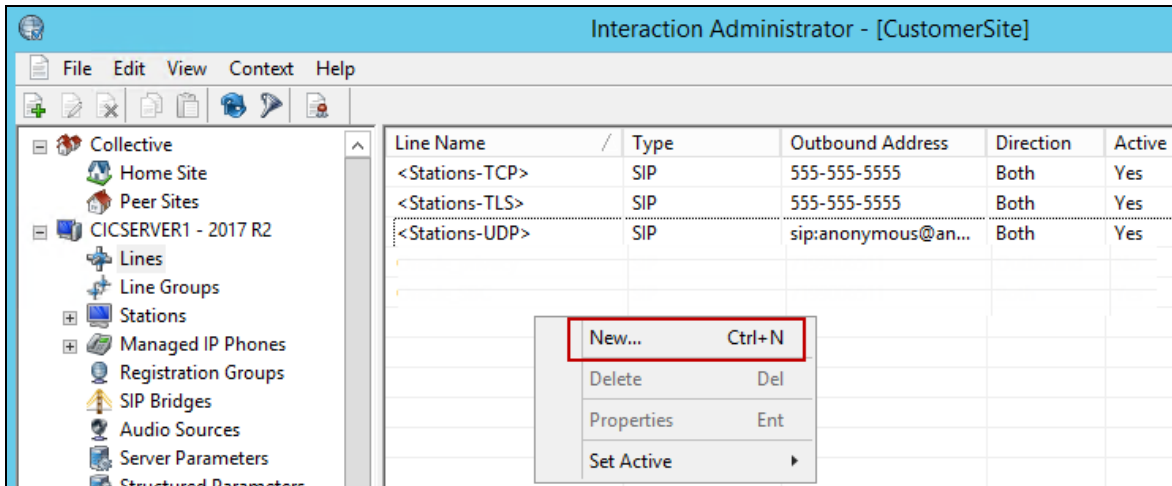


Figure 3: Line Configuration

3.1.2.1 Line Configuration

1. Select the **SIP Line Configuration** tab
2. In left navigation pane, select **Line**
3. Set **Line Usage**: *General Purpose* is selected from the drop down menu
4. Set **Domain Name**: Enter the FQDN of the server provided at the domain controller.
5. Set **Fax Protocol**: T30 then T38 were selected for this example
6. Check **Enable Fax Detection**. This supports both Fax Protocols T30 & T38. This can be changed based on the carrier requirement.
7. All the other values are set to default values

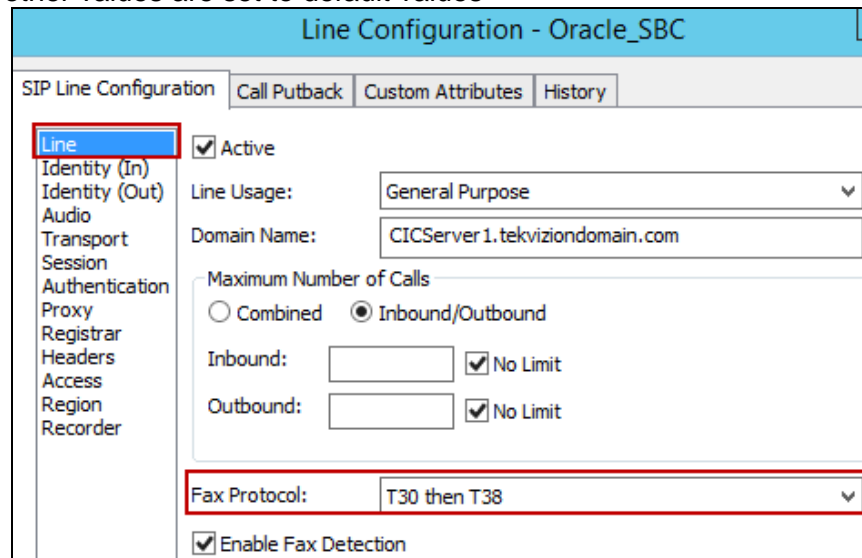


Figure 4: Line Configuration – Cont.

Line Configuration - Oracle_SBC

SIP Line Configuration | Call Putback | Custom Attributes | History

Line
Identity (In)
Identity (Out)
Audio
Transport
Session
Authentication
Proxy
Registrar
Headers
Access
Region
Recorder

Enable Fax Detection

Maximum Number of Faxes
 Combined Inbound/Outbound

Inbound: No Limit
 Outbound: No Limit

Auto Disconnect when Silence Detected in Voice Mail
 Silence Time (ms):

Call Analysis Type:

Figure 5: Line Configuration – Cont.

Line Configuration - Oracle_SBC

SIP Line Configuration | Call Putback | Custom Attributes | History

Line
Identity (In)
Identity (Out)
Audio
Transport
Session
Authentication
Proxy
Registrar
Headers
Access
Region
Recorder

Outbound: No Limit

Auto Disconnect when Silence Detected in Voice Mail
 Silence Time (ms):

Call Analysis Type:

Allow Deferred Answer
 Playback Early Media to Inbound Calls
 Enable SIP Prack/Update for EarlyMedia Support

Max Probation Time (s):

⏪ ⏩ Confirm auto-save OK Cancel Ap

Figure 6: Line Configuration – Cont.

3.1.2.2 Identity (In)

1. In left navigation pane, click **Identity (In)**
2. All the values are set to default values

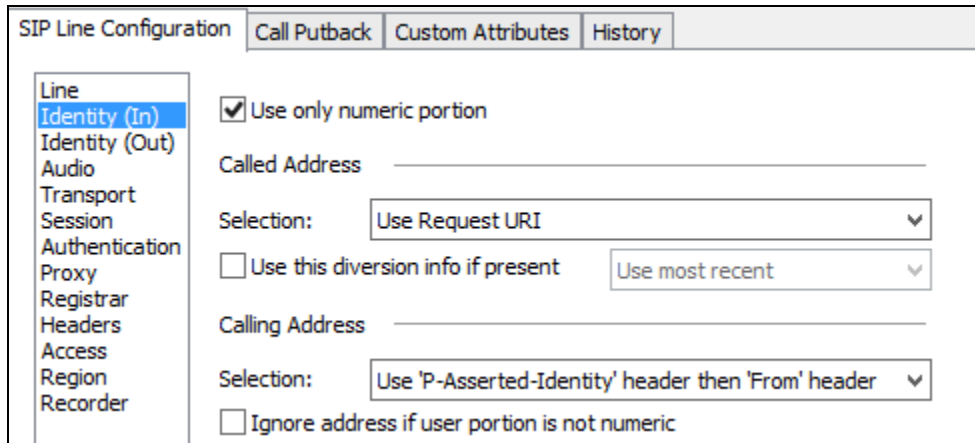


Figure 7: Line Configuration – Identity (In)

3.1.2.3 Identity (Out)

1. In left navigation pane, select **Identity (Out)**
2. Set **Line Value 1**
3. Set **Name**: Oracle is given for this example
4. Set **Address**: Enter the pilot number of the DID range assigned by ITSP
5. Click **OK**
6. All the other values are set to default values

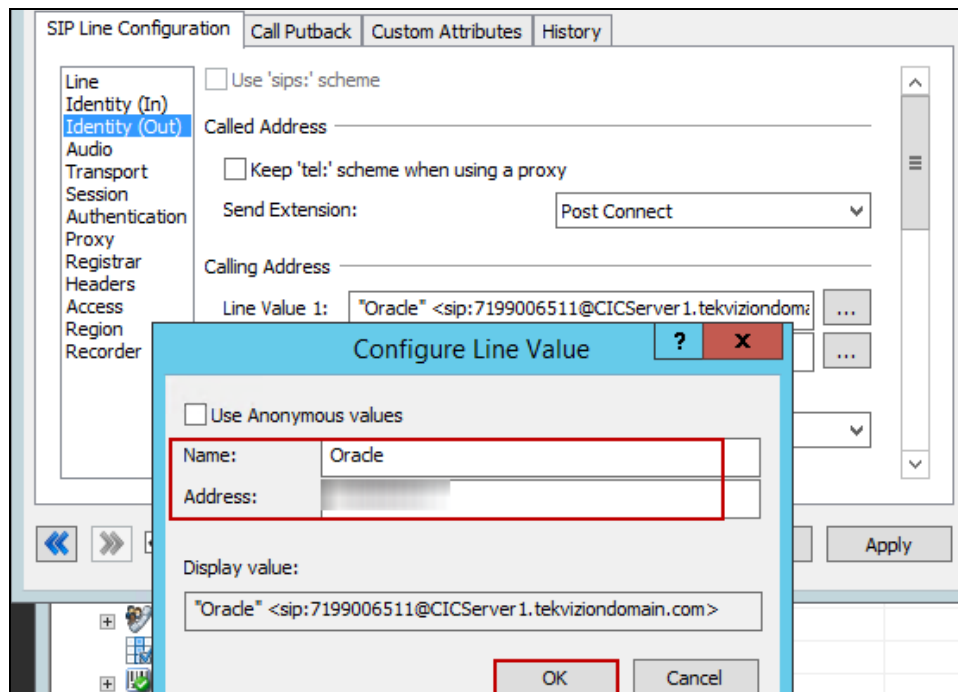


Figure 8: Line Configuration - Identity (Out)

The screenshot shows the 'Line Configuration - Oracle_SBC' interface. The 'SIP Line Configuration' tab is active, and the 'Identity (Out)' sub-tab is selected in the left-hand menu. The main configuration area is titled 'Calling Address (Normal Calls)'. It contains several dropdown menus: 'From' Header Address (Use passed value if present), 'From' Header Name (Use passed value if present), 'P-Asserted-Identity' Header Address (None), 'P-Asserted-Identity' Header Name (None), Diverted Header Address (None), and Diverted Header Name (None). There is also a text input field for 'Calling Address (Normal Calls)' at the top and another for 'Calling Address (Diverted Calls)' at the bottom.

Figure 9: Line Configuration - Identity (Out) – Cont.

This screenshot continues the configuration for 'Identity (Out)'. The 'SIP Line Configuration' tab is active, and the 'Identity (Out)' sub-tab is selected. The main configuration area is titled 'Calling Address (Diverted Calls)'. It contains several dropdown menus: 'Diverted Header Name' (None), 'From' Header Address (Use passed value if present), 'From' Header Name (Use passed value if present), 'P-Asserted-Identity' Address (None), 'P-Asserted-Identity' Name (Use diverted value), Diverted Header Address (Use diverted value), and Diverted Header Name (None). The last dropdown menu is highlighted with a red border.

Figure 10: Line Configuration - Identity (Out) – Cont.

3.1.2.4 Audio

1. In left navigation pane, select **Audio**
2. Set **Audio Path**: *Always In* is selected. This disables the Media Bypass.
3. Set **DTMF Type**: *RFC2833 if supported, otherwise inband* is selected
4. Set **DTMF Payload**: 101 is selected for this example
5. All the values are set to default values

SIP Line Configuration		Call Putback	Custom Attributes	History
Line Identity (In) Identity (Out) Audio Transport Session Authentication Proxy Registrar Headers Access Region Recorder	Audio Path:	Always In		
	DTMF Type:	RFC2833 if supported, otherwise inband		
	DTMF Payload:	101		
	<input type="checkbox"/>	Voice Activation Detection (VAD)		
	<input checked="" type="checkbox"/>	Echo Cancellation		
	<input checked="" type="checkbox"/>	Allow Multiple Codecs in Outbound SDP Offer		

Figure 11: Line Configuration – Audio

3.1.2.5 Transport

1. In left navigation pane, select **Transport**
2. All the values are set to default values

SIP Line Configuration		Call Putback	Custom Attributes	History
Line Identity (In) Identity (Out) Audio Transport Session Authentication Proxy Registrar Headers Access Region Recorder	Transport Protocol:	UDP		
	Audio Protocol:	RTP	Security:	Minimal
	Adapter Name:	Ethernet		
	Intel(R) 82574L Gigabit Network Connection			
	Receive Port:	5060	Connect Timer (ms):	2000
	Maximum Packet Retry:	4	T1 Timer (ms):	500
	Maximum Invite Retry:	3	T2 Timer (ms):	1000
	Reinvite Delay (ms):	50		
	Retryable Reason Codes:	480, 500-599		

Figure 12: Line Configuration – Transport

SIP Line Configuration		Call Putback	Custom Attributes	History
Line Identity (In) Identity (Out) Audio Transport Session Authentication Proxy Registrar Headers Access	Retryable Cause Codes:	1-5, 25, 27, 28, 31, 34, 38, 41, 42, 44, 46, 62, 63, 79, 91, 96, 97		
	SIP DSCP Value:	18 (24, 011000) CS3		
	Inbound Progress Timer (ms):	5000		
	<input type="checkbox"/>	No Inbound Progress Timer		
	SIP Answer Delay (ms):	500		

Figure 13: Line Configuration – Transport – Cont.

3.1.2.6 Session

1. In left navigation pane, select **Session**
2. All the values are set to default values

The screenshot shows the 'SIP Line Configuration' window with the 'Session' tab selected. The left navigation pane lists various configuration categories, with 'Session' highlighted. The main configuration area contains the following settings:

- Use SIP Session Timer
- SIP Session Timeout: seconds
- Disconnect on Broken RTP
- Media Timing:
- Media reINVITE Timing:
- Terminate Analysis on Connect
- Disable Media Server Passthru
- ASR Enabled

Figure 14: Line Configuration – Session

3.1.2.7 Authentication

1. In left navigation pane, select **Authentication**
2. All the values are set to default values

The screenshot shows the 'SIP Line Configuration' window with the 'Authentication' tab selected. The left navigation pane lists various configuration categories, with 'Authentication' highlighted. The main configuration area contains the following settings:

- Authentication
- User Name:
- Password:
- Confirm Password:

Figure 15: Line Configuration – Authentication

3.1.2.8 Proxy

1. In left navigation pane, select **Proxy**
2. Set **Address**: Enter the Address of the next hop (i.e. Oracle Acme Packet SBC 3900 LAN IP Address). 10.64.4.127 is used for this example.
3. Set **Port Number**: 5060 is given for this example
4. Click **OK**

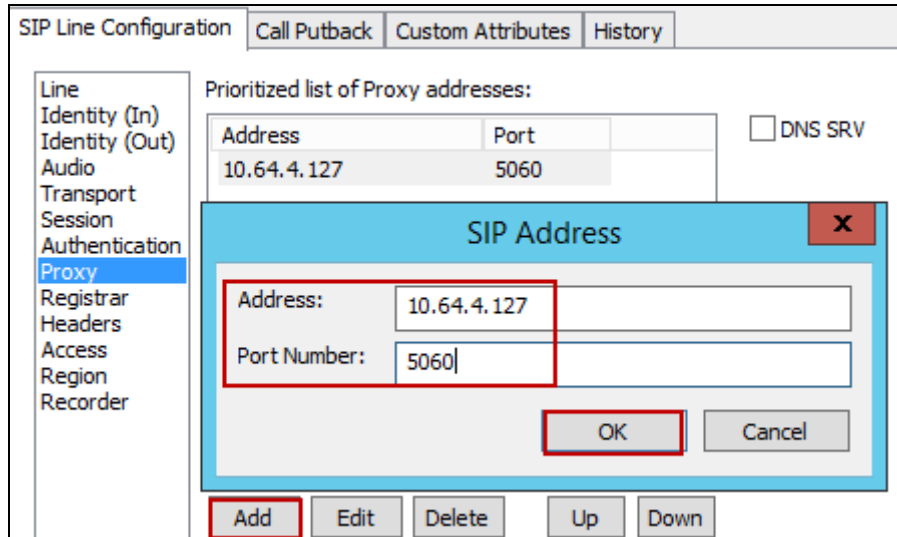


Figure 16: Line Configuration – Proxy

3.1.2.9 Registrar

1. In left navigation pane, select **Registrar**
2. All the values are set to default values

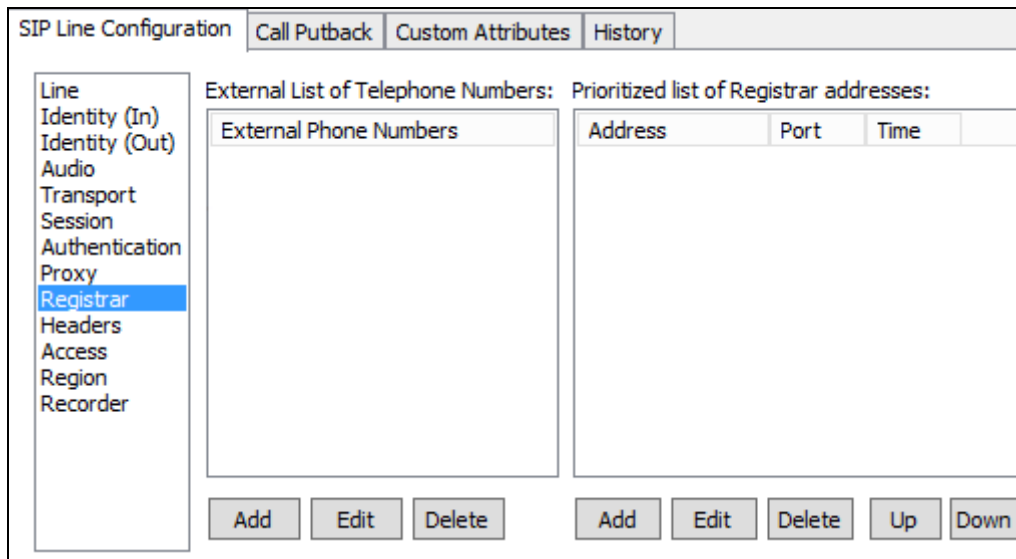


Figure 17: Line Configuration – Registrar

3.1.2.10 Headers

1. In left navigation pane, select **Headers**
2. All the values are set to default values

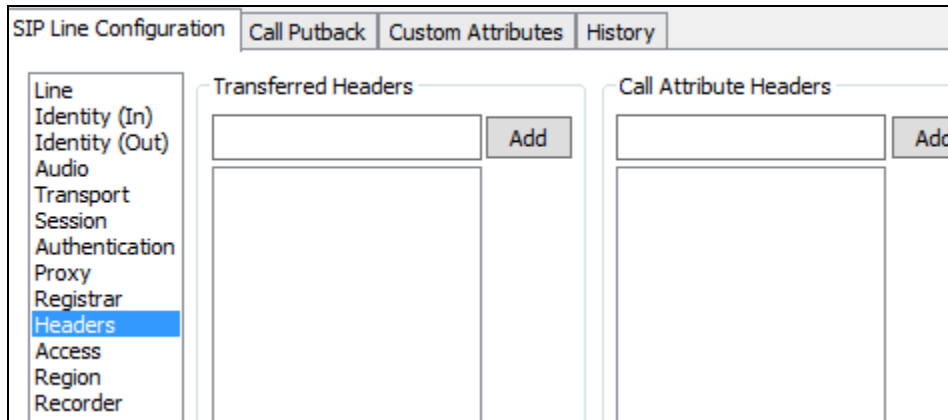


Figure 18: Line Configuration – Headers

3.1.2.11 Access

1. In left navigation pane, select **Access**
2. Under **All Computers will be:** Check **Denied Access**
3. Click **Add**
4. Set **Address:** Enter the Address of the next hop (i.e. Oracle Acme Packet SBC 3900 IP Address). 10.64.4.127 is used for this example.

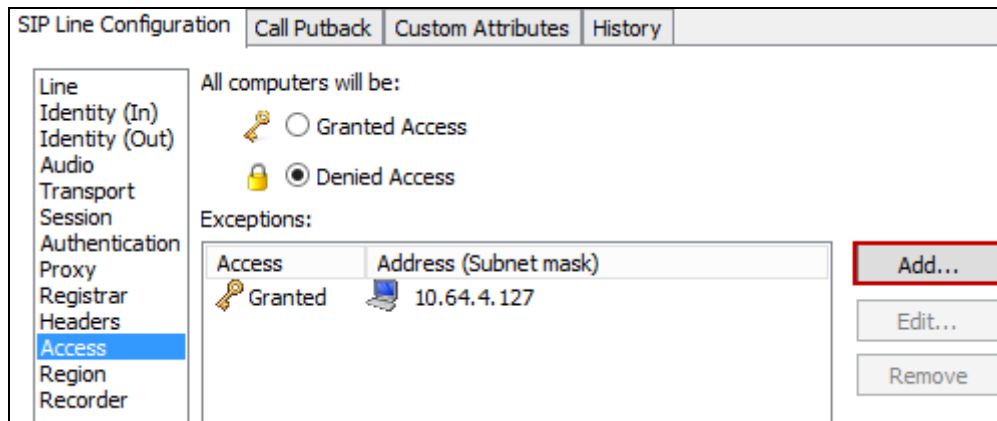


Figure 19: Line Configuration – Access

3.1.2.12 Region

1. In left navigation pane, select **Region**
2. All the values are set to default values as shown in the figure below

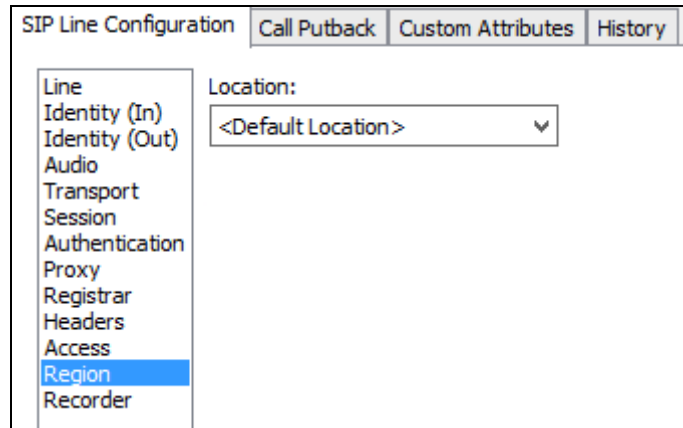


Figure 20: Line Configuration – Region

3.1.2.13 Recorder

1. In left navigation pane, select **Recorder**
2. All the values are set to default values

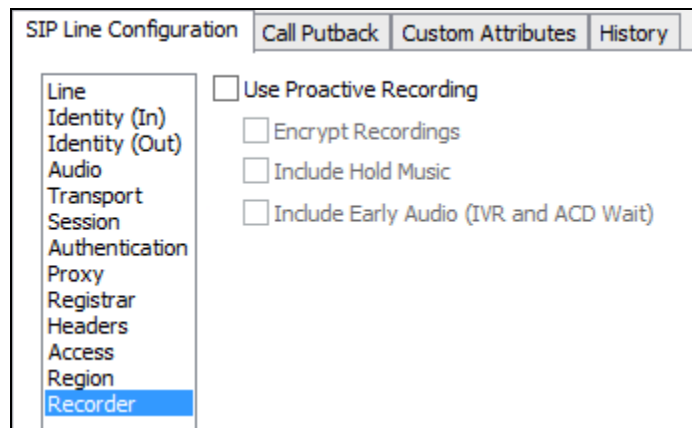


Figure 21: Line Configuration – Recorder

3.1.2.14 Call Putback

3.1.2.14.1 Sending SIP REFER Messages Disabled

1. Select the **Call Putback** tab
2. **Enable sending SIP REFER messages** is unchecked for Internal (Blind & Consultative) Transfers

SIP Line Configuration | Call Putback | Custom Attributes | History

Enable sending SIP REFER messages

Enable sending SIP REFER messages to Lines in other Line Groups

Enable processing of received SIP REFER messages

This feature allows processing of external transfer requests by the IC server, which may result in new calls and additional carrier costs.

Figure 22: Line Configuration – Call Putback

3.1.2.15 Custom Attributes

1. Select the **Custom Attributes** tab
2. All the values are set to default values

SIP Line Configuration | Call Putback | Custom Attributes | History

Name	Value

Add...
Edit...
Delete
Manage Attributes

Figure 23: Line Configuration – Custom Attributes

3.1.3 Line Groups

1. Navigate to **CICSERVER1 > Line Groups**
2. Right-Click and select **New**
3. Set **Enter the Group Name**: Oracle_SBC Group is given for this example
4. Click **OK**

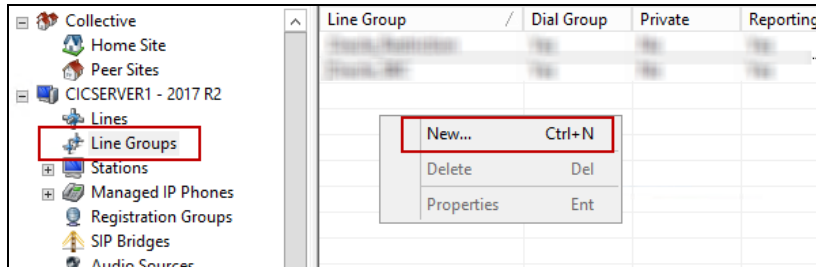


Figure 24: Line Groups

3.1.3.1 Line Group Configuration

1. **Use for Reporting**: Checked
2. **Use as Dial Group**: Checked
3. Under **Hunt Selection Method**: Select *Descending Sequential*
4. Click **Apply**

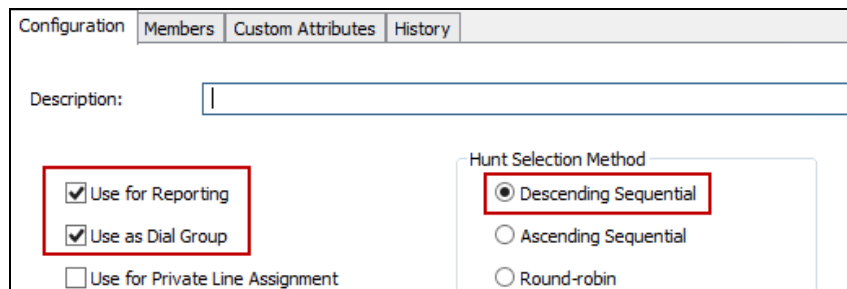


Figure 25: Line Group Configuration

3.1.3.2 Members

1. Select the **Members** tab
2. From **Available Lines**: Select Oracle_SBC and Click **Add** →
3. The **Oracle_SBC** Line should be moved to “Currently Selected Lines” as shown below
4. Click **Apply**

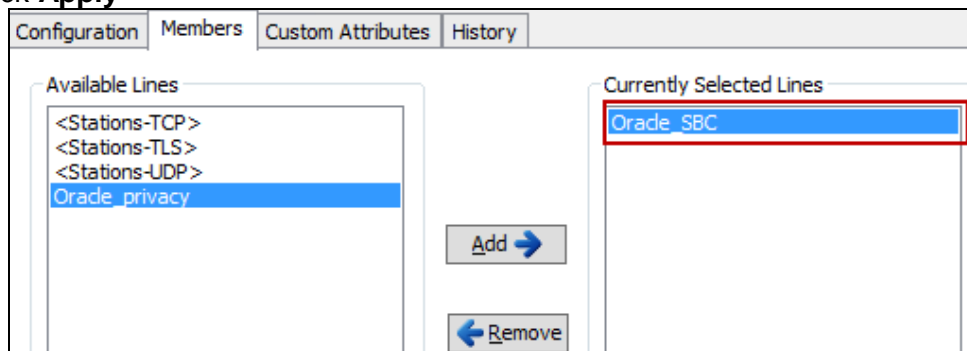


Figure 26: Line Groups – Members

3.1.4 Dial Plan Configuration

1. Navigate to system **Configuration > Phone numbers > Configuration**
2. Under **Regional Dial Plan**: Click Dial Plan
3. To route the 10 digit outbound call out from the IC Server to the Carrier, the Dial plan needs to be configured as shown below. To use this dial plan, dial the access code (9) followed by PSTN number.
4. Click **Add Group** and add the Line Group added in Section 4.1.3

Regional Dial Plan - Edit Pattern

Input Pattern: Standardized Number:

Location Filter: Default Dial String:

List Name: Display String:

Classification: Edit Base:

Account Code Verification Components:

Description:

Dial Group	Filter	Classification	Dial String
Oracle_SBC	<All>	Long Distance	1NxxNxxXXXXZ

Buttons: Add Group, Edit, Remove, Up, Down, <<, >>, **OK**, Cancel, Help

Figure 27: Regional Dial Plan

4 Oracle SBC 3900 Configuration

certificate-record

```

name          ININ
state         TX
locality      Plano
common-name   ininoracle

```

certificate-record

```

name          rootcert
state         TX
locality      Plano
common-name   CICSERVER1_Server Group

```

local-policy

```

from-address  *
to-address    10.64.4.127
source-realm  *
policy-attribute
  next-hop    152.188.29.XXX
  realm       outside

```

local-policy

```

from-address  *
to-address    *
source-realm  inside
policy-attribute
  next-hop    152.188.29.XXX
  realm       outside

```

local-policy

```

from-address  *
to-address    *
source-realm  outside
policy-attribute
  next-hop    SAG:ININ
  realm       inside

```

media-manager

```

hnt-rtcp     enabled
anonymous-sdp enabled

```

media-sec-policy

name	SRTP
inbound	
profile	SRTP
mode	srtp
protocol	sdes
outbound	
profile	SRTP
mode	srtp
protocol	sdes
media-sec-policy	
name	noSRTP
network-interface	
name	s0p0
description	WAN
ip-address	192.65.79.XXX
netmask	255.255.255.128
gateway	192.65.79.XXX
icmp-address	192.65.79.XXX
network-interface	
name	s0p1
description	LAN
ip-address	10.64.4.127
netmask	255.255.255.0
gateway	10.64.1.1
hip-ip-list	10.70.65.20
	10.64.4.127
	10.64.4.238
icmp-address	10.70.65.20
	10.64.4.127
	10.64.4.238
telnet-address	10.70.65.20
phy-interface	
name	s0p0
operation-type	Media
duplex-mode	
speed	


```

phy-interface
  name          s0p1
  operation-type Media
  port          1
realm-config
  identifier     inside
  network-interfaces s0p1:0
  mm-in-realm   enabled
  restricted-latching sdp
  restriction-mask 24
  options       refer-reinvite
  delay-media-update enabled
realm-config
  identifier     outside
  network-interfaces s0p0:0
  mm-in-realm   enabled
  restricted-latching sdp
  restriction-mask 24
  options       refer-reinvite
  delay-media-update enabled
sdes-profile
  name          SRTP
  crypto-list   AES_CM_128_HMAC_SHA1_80
               AES_CM_128_HMAC_SHA1_32
session-agent
  hostname      10.64.4.4
  ip-address    10.64.4.4
  realm-id      inside
  ping-method   OPTIONS;hops=0
  ping-interval 30
session-agent
  hostname      10.64.4.5
  ip-address    10.64.4.5
  realm-id      inside
  ping-method   OPTIONS;hops=0
  ping-interval 30

```

```

session-agent
  hostname          152.188.29.XXX
  ip-address        152.188.29.XXX
  port              5072
  realm-id          outside
  ping-method       OPTIONS;hops=0
  ping-interval     30
session-group
  group-name        ININ
  dest              10.64.4.4
                  10.64.4.5
sip-config
  home-realm-id     inside
  registrar-domain  *
  registrar-host    *
  registrar-port    5060
  options           max-udp-length=0
  extra-method-stats enabled
sip-interface
  realm-id          inside
  sip-port
    address         10.64.4.127
    allow-anonymous agents-only
  sip-port
    address         10.64.4.127
    port            5061
    transport-protocol TCP
    allow-anonymous agents-only
  registration-caching enabled
sip-interface
  realm-id          outside
  sip-port
    address         192.65.79.XXX
    allow-anonymous agents-only
  sip-port
    address         192.65.79.XXX

```

port	8060
allow-anonymous	registered
nat-traversal	always
registration-caching	enabled
route-to-registrar	enabled
out-manipulationid	NAT_IP
sip-profile	Replaces
sip-manipulation	
name	AlterRefer
header-rule	
name	storeReferTo
header-name	Refer-To
action	store
msg-type	request
methods	REFER
match-value	(.*)(@.*)
header-rule	
name	DelReferTo
header-name	Refer-To
action	delete
msg-type	request
methods	REFER
header-rule	
name	AddReferTo
header-name	Refer-To
action	add
msg-type	request
methods	REFER
new-value	\$storeReferTo.\$1+"@"+\$REMOTE_IP+">"
header-rule	
name	delxinin
header-name	x-inin-crn
action	delete
msg-type	request
methods	REFER
header-rule	

name	alterReferredBy
header-name	Referred-By
action	manipulate
msg-type	request
methods	REFER
element-rule	
name	alterReferredBy_er
type	uri-host
action	find-replace-all
new-value	\$LOCAL_IP
sip-manipulation	
name	NAT_IP
header-rule	
name	From
header-name	From
action	manipulate
msg-type	request
methods	INVITE
element-rule	
name	From_er
type	uri-host
action	find-replace-all
new-value	\$LOCAL_IP
header-rule	
name	To
header-name	To
action	manipulate
msg-type	request
methods	INVITE
element-rule	
name	To_er
type	uri-host
action	find-replace-all
new-value	\$REMOTE_IP
header-rule	
name	Div

header-name	Diversion
action	manipulate
msg-type	request
methods	INVITE
element-rule	
name	Div_er
type	uri-host
action	find-replace-all
new-value	\$LOCAL_IP
header-rule	
name	storeReferredBy
header-name	Referred-By
action	store
msg-type	request
methods	INVITE
match-value	(.*)(@.*)
header-rule	
name	delReferredBy
header-name	Referred-By
action	delete
msg-type	request
methods	INVITE
header-rule	
name	addPai
header-name	P-Asserted-Identity
action	add
comparison-type	boolean
msg-type	request
methods	INVITE
match-value	\$storeReferredBy
new-value	\$storeReferredBy.\$1+"@"+\$LOCAL_IP
header-rule	
name	alterReplaces
header-name	Refer-To
msg-type	request
methods	REFER

element-rule	
name	alterReplaces_er
parameter-name	Replaces
type	header-value
action	delete-element
header-rule	
name	CallAlterRefer
header-name	Refer-To
action	sip-manip
new-value	AlterRefer
sip-monitoring	
sip-profile	
name	Replaces
replace-dialogs	enabled
static-flow	
in-realm-id	outside
in-destination	192.65.79.XXX:8088
out-realm-id	inside
out-source	10.64.4.127
out-destination	10.64.4.4:8088
protocol	TCP
alg-type	NAPT
start-port	40000
end-port	45000
static-flow	
in-realm-id	outside
in-destination	192.65.79.XXX:8089
out-realm-id	inside
out-source	10.64.4.127
out-destination	10.64.4.4:8089
protocol	TCP
alg-type	NAPT
start-port	45002
end-port	49500
steering-pool	
ip-address	10.64.4.127

start-port	49452
end-port	65535
realm-id	inside
steering-pool	
ip-address	192.65.79.XXX
start-port	49542
end-port	65535
realm-id	outside
system-config	
process-log-level	DEBUG
default-gateway	192.65.7X.XX
tls-profile	
name	ININ
end-entity-certificate	ININ
trusted-ca-certificates	rootcert
mutual-authenticate	enabled
tls-version	tlsv12
web-server-config	