



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Computer Instruments 7.0 with Avaya IP Office Server Edition 11.1 – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for Computer Instruments eONE 7.0 to interoperate with Avaya IP Office Server Edition 11.1. Computer Instruments eONE is an IVR development platform that provides self-service IVR and Web applications.

In the compliance testing, Computer Instruments eONE used SIP trunk with Avaya IP Office Server Edition to support inbound and outbound IVR applications.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These Application Notes describe the configuration steps required for the Computer Instruments eONE 7.0 to interoperate with Avaya IP Office Server Edition 11.1. eONE is an IVR development platform that provides self-service IVR and Web applications.

In the compliance testing, eONE used SIP trunk with IP Office to support inbound and outbound IVR applications.

The IP Office Server Edition configuration consisted of two IP Office systems, a primary Linux server and an expansion IP500V2 that were connected via Small Community Network (SCN) trunk.

The eONE solution consisted of distributed components across multiple servers. The eONE solution used in the compliance testing utilized two servers – an eONE server and a Media server. The eONE server is responsible for eONE configuration via a web-based interface and included the CIMedia MRCP Connector for support of text-to-speech (TTS). The Media server is responsible for SIP trunk connection with IP Office and included the CIMedia ARC SIP Telecom Services for support of SIP protocol and the CIMedia ARC VXML Services for support of VXML.

eONE supports both on-premise and cloud deployments, and the compliance testing used the on-premise deployment method with eONE residing in the DevConnect test lab.

To facilitate testing, two custom applications were developed by Computer Instruments for testing of inbound and outbound applications that included greetings, menu option selection via DTMF, announcements, and transfer to internal and external destinations.

## 2. General Test Approach and Test Results

The feature test cases were performed manually. The eONE inbound application was tested by manually placing calls from users on the PSTN and on both IP Office systems to the eONE inbound application. The eONE inbound application played greeting and collected DTMF input from the caller to decide on the feature to provide, such as announcement playback and transfer to internal or external destinations.

The eONE outbound application was tested by manually requesting callbacks to users on the PSTN and on both IP Office systems. The callback requests were initiated from the Web page associated with the eONE outbound application.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet connection to eONE.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the SIP trunk interface between IP Office and eONE did not include use of any specific encryption features as requested by Computer Instruments.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the third party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the third party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g. jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another, and may affect the reliability or performance of the overall solution. Different network elements (e.g. session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations, and evaluate if the solution is viable to be deployed in the cloud.

The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations.

## **2.1. Interoperability Compliance Testing**

The interoperability compliance test included feature and serviceability testing.

The feature testing included OPTIONS, G.711MU, media shuffling, session refresh, REFER, hold/reconnect, inbound DTMF, dial ahead, outgoing call screening, multiple calls, call forwarding, inbound, outbound, and supervised transfer via REFER to internal and external destinations.

The serviceability testing focused on verifying the ability of eONE to recover from adverse conditions, such as disconnecting and reconnecting the Ethernet connection to eONE.

## **2.2. Test Results**

All test cases were executed and verified.

## **2.3. Support**

Technical support on eONE can be obtained through the following:

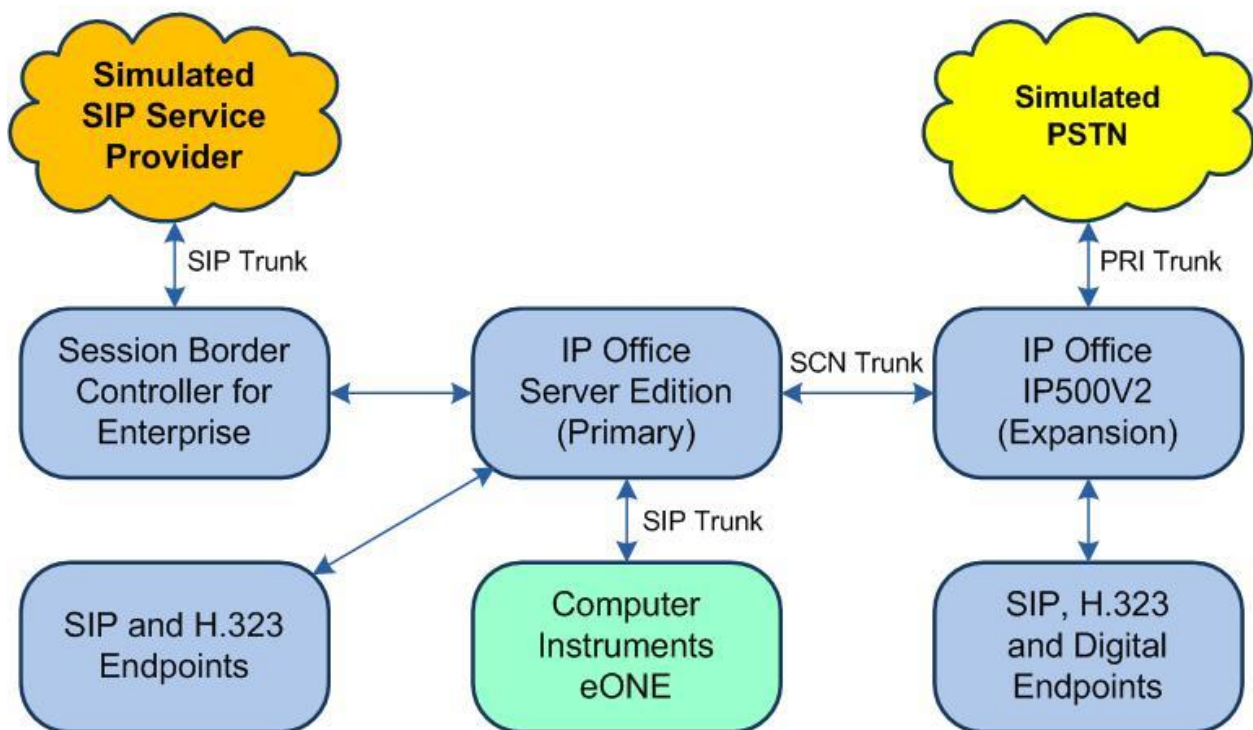
- **Phone:** (888) 451-0851
- **Web:** [http://instruments.com/tech\\_support.html](http://instruments.com/tech_support.html)
- **Email:** [support@instruments.com](mailto:support@instruments.com)

### 3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**. Each IP Office system has connectivity to the PSTN for testing of cross systems PSTN scenarios.

The detailed administration of IP Office resources is not the focus of these Application Notes and will not be described. As shown in **Figure 1** below, SIP trunk was used between the primary IP Office system and eONE.

A five-digit dial plan was used to facilitate routing with eONE. Unique extension ranges were assigned to users on the primary IP Office system (210xx), to users on the expansion IP Office system (220xx), and to eONE (21880).



**Figure 1: Compliance Testing Configuration**

## 4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya IP Office Server Edition (Primary) in Virtual Environment	11.1.0.0.0
Avaya IP Office on IP500 V2 (Expansion)	11.1.0.0.0
Avaya 1120E IP Deskphone (SIP)	4.4.23.0
Avaya J129 IP Deskphone (SIP)	4.0.4.0.10
Avaya 1608-I IP Deskphone (H.323)	1.3120
Avaya 9611G IP Deskphone (H.323)	6.8202
Avaya 1408 Digital Deskphone	48.02
Computer Instruments eONE <ul style="list-style-type: none"><li>• CIMedia ARC SIP Telecom Services on Linux</li><li>• CIMedia arcVXML3.6 Services</li><li>• CIMedia Arc MRCP Connector</li></ul>	7.0 3.6 Build 12 3.6 Build 11 4.0 Build 24

*Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.*

## 5. Configure Avaya IP Office

This section provides the procedures for configuring the IP Office systems. The procedures include the following area:

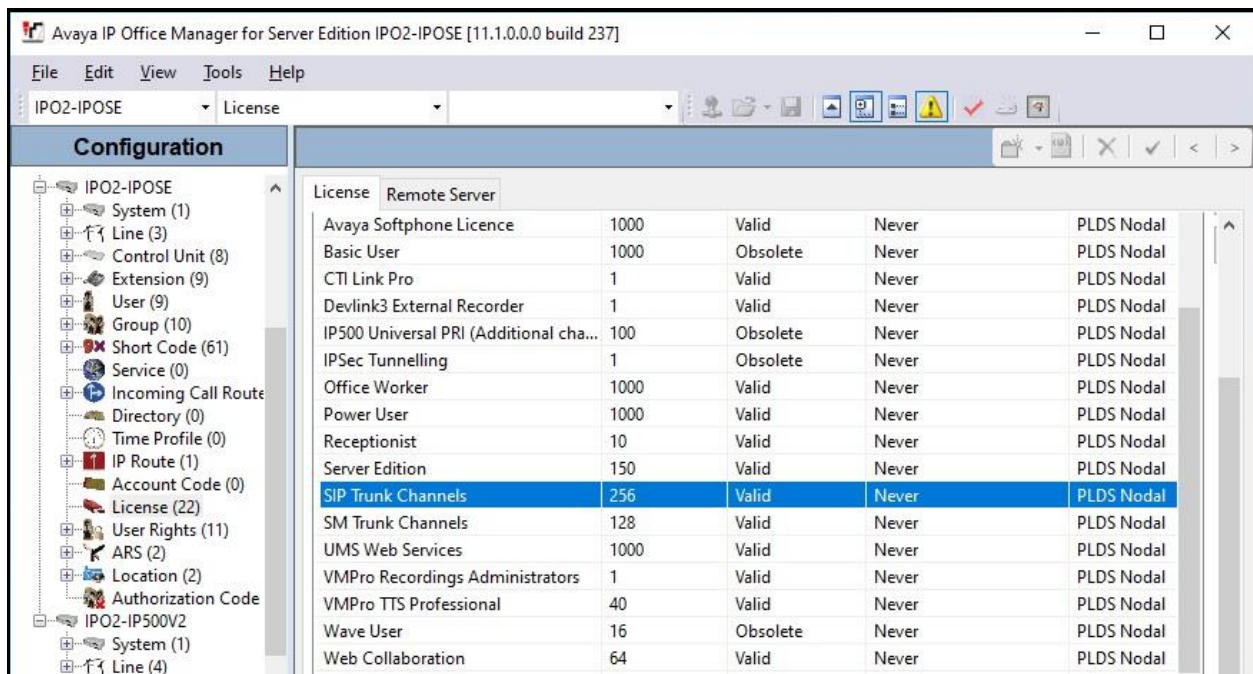
- Verify license
- Administer system
- Administer line
- Administer incoming call route
- Administer short code

### 5.1. Verify License

From a PC running the IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Select the proper primary IP Office system, and log in using the appropriate credentials.

The **Avaya IP Office Manager for Server Edition IPO2-IPOSE** screen is displayed, where **IPO2-IPOSE** is the name of the primary IP Office system.

From the configuration tree in the left pane, select **License** under the primary IP Office system, in this case “IPO2-IPOSE”, and a list of licenses is displayed in the right pane. Verify that there is a license for **SIP Trunk Channels** and that the **Status** is “Valid”, as shown below.

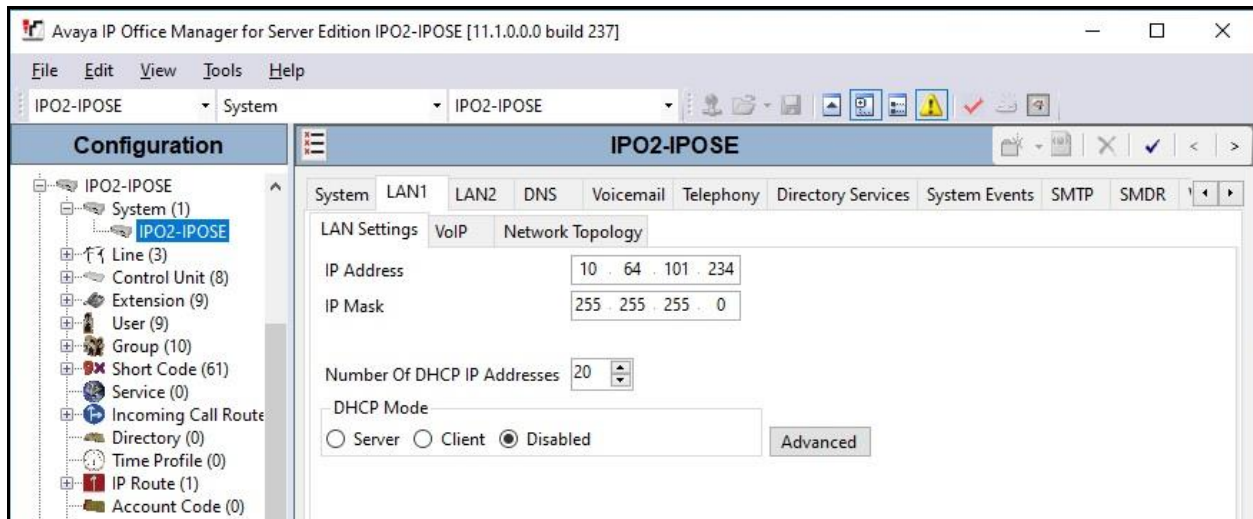


License	Remote Server				
Avaya Softphone Licence	1000	Valid	Never	PLDS Nodal	
Basic User	1000	Obsolete	Never	PLDS Nodal	
CTI Link Pro	1	Valid	Never	PLDS Nodal	
Devlink3 External Recorder	1	Valid	Never	PLDS Nodal	
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal	
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal	
Office Worker	1000	Valid	Never	PLDS Nodal	
Power User	1000	Valid	Never	PLDS Nodal	
Receptionist	10	Valid	Never	PLDS Nodal	
Server Edition	150	Valid	Never	PLDS Nodal	
<b>SIP Trunk Channels</b>	<b>256</b>	<b>Valid</b>	<b>Never</b>	<b>PLDS Nodal</b>	
SM Trunk Channels	128	Valid	Never	PLDS Nodal	
UMS Web Services	1000	Valid	Never	PLDS Nodal	
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal	
VMPro TTS Professional	40	Valid	Never	PLDS Nodal	
Wave User	16	Obsolete	Never	PLDS Nodal	
Web Collaboration	64	Valid	Never	PLDS Nodal	

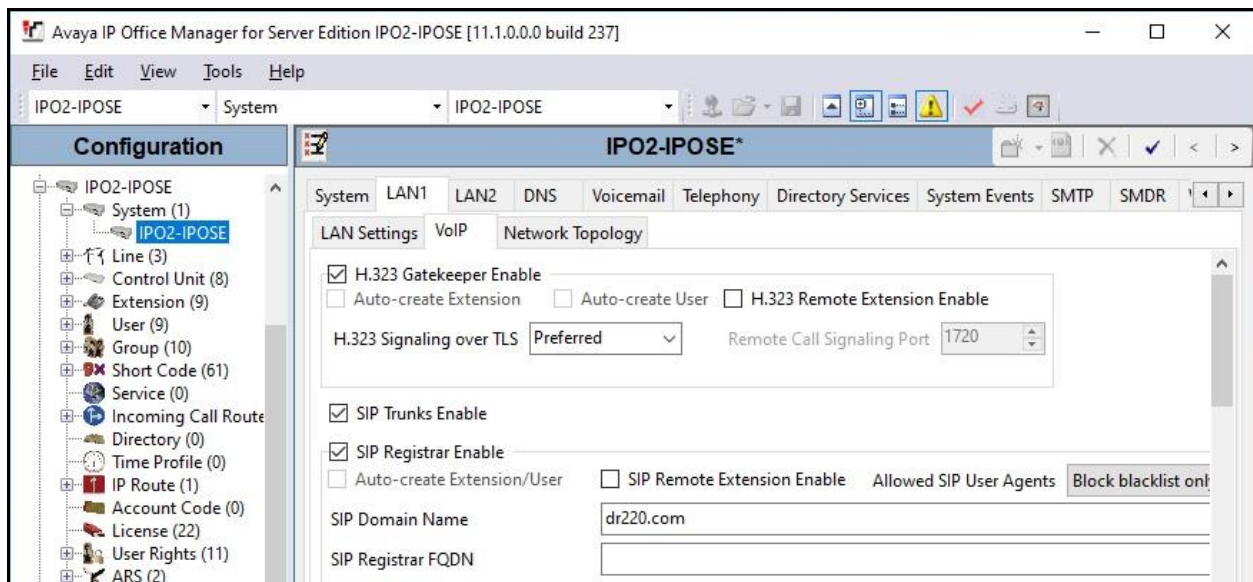
## 5.2. Administer System

From the configuration tree in the left pane, select **System** under the IP Office system used for SIP trunk connection with eONE, to display the system screen in the right pane.

Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure eONE. Note that IP Office can support SIP trunk on the LAN1 and/or LAN2 interfaces, and the compliance testing used the LAN1 interface.

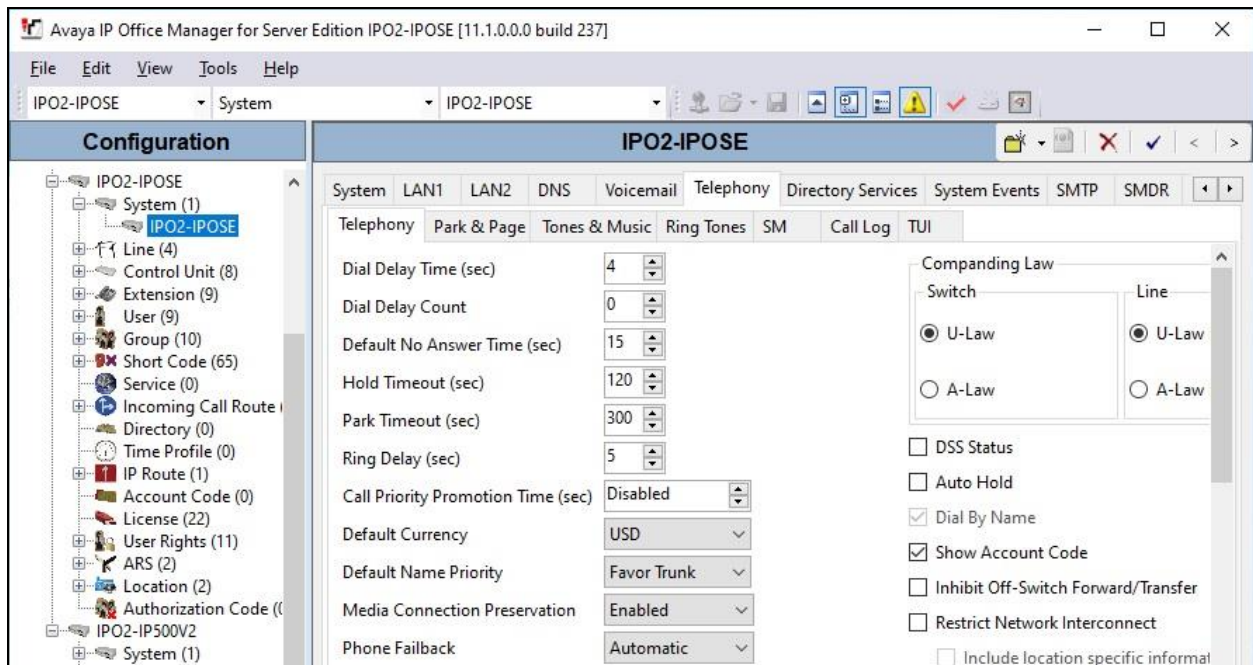


Select the **VoIP** sub-tab. Make certain that **SIP Trunks Enable** is checked, as shown below. Retain the default values in the remaining fields.





Select the **Telephony** tab, followed by the **Telephony** sub-tab in the right pane. Uncheck **Inhibit Off-Switch Forward/Transfer** if transfer from eONE to PSTN destinations is desired. In the compliance testing, this parameter was disabled.



### 5.3. Administer Line

From the configuration tree in the left pane, right-click on **Line** under the IP Office system used for SIP trunk connection with eONE, and select **New → SIP Line** from the pop-up list to add a new SIP line (not shown).

Select the **SIP Line** tab. For **ITSP Domain Name**, enter the IP address of the eONE Media server with the CIMedia ARC SIP Telecom Services component.

For **Incoming Supervised REFER** and **Outgoing Supervised REFER**, select “Always” as shown below.

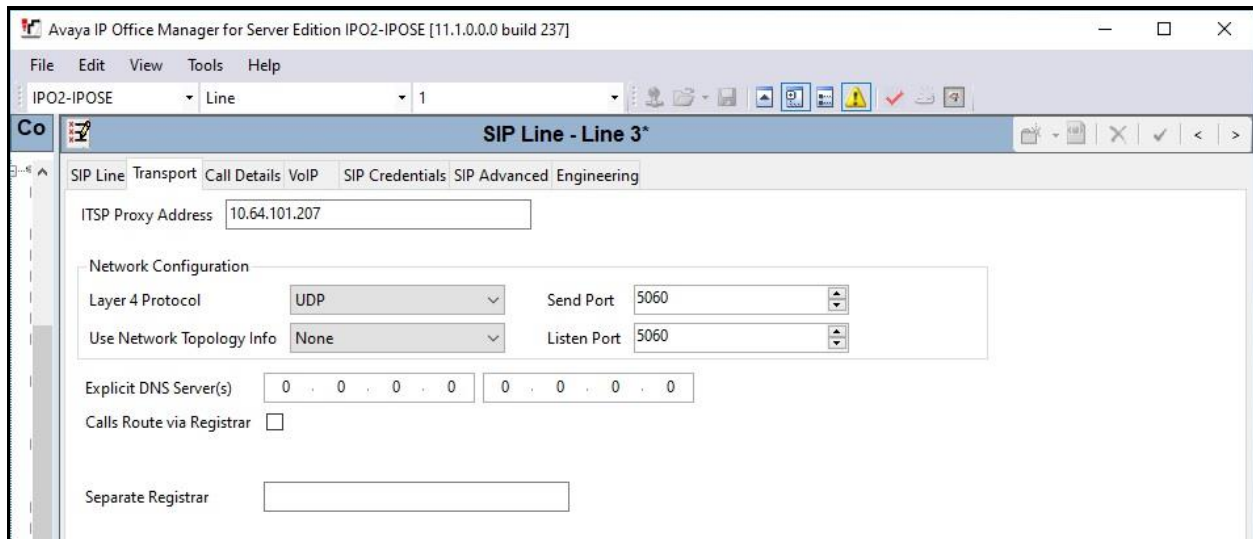
Retain the defaults in the remaining fields.

The screenshot displays the Avaya IP Office Manager for Server Edition interface, specifically the configuration page for 'SIP Line - Line 3'. The window title is 'Avaya IP Office Manager for Server Edition IPO2-IPOSE [11.1.0.0.0 build 237]'. The left pane shows a configuration tree with 'IPO2-IPOSE' selected, and 'Line' is highlighted. The main pane is divided into several tabs: 'SIP Line', 'Transport', 'Call Details', 'VoIP', 'SIP Credentials', 'SIP Advanced', and 'Engineering'. The 'SIP Line' tab is active, showing various configuration fields. The 'Line Number' is set to 3. The 'ITSP Domain Name' is 10.64.101.207. The 'Local Domain Name' is empty. The 'URI Type' is set to 'SIP URI'. The 'Location' is set to 'Cloud'. The 'Prefix' is empty. The 'National Prefix' is 0. The 'International Prefix' is 00. The 'Country Code' is empty. The 'Name Priority' is set to 'System Default'. The 'Description' is eONE. The 'In Service' checkbox is checked. The 'Check OOS' checkbox is checked. The 'Session Timers' section shows 'Refresh Method' set to 'Auto' and 'Timer (sec)' set to 'On Demand'. The 'Redirect and Transfer' section shows 'Incoming Supervised REFER' set to 'Always', 'Outgoing Supervised REFER' set to 'Always', 'Send 302 Moved Temporarily' unchecked, and 'Outgoing Blind REFER' checked.

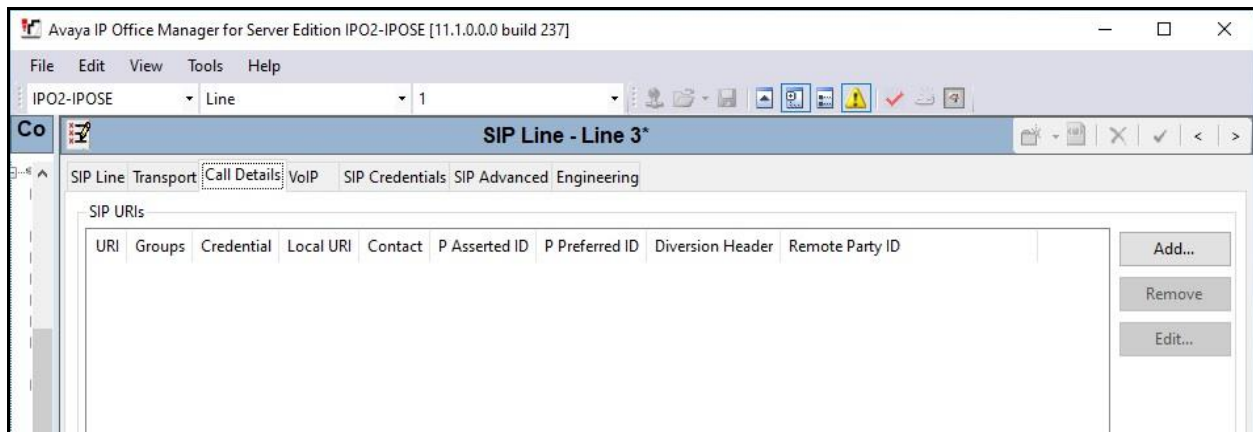
Field	Value
Line Number	3
ITSP Domain Name	10.64.101.207
Local Domain Name	
URI Type	SIP URI
Location	Cloud
Prefix	
National Prefix	0
International Prefix	00
Country Code	
Name Priority	System Default
Description	eONE
In Service	<input checked="" type="checkbox"/>
Check OOS	<input checked="" type="checkbox"/>
Refresh Method	Auto
Timer (sec)	On Demand
Incoming Supervised REFER	Always
Outgoing Supervised REFER	Always
Send 302 Moved Temporarily	<input type="checkbox"/>
Outgoing Blind REFER	<input checked="" type="checkbox"/>

Select the **Transport** tab. For **ITSP Proxy Address**, enter the IP address of the eONE Media server with the CIMedia ARC SIP Telecom Services component. Uncheck **Calls Route via Registrar** as shown below and retain the default values in the remaining fields.

Note that eONE can support TLS, UDP and TCP, and the compliance testing used the UDP protocol.



Select the **Call Details** tab, followed by **Add** in the **SIP URIs** sub-section.



The screen below is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Incoming Group:** An available incoming group number.
- **Outgoing Group:** An available outgoing group number.
- **Max Sessions:** The maximum number of simultaneous calls.

**SIP Line - 3 | Call Details | SIP URI**

New URI

Incoming Group: 3 Max Sessions: 4

Outgoing Group: 3

Credentials: D: <None>

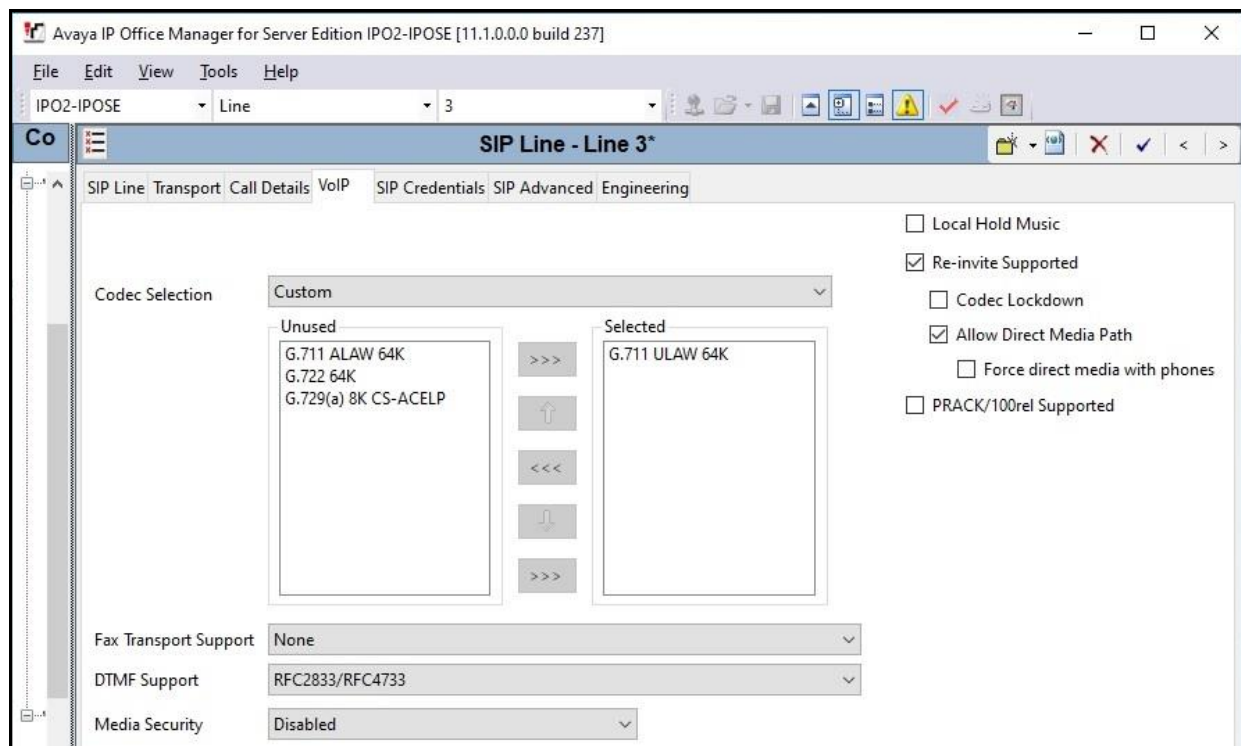
	Display	Content	Field meaning		
			Outgoing Calls	Forwarding/Twinning	Incoming Calls
Local URI	Auto	Auto	Caller	Original Caller	Called
Contact	Auto	Auto	Caller	Original Caller	Called
P Asserted ID	<input type="checkbox"/> None	None	None	None	None
P Preferred ID	<input type="checkbox"/> None	None	None	None	None
Diversion Header	<input type="checkbox"/> None	None	None	None	None
Remote Party ID	<input type="checkbox"/> None	None	None	None	None

OK Cancel Help

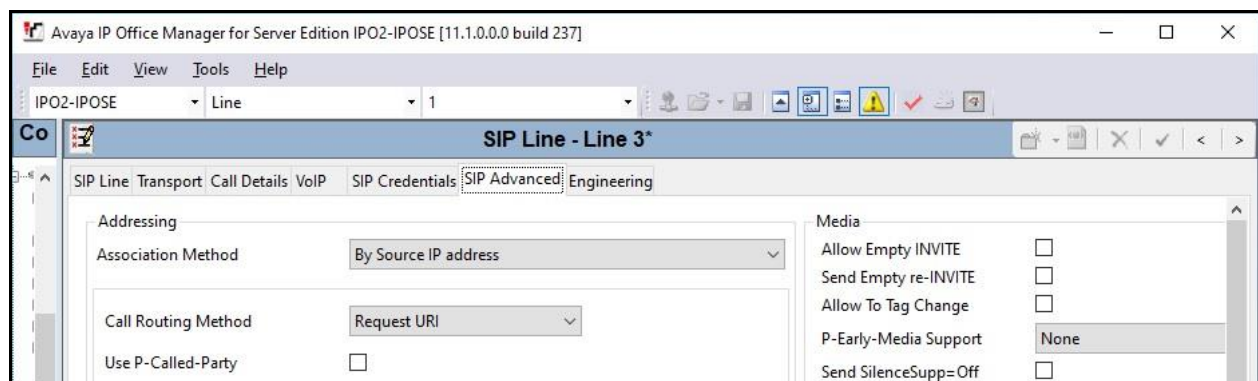
Select the **VoIP** tab. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Codec Selection:** “Custom”
- **Selected:** Retain the relevant G.711 codec variant.
- **Media Security:** “Disabled”
- **Re-invite Supported:** Check this parameter.
- **Allow Direct Media Path:** Check this parameter.

Note that eONE only supports the G.711 codec variant.



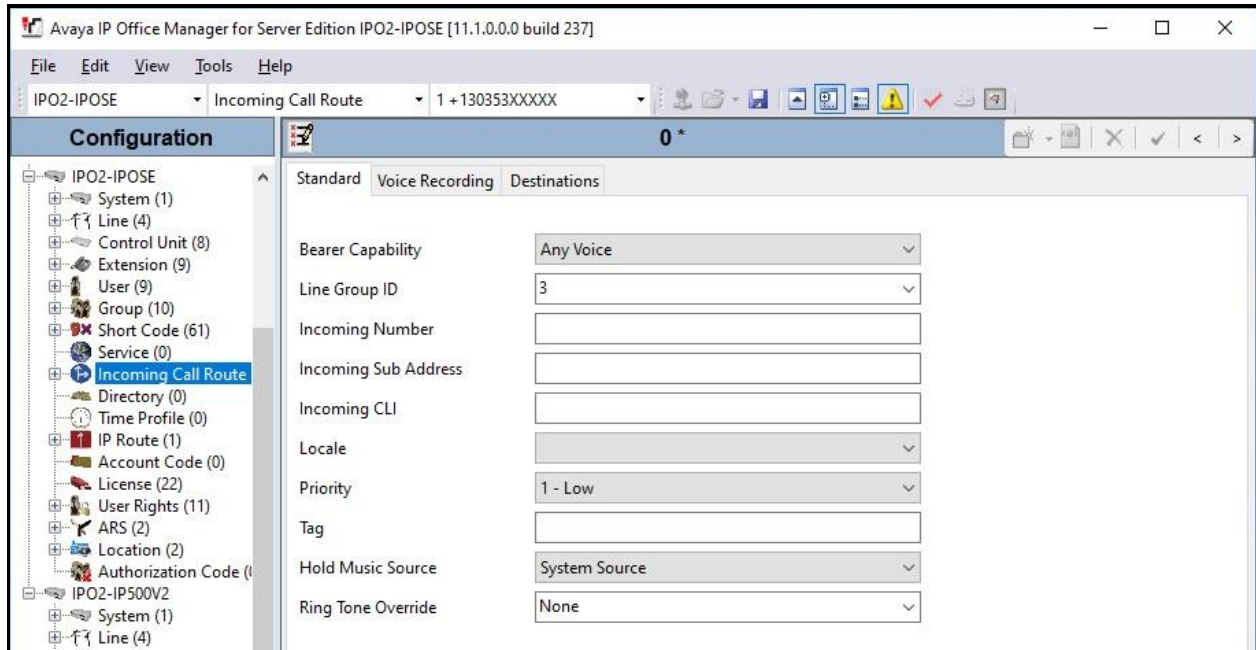
Select the **SIP Advanced** tab. For **Call Routing Method**, select “Request URI” as shown below. Retain the default values in the remaining fields.



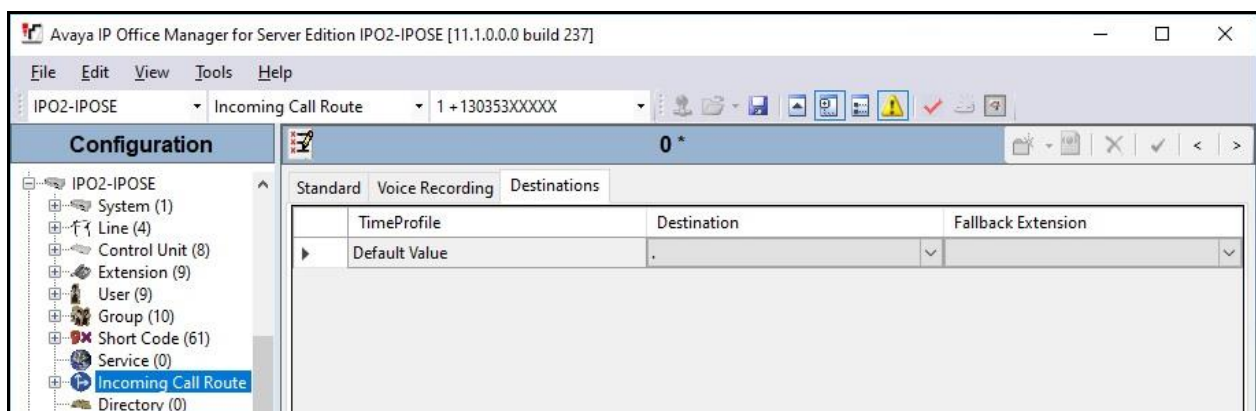
## 5.4. Administer Incoming Call Route

From the configuration tree in the left pane, right-click on **Incoming Call Route** under the IP Office system used for SIP trunk connection with eONE and select **New** from the pop-up list to add a new route for incoming calls from eONE.

For **Line Group Id**, select the incoming group number from **Section 5.3**, in this case “3”. Retain the default value in the remaining fields.



Select the **Destinations** tab. For **Destination**, enter “.” to match any dialed number from eONE.





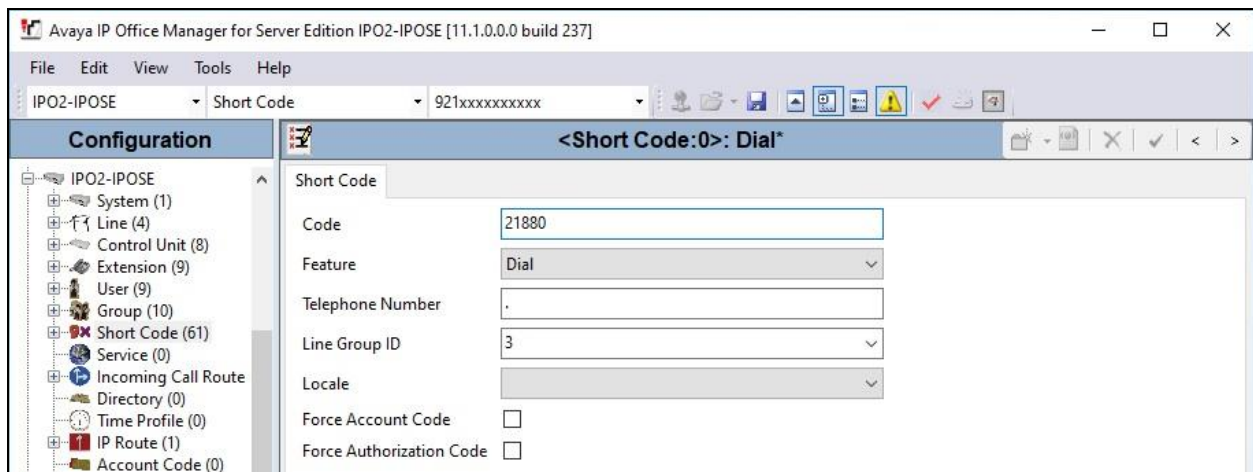
## 5.5. Administer Short Code

Configure a set of short codes for routing of outgoing calls to eONE and for routing of incoming calls from eONE to the PSTN.

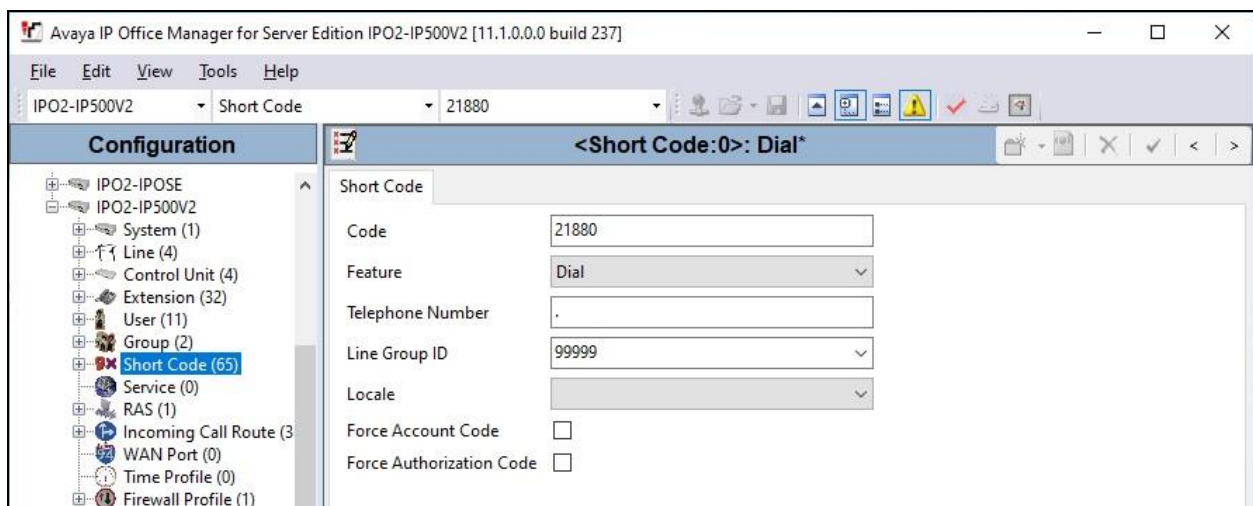
### 5.5.1. Short Code for Outgoing Calls to eONE

From the configuration tree in the left pane, right-click on **Short Code** under the primary IP Office system and select **New** from the pop-up list to add a new short code for outgoing calls to eONE. In the compliance testing, all calls to 21880 are routed over the SIP trunk to eONE.

For **Code**, enter the appropriate value, in this case “21880”. For **Telephone Number**, enter “.” to match the dialed number. For **Line Group ID**, select the outgoing group number from **Section 5.3**. Retain the default values in the remaining fields.



Repeat this section to add similar short code for the expansion IP Office system, which is named **IPO2-IP500V2** in this case. For **Line Group ID**, select the applicable outgoing group ID for the SCN trunk that connects to the primary IP Office system, in this case “99999” as shown below.



### 5.5.2. Short Code for Incoming Calls from eONE to PSTN

From the configuration tree in the left pane, right-click on **Short Code** under the primary IP Office system and select **New** from the pop-up list to add a new short code for incoming calls from eONE to the PSTN. In the compliance testing, eONE will add the prefix “91” for outbound call to the PSTN, and therefore a new short code is created for routing of such call.

For **Code**, enter the appropriate value, in this case “91N”.

For **Telephone Number**, enter pertinent value with desired manipulation of called and calling numbers. In the compliance testing, “9NS3035321880” was used to strip “1” from the called number prefix sent by eONE and to use “3035321880” as the calling number, as shown below.

For **Line Group ID**, enter the outgoing group number for the PSTN line, in this case “8”.

Retain the default values in the remaining fields.

The screenshot displays the Avaya IP Office Manager for Server Edition interface. The title bar indicates the version is IPO2-IPOSE [11.1.0.0.0 build 237]. The menu bar includes File, Edit, View, Tools, and Help. Below the menu bar, there are dropdowns for 'IPO2-IPOSE' and 'Short Code', with '91N' selected in the latter. The main window is divided into two panes. The left pane, titled 'Configuration', shows a tree view of the system hierarchy: IPO2-IPOSE, System (1), Line (4), Control Unit (8), Extension (9), User (9), Group (10), Short Code (65), Service (0), Incoming Call Routing (0), Directory (0), Time Profile (0), IP Route (1), Account Code (0), License (22), and User Rights (11). The 'Short Code (65)' item is selected. The right pane, titled '91N: Dial', shows the configuration fields for the selected short code. The fields are: Code (91N), Feature (Dial), Telephone Number (9NS3035321880), Line Group ID (8), and Locale (empty). There are also checkboxes for Force Account Code and Force Authorization Code, both of which are unchecked.

Field	Value
Code	91N
Feature	Dial
Telephone Number	9NS3035321880
Line Group ID	8
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>



## 6. Configure Computer Instruments eONE

This section provides the procedures for configuring eONE. The procedures include the following areas:

- Launch web interface
- Administer company management
- Administer system configuration
- Administer collect and store
- Administer extension manager
- Administer menu manager
- Administer VXML file

The configuration of eONE is typically performed by Computer Instruments deployment engineers, and the procedural steps are presented in these Application Notes for informational purposes.

### 6.1. Launch Web Interface

Access the eONE web interface by using the URL **http://ip-address/eci/voiceadmin/LoginPage.aspx** in a browser window, where “ip-address” is the IP address of the eONE server with the web-based interface.

The **Login** screen below is displayed. Log in using the appropriate credentials.



The screenshot shows a web-based login interface. It features a light blue background. In the center, there is a white rectangular box with a blue header bar containing the word "Login" in white. Below the header, there are two input fields: "User ID:" followed by a text box, and "Password:" followed by a text box. Below these fields is a button labeled "Login".

The **Welcome to Computer Instruments** screen is displayed next.

Note that the relevant tenant in this case is “SIPIPO (43)” as shown below, which was pre-configured.



## 6.2. Administer Company Management

Expand and select **Web Administrator** → **Company Management** from the left pane to display the **Company Management** screen. Scroll down to the bottom of the screen to select the pertinent company entry and click **Edit** (not shown).

For **PBX Domain/IP**, enter the IP address of the primary IP Office system from **Section 5.2**.

For **Time Zone**, select the appropriate zone. For **ASR & TTS**, uncheck resources that are not used. In the compliance testing, only **TTS** was used.

For **Transfer Type**, select **Supervised Transfer**.

Retain the pre-configured values in the remaining fields.

The screenshot displays the 'Company Management' web interface. On the left is a navigation pane with the following menu items: Voice Administrator, Web Administrator (expanded), Company Management (selected), Company Access Rights, User Maintenance, Import Tenant Data, User Activity Report, Color Scheme, Community Notification, and Log-Out. A 'Hide Menu' button is located next to the 'Web Administrator' section. The main content area is titled 'Company Management' and contains the following fields and options:

- Company Name: SIPIPO
- PBX Domain/IP: 10.64.101.234
- Prompt recordings Path: D:\Program Files\CII\Speech\SIPIPO\
- Exports Path: D:\Exports\SIPIPO\
- Imports Path: D:\Imports\SIPIPO\
- VTSystem Database Host (Name or IP): localhost
- ☒ Use Standard 'VTSystem' credentials
- User ID for VTSystem: [empty field]
- VTSystem Password: [masked with dots]
- Confirm VTSystem Password: [masked with dots]
- ☒ Use Standard 'User' credentials for 'subscriber' schema
- User ID for 'subscriber' Schema: [empty field]
- User Password for 'subscriber' Schema: [masked with dots]
- Confirm User Password: [masked with dots]
- Time Zone: Eastern Standard Time (dropdown menu)
- Form filler type: Standard (dropdown menu)
- ASR & TTS: ☐ ASR ☒ TTS
- ☐ Google Resources
- OC Priority: 5
- Max OCs: [empty field]
- Transfer Type: ☐ Blind Transfer ☒ Supervised Transfer

On the right side of the main content area, there is a section titled 'Tenant Creation & Copy eONE Data: Log and Status' with a large empty box below it.

### 6.3. Administer System Configuration

Expand and select **Voice Administrator** → **System Config** from the left pane to display the **Base System Defaults** screen.

Enter the following values for the specified fields and retain the default values for the remaining fields.

- **PBX Integration:** “Avaya IP Office”
- **Dial Plan Digits:** The maximum length of internal extensions, in this case “5”.
- **Outside Line Access Prefix:** The applicable prefix for calls to the PSTN, in this case “9”.
- **Outbound From:** The eONE extension from **Section 3**.

Note that for outbound calls from eONE to the PSTN, eONE will insert the value of the **Outside Line Access Prefix** plus the digit “1” as the called number.

The screenshot displays the 'Base System Configuration' web interface. The left sidebar contains a navigation menu with options: Voice Administrator, System Config, Voice Reports, Prompt Manager, Menu Manager, Audio Manager, Extension Manager, Form Manager, Locator Manager, CollectAndStore Config, Configurations, Grammar Manager, Import Manager, Web Administrator, and Log-Out. The main content area is titled 'Base System Configuration' and has tabs for Defaults, Application, Channel, Dialing, and ASR User Directory. The 'System Defaults' tab is active, showing various configuration fields. The 'PBX Integration' is set to 'Avaya IP Office'. 'Default Application' is '1000 - Default Application', 'Default Operator' is '100 - OPERATOR, DEFAULT', and 'Default Language' is 'English'. 'Default Gender' has radio buttons for Male and Female, with Female selected. 'Dial Plan Digits' is set to 5 and 'Max Mode Digits' is 15. 'Outside Line Access Prefix' is 9. There are checkboxes for 'Transfer Dress' (checked) and 'Transfer Fix Phone' (checked). 'Outbound From' is 21880. On the right, there are sections for 'Max Tries' (Count: 3, Action: Direct Transfer, Parameter: 100 - Operator, Default), 'Tech Trouble' (Action: Direct Transfer, Parameter: 100 - Operator, Default, Enable SOC unchecked), 'Resources' (ASR: No ASR, TTS: Default TTS), and 'Company / Tenant Notes'. A 'Save Settings' button is at the bottom right.

Field	Value
PBX Integration	Avaya IP Office
Default Application	1000 - Default Application
Default Operator	100 - OPERATOR, DEFAULT
Default Language	English
Default Gender	Female
Dial Plan Digits	5
Max Mode Digits	15
Outside Line Access Prefix	9
Transfer Dress	Checked
Transfer Prefix	
Transfer Suffix	
Toll Call Suffix	
Local Call Suffix	
Intl. Call Prefix	
Intl. Call Suffix	
Clear UV Call Data	Unchecked
Transfer Fix Phone	Checked
Outbound From	21880
Consultation tfr. Audio	
Max Tries Count	3
Max Tries Action	Direct Transfer
Max Tries Parameter	100 - Operator, Default
Tech Trouble Action	Direct Transfer
Tech Trouble Parameter	100 - Operator, Default
Enable SOC	Unchecked
ASR	No ASR
TTS	Default TTS

Select the **Channel** tab. In the **DNIS/Channel Settings** sub-section, select the default entry. For **Application**, select the applicable pre-configured inbound application, in this case “1011 – SIL\_Inbound”.

For **Extension**, enter the eONE extension from **Section 3**.

Retain the default values in the remaining fields.

The screenshot displays the 'Base System Configuration' window with the 'Channel' tab selected. The 'DNIS/Channel Setting' sub-section is active, showing a table with columns 'EXTENSION', 'APPLICATION', and 'Ch. #'. The table contains one entry: '0' for extension and 'Default Application' for application. Below the table, there is an 'Add New Channel' button, an 'Application' dropdown menu set to '1011 - SIL\_Inbound', an 'Extension' text field containing '21880', and an 'Update' button. To the right, the 'Report Setting' sub-section is visible, showing a table with columns 'NUMBER' and 'APPLICATION', containing one entry: '0' for number and 'Default Appli' for application. Below this table, there are 'DNIS' and 'Application' dropdown menus, and 'Save' and 'Delete' buttons.

EXTENSION	APPLICATION	Ch. #
0	Default Application	

Application: 1011 - SIL\_Inbound  
Extension: 21880

NUMBER	APPLICATION
0	Default Appli

DNIS:   
Application:

## 6.4. Administer Collect and Store

Expand and select **Voice Administrator → CollectAndStore Config** from the left pane to display the **Collect And Store** screen.

For **Description**, select the pertinent pre-configured entry associated with the inbound application, in this case “1006-SIL\_InboundGetANIDNIS”.

In the custom inbound application, parameter **UV4** stores the external transfer-to destination. Update content of **UV4** to the desired external destination as shown below. Retain the default values in the remaining fields.

The screenshot shows the 'Collect And Store' configuration window. On the left is a navigation pane with 'Voice Administrator' expanded and 'CollectAndStore Config' selected. The main area is titled 'Collect and store configuration:'. It contains a 'Description' dropdown set to '1006-SIL\_InboundGetANIDNIS', a 'Type' dropdown set to 'ECI Call Header Collection', and a 'Definition' text area containing a comma-delimited list of session properties. Below the definition is a 'Parameter' dropdown set to '1022 - SIL\_InboundWelcome'. At the bottom are 'Save' and 'Cancel' buttons.

Collect and store configuration:

Description: 1006-SIL\_InboundGetANIDNIS [New] [Rename]

Type: ECI Call Header Collection

Definition:

Supply comma delimited list of %UVX%=Property Values.Valid properties are:CTI\_UCID, CTI\_CHANNEL, CTI\_DATETIME, CTI\_ANI, CTI\_DNIS, CTI\_CALLID, CTI\_STATE, CTI\_STATIONEXTENSION, CTI\_UII, CTI\_SESSIONID, CTI\_SESSIONLABEL, SESSION\_AAI, SESSION\_ANI, SESSION\_CALLTAG, SESSION\_CHANNEL, SESSION\_CONVERSEFIRST, SESSION\_CONVERSESECOND, SESSION\_CURRENTLANGUAGE, SESSION\_DNIS, SESSION\_EXITREASON, SESSION\_EXITINFO1, SESSION\_EXITINFO2, SESSION\_EXITCUSTOMERID, SESSION\_EXITPREFERREDPATH, SESSION\_EXITTOPIC, SESSION\_LASTERROR, SESSION\_MEDIATYPE, SESSION\_PROTOCOLNAME,

%UV1%=SESSION\_ANI,%UV2%=SESSION\_DNIS,%UV3%='1234',%UV4%='2126630031'

Action: Direct Audio

Parameter: 1022 - SIL\_InboundWelcome

[Save] [Cancel]

Repeat the procedures in this section to update the external transfer-to destination associated with the outbound application, as shown below.

The screenshot shows the 'Collect And Store' configuration window for an outbound application. The 'Description' dropdown is set to '1007-SIL\_OutboundSetData'. The 'Type' dropdown is set to 'ECI Call Header Collection'. The 'Definition' text area contains a comma-delimited list of session properties. The 'Parameter' dropdown is set to '1023 - SIL\_OutboundWelcome'. At the bottom are 'Save' and 'Cancel' buttons.

Collect and store configuration:

Description: 1007-SIL\_OutboundSetData [New] [Rename]

Type: ECI Call Header Collection

Definition:

Supply comma delimited list of %UVX%=Property Values.Valid properties are:CTI\_UCID, CTI\_CHANNEL, CTI\_DATETIME, CTI\_ANI, CTI\_DNIS, CTI\_CALLID, CTI\_STATE, CTI\_STATIONEXTENSION, CTI\_UII, CTI\_SESSIONID, CTI\_SESSIONLABEL, SESSION\_AAI, SESSION\_ANI, SESSION\_CALLTAG, SESSION\_CHANNEL, SESSION\_CONVERSEFIRST, SESSION\_CONVERSESECOND, SESSION\_CURRENTLANGUAGE, SESSION\_DNIS, SESSION\_EXITREASON, SESSION\_EXITINFO1, SESSION\_EXITINFO2, SESSION\_EXITCUSTOMERID, SESSION\_EXITPREFERREDPATH, SESSION\_EXITTOPIC, SESSION\_LASTERROR, SESSION\_MEDIATYPE, SESSION\_PROTOCOLNAME,

%UV3%='2298',%UV4%='7037030032'

Action: Direct Audio

Parameter: 1023 - SIL\_OutboundWelcome

[Save] [Cancel]



## 6.5. Administer Extension Manager

Expand and select **Voice Administrator** → **Extension Manager** from the left pane to display the **Extension Manager** screen.

Follow reference [3] to create an entry for every internal extension that can be used by eONE as transfer destination. In the compliance testing, a total of five entries were created and shown below is one of the entries.

The screenshot displays the 'Extension Manager' web interface. On the left is a dark blue sidebar with a 'Hide Menu' button and a list of navigation items: Voice Administrator, System Config, Voice Reports, Prompt Manager, Menu Manager, Audio Manager, Extension Manager (highlighted), Form Manager, Locator Manager, CollectAndStore Config, Configurations, Grammar Manager, Import Manager, Web Administrator, and Log-Out. The main content area has a title bar 'Extension Manager' and a 'Select Extension:' dropdown menu currently showing '21031 - Primary, H323' with an 'Add' button. Below this are two tabs: 'Extension' (active) and 'Mailbox'. The 'Extension' tab contains a form with the following fields: 'First Name' (H323), 'Last Name' (Primary), 'Email Address' (empty), 'Email Server(Name or IP)' (empty), 'Email Server Type' (POP3/SMTP dropdown), 'Email Login(User Name)' (empty), and 'Email Password' (empty). Below these fields are three checkboxes: 'Allow Call Transfer' (checked), 'Transcriber' (unchecked), and 'Administrator' (unchecked). There are two expandable sections: 'Re-Route Transfers to Another Extension' and 'Execute An Application', each with a dropdown menu. At the bottom of the form are three buttons: 'Save', 'Renumber', and 'Bulk Add'.

## 6.6. Administer Menu Manager

Select **Voice Administrator** → **Menu Manager** from the left pane to display the **Menu Manager** screen.

For **Menu**, select the pertinent pre-configured menu entry associated with the inbound application, in this case “1003 - SIL\_MENU”.

Under **Spanish prompt and settings**, press the keypad associated with the menu option for transfer to internal destination, in this case “3”.

For **Button Parameter**, select the desired internal destination as shown below.

Repeat the procedures in this section to administer the transfer internal destination for the outbound application where applicable. In the compliance testing, the same menu entry was used for both the inbound and outbound applications.

The screenshot displays the 'Menu Manager' web interface. On the left is a navigation pane with 'Voice Administrator' selected, containing sub-items like System Config, Voice Reports, Prompt Manager, Menu Manager, Audio Manager, Extension Manager, Form Manager, Locator Manager, CollectAndStore Config, Configurations, Grammar Manager, Import Manager, Web Administrator, and Log-Out. The main area is titled 'Menu Manager' and shows configuration for 'Menu: 1003 - SIL\_MENU'. It includes fields for 'TTS: Default TTS', 'Type: Standard DTMF Menu', 'Max Tries Options: Count: 3', 'Action: Disconnect', and 'Parameter: [[DISCONNECT]]'. Below are sections for 'English prompt and settings' and 'Spanish prompt and settings'. The English section contains a list of prompts: 'To listen to Short Announcement press 1', 'For Long Announcement press 2', 'To test Transfer to internal extension press 3', 'For Transfer to external number press 4', and 'To Play text using Text-To-Speech press 5'. The Spanish section has a 'Use Auto-Transfer Keys?' checkbox checked and a keypad with buttons 1-9, \*, 0, and #. A 'Timeout' button is also present. A 'Menu Button 3' section shows 'Button Action: Direct Transfer' and 'Button Parameter: 21031 - Primary H323'. There are also options for 'Change Language / Gender' with dropdowns for 'English' and 'Female'. A 'Save' button is at the bottom right.



## 6.7. Administer VXML File

Log into the Linux shell of the eONE Media server containing the CIMedia ARC VXML Services component.

Navigate to the **/home/arc/.ISP/Telecom/Exec/vxi** directory and use the copy command to create a new VXML configuration file as shown below, where “arcVXML2.vxml.cfg” is the existing default configuration file for ARC VXML 2.0 application and “21880” is the eONE extension from **Section 3**. Note that the eONE extension must be used as part of the name of the new configuration file.

Retain all default values in the newly created VXML configuration file.

```
[xxx@CI-TESTMS ~]# cd /home/arc/.ISP/Telecom/Exec/vxi
[xxx@CI-TESTMS vxi]#
[xxx@CI-TESTMS vxi]# cp arcVXML2.vxml.cfg arcVXML2.21880.vxml.cfg
```

Open the newly created file, in this case “arcVXML2.21880.vxml.cfg”. Scroll down to the bottom of the file.

For **APP\_NAME**, enter any descriptive name. For **SCRIPT**, enter the URL shown below where “10.64.101.206” is the IP address of the eONE server.

Retain the default values in the remaining fields.

```
#APP_NAME=arcVXML2
CACHE_DIR=/tmp/VXML2
KEEP_CACHE_DIR=1
SPEECH_REC=0
RESERVE_SPEECH_RESOURCE=0
VALIDATE_SCRIPTS=0
DEFAULT_COMPRESSION=COMP_WAV
#TRANSFER_MODE=BLIND
TRANSFER_MODE=LISTEN_ALL
#TRANSFER_FORMAT=IP
TRANSFER_FORMAT=NONE
#TTS_SERVER=LOQUENDO,MSS
TTS_SERVER=MSS
TTS_LANGUAGE=ENGLISH_AMERICAN
SR_LANGUAGE=ENGLISH_AMERICAN
#HTTP_VERSION=1.0
MRCP_ASR=MSS
NETWORK_ANNOUNCEMENT=0
#SKIP_TIME_IN_SECONDS=2
SCRIPT_MAXAGE=0
SCRIPT_MAXSTALE=0
APP_NAME=SIL_Inbound
SCRIPT=http://10.64.101.206/eCI/VXML/eONEMS_Inbound.vxml
```

## 7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of IP Office and eONE.

### 7.1. Verify Avaya IP Office

From the **Avaya IP Office Manager for Server Edition IPO2-IPOSE** screen shown in **Section 5.1**, select **File → Advanced → System Status** to launch the System Status application, and log in using the appropriate credentials.

The **Avaya IP Office System Status – IPO2-IPOSE** screen is displayed. Expand **Trunks** in the left pane and select the SIP line from **Section 5.3**, in this case “3”.

Verify that the **SIP Trunk Summary** screen shows all channels with **Current State** of “Idle”, as shown below.

Avaya IP Office System Status - IPO2-IPOSE (10.64.101.234) - IP Office Linux PC 11.1.0.0.0 build 237

### IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (5)
- Extensions (2)
- Trunks (4)
  - Line: 1
  - Line: 2
  - Line: 3**
  - Line: 8
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

#### SIP Trunk Summary

Line Service State: In Service  
Peer Domain Name: 10.64.101.207  
Resolved Address: 10.64.101.207  
Line Number: 3  
Number of Administered Channels: 4  
Number of Channels in Use: 0  
Administered Compression: G711 Mu  
Enable Faststart: Off  
Silence Suppression: Off  
Media Stream: RTP  
Layer 4 Protocol: UDP  
SIP Trunk Channel Licenses: 256  
SIP Trunk Channel Licenses in Use: 0  
SIP Device Features: REFER (Incoming and Outgoing)

Channel Number	U...	Call Ref	Current State	Time in State	Remote Media A...	Co...	Conne...	Caller ID or Dial...	Other Party on Call	Direction of Call	Round Trip D...	Receive Jitter	Receive Packe...	Transmit Jitter	Transmit Packe...
1			Idle	00:20:17											
2			Idle	00:20:17											
3			Idle	00:20:17											
4			Idle	00:20:17											

Trace Trace All Pause Ping Call Details Graceful Shutdown Force Out of Service Print... Save As...

7:50:14 AM Online

## 7.2. Verify Computer Instruments eONE

This section provides the tests that can be performed to verify the eONE inbound and outbound applications.

### 7.2.1. Inbound Application

Establish an incoming trunk call from PSTN with eONE. Verify that the calling party hears the appropriate greeting associated with the inbound application.

### 7.2.2. Outbound Application

In a browser window, enter the URL associated with the eONE outbound application, in this case **<http://10.64.101.206/MakeACallIPO.html>** where “10.64.101.206” is the eONE server with the web-based interface.

The screen below is displayed. For **PHONE**, enter the phone number associated with an external destination on the PSTN. Retain the default values in the remaining fields and click **CallMeBackNow!**

Verify that an outbound call is initiated from eONE to the external destination. Answer the call at the external destination and verify that the called party hears the appropriate greeting associated with the outbound application.

TENANT:	<input type="text" value="43"/>
CALL TYPE:	<input type="text" value="1000"/>
APPLICATION:	<input type="text" value="1012"/>
PHONE:	<input type="text" value="2126630032"/>
	<input type="button" value="CallMeBackNow!"/>

## 8. Conclusion

These Application Notes describe the configuration steps required for Computer Instruments 7.0 to successfully interoperate with Avaya IP Office Server Edition 11.1.

## 9. Additional References

This section references the product documentation relevant to these Application Notes.

1. *Administering Avaya IP Office™ Platform with Manager*, Release 11.1, Issue 1, April 2020, available at <http://support.avaya.com>.
2. *eONE User's Manual*, Version 7.0, 2020, available at <http://instruments.com>.

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