



Avaya Solution & Interoperability Test Lab

Application Notes for Empix evolve Connect2 Server v3.1 with Avaya IP Office 7.0 – Issue 1.0

Abstract

These Application Notes describe the compliance testing of Empix evolve Connect2 Server with Avaya IP Office. Empix evolve Connect2 Server is a mobility enhancement program which provides the users of local IP Office extensions with the means to use their mobile handsets to make and receive calls via IP Office.

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

The Empix evolve Connect2 Server is a member of the xtension **evolve** program suite. Empix evolve Connect2 Server can be used alone, or together with the other xtension **evolve** components. Empix evolve Connect2 Server is a PC-resident service which provides mobility service to local IP Office extensions. Empix evolve Connect2 Server provides mobile endpoints which are paired with local IP Office extensions with the following functionality:

- Answer incoming calls which were made to the paired local IP Office extension.
- Hold/retrieve.
- “Call through”, allowing mobile endpoints to use existing connections to IP Office to make calls to PSTN endpoints.
- Initiate a second call.
- Switch between calls.
- Transfer a call to a local IP Office endpoint.
- Remotely activate / deactivate Empix evolve Connect2 service.
- Remotely activate / deactivate Do Not Disturb.

2. General Test Approach and Test Results

All tests were performed manually. Only functional testing was performed: no performance testing was done.

2.1. Interoperability Compliance Testing

The following tests were performed as part of the compliance testing.

- Verify the ability to simultaneously signal incoming calls to both the user’s local extension and mobile phone, and to answer such calls from either the local extension or mobile phone.
- Verify the ability to do hold/retrieve from mobile phones.
- Verify the ability of a mobile phone to use the IP Office to make a “call through” to a PSTN endpoint.
- Verify the ability of a mobile phone user to establish a second call and switch between calls.
- Verify the ability of a mobile phone user to transfer a call to another endpoint.
- Verify the ability of a mobile phone to activate/deactivate the Empix evolve Connect2 service for that phone.
- Verify the ability of a mobile phone to activate/deactivate Do Not Disturb (DND) for that phone.
- Verify that calls to mobile phones which have activate DND are routed to the system attendant.
- Verify the ability of Empix evolve Connect2 Server to recover from power failure without manual intervention.
- Verify the ability of Empix evolve Connect2 Server to recover from interruptions to its LAN interface.

2.2. Test Results

All tests produced the expected result.

2.3. Support

Support is available from Empix at:

Empix evolve srl
+39 0733 866 870
support@empixevolve.com
<http://www.empixevolve.com>

3. Reference Configuration

The following **Figure 1** illustrates the configuration which was used for testing.

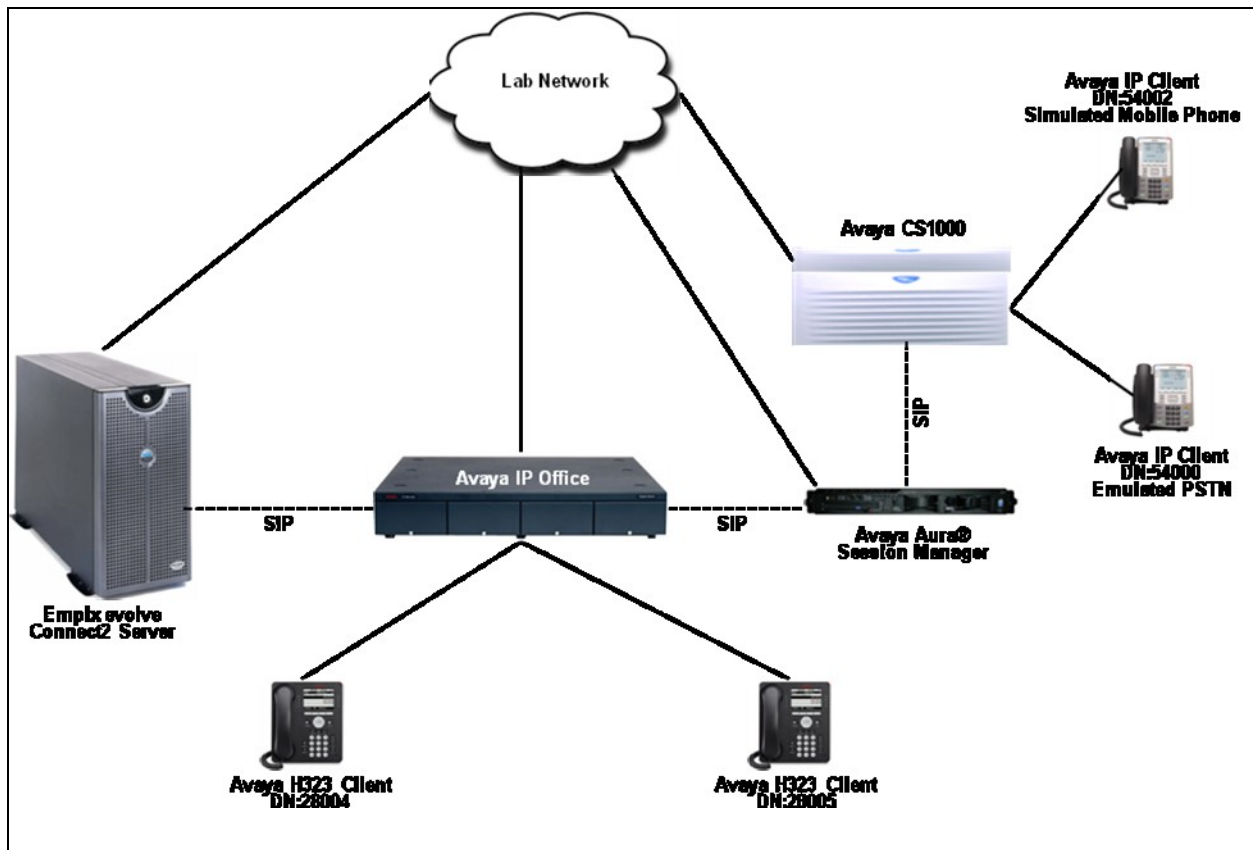


Figure 1: Empix evolve Connect2 Server Lab Configuration

4. Equipment and Software Validated

Software Component	Version
Avaya IP Office	7.0
Avaya CS1000	7.5
Avaya Aura® Session Manager	6.1
Avaya IP Clients	0625C8A
Avaya H323 Clients	6.1(S9608_11HALBR6_1r28_V4r52)
Evolve Server platform OS	Microsoft Windows XP SP3
Empix evolve Connect2 Server	XTENSION EVOLVE 3.1 BUILD 20100802.013

Table 1: Version Numbers of Equipment and Software

5. Configure IP Office

The configuration and verification operations illustrated in this section were performed using the Avaya IP Office Manager program. When this program is started, a tree structure consisting of icons representing the configurable components of the system is displayed as shown in **Figure 2**. When one of these icons is selected, the corresponding system component can be configured.

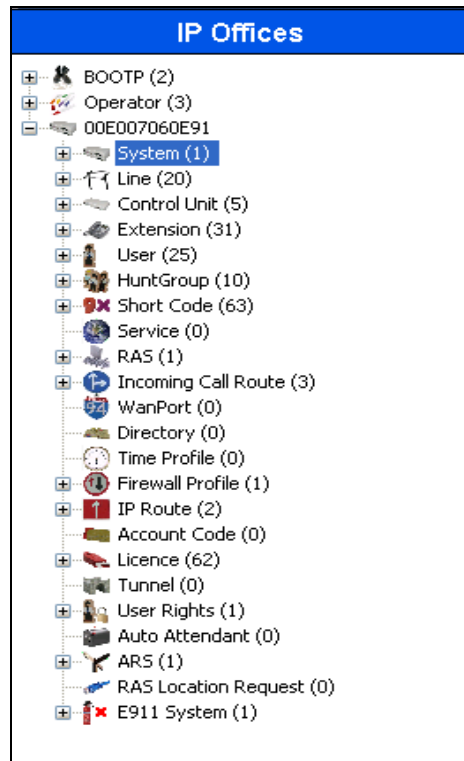


Figure 2: IPO Manager Component Tree

This section explains the configuration of the following components of IP Office that is required for the compliance testing:

- Licensing.
- System Configuration.
- Extension Configuration.
- User Configuration.
- Line Configuration.
- Short Code Configuration.
- Incoming Call Route Configuration.

5.1. Licensing

IP Telephones included in the configuration each consume an **Avaya IP Endpoint** licenses. For complete information on Licensing refer to documents referred in **Section 9[1]**.

Sufficient IP Office **SIP Trunk Channels** license instances are required to cover the maximum number of simultaneous active Empix evolve Connect2 calls which are to be handled by IP Office. Note that each mobile call requires two concurrent SIP channels active.

5.2. System Configuration

Select the **System** icon shown in **Figure 2** and enter the parameters shown in the **Table 1**. Select the tab(s) shown in the “Item” column to configure the parameters for that tab. Refer to **Figures 3 and 4**.

Item	Parameter	Usage
LAN1	IP Address	Specify the IP address to be assigned to IP Office.
LAN Settings	IP Mask	Enter the IP mask to be used by the LAN.
Telephony Telephony	Inhibit Off-Switch Forward / Transfer	Uncheck this box.

Table 1: System Configuration Parameters

The screenshot shows the Avaya IP Office System Configuration interface. The top bar displays the identifier '00E007060E91*'. Below the bar, there are several tabs: System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, and System Events. The 'LAN1' tab is selected and highlighted with a red circle. Below the tabs, there are sub-tabs: LAN Settings, VoIP, Network Topology, and SIP Registrar. The 'LAN Settings' sub-tab is also selected and highlighted with a red circle. The main configuration area shows the following parameters:

- IP Address: 192 . 168 . 150 . 109
- IP Mask: 255 . 255 . 255 . 0
- Primary Trans. IP Address: 0 . 0 . 0 . 0
- RIP Mode: RIP 2 Broadcast (RIP 1 Compatibil) [dropdown arrow]
- ☐ Enable NAT
- Number Of DHCP IP Addresses: 1 [up/down arrows]
- DHCP Mode: ☐ Server ☐ Client ☐ Dialin ☒ Disabled

An 'Advanced' button is located at the bottom right of the configuration area.

Figure 3: System Parameters: LAN1

00E007060E91*

System LAN1 LAN2 DNS Voicemail **Telephony** Directory Services System Events SMTP SMDR Twinning VCM CCR

Telephony Tones & Music Call Log

Analogue Extensions

Default Outside Call Sequence Normal

Default Inside Call Sequence Ring Type 1

Default Ring Back Sequence Ring Type 2

Restrict Analogue Extension Ringer Voltage ☐

Dial Delay Time (secs) 4

Dial Delay Count 0

Default No Answer Time (secs) 15

Hold Timeout (secs) 15

Park Timeout (secs) 180

Ring Delay (secs) 5

Call Priority Promotion Time (secs) Disabled

Default Currency USD

Automatic Codec Preference G.711 ULAW 64K

Companding Law

Switch

☒ ULAW

☐ ALAW

Line

☒ ULAW Line

☐ ALAW Line

☐ DSS Status

☐ Auto Hold

☒ Dial By Name

☒ Show Account Code

☐ Inhibit Off-Switch Forward/Transfer

☐ Restrict Network Interconnect

☐ Drop External Only Impromptu Conference

☐ Visually Differentiate External Call

Figure 4: System Parameters: Telephony

5.3. Extension Configuration

To add a new extension, right-click the **Extension** icon shown in **Figure 2** and select **New H.323 Extension** (not shown). Set the **Base Extension** parameter to the extension to be assigned, and accept the default values for the remaining parameters as shown in **Figure 5** below. Repeat for any number of extensions that will be required.

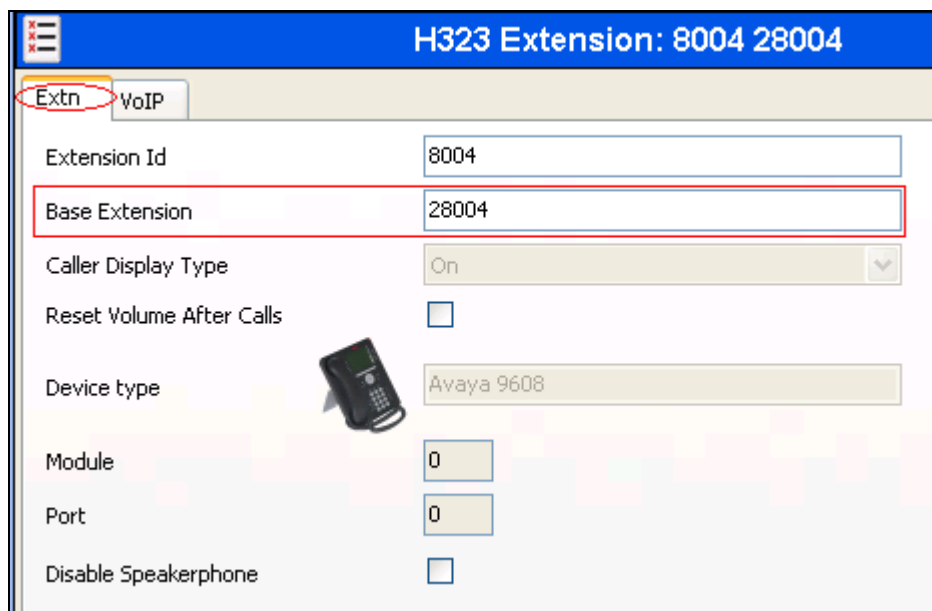


Figure 5: Extension Screen

5.4. User Configuration

Right-click the **User** icon shown in **Figure 2** and select **New** (not shown) to create a user for each of the extension created using the parameters shown in the **Table 2**. Refer to **Figure 6**.

Item	Parameter	Usage
User	Name	Enter a descriptive name to identify the user.
	Password	Enter a password.
	Confirm Password	Confirm the password entered above.
	Full Name	Enter a descriptive name to identify the user.
	Extension	Enter the telephone extension to be assigned to the endpoint (which was allocated in Figure 5).

Table 2: User Configuration Parameters

Extn28004 9608: 28004

User Voicemail DND ShortCodes Source Numbers Telephony Forwarding Dial In Voice Recording

Name Extn28004 9608

Password ****

Confirm Password ****

Full Name Extn 28004 IPO

Extension 28004

Locale

Priority 5

System Phone Rights None

Profile Basic User

Figure 6: User Screen

5.5. Line Configuration

This section contains a description of the IP Office configuration for use with a SIP interface to the PSTN, which was used for conformance testing. Other types of PSTN trunks can be used as well. Right click the **Line** icon shown in **Figure 2**, select **SIP Line** (not shown) to be configured, and enter the parameters shown in **Table 3**. Refer to **Figures 7, 8 and 9**.

Item	Parameter	Usage
SIP Line	Line Number	Assign an available Line from the drop down box.
	ITSP Domain Name	Enter the relevant domain name configured in Session Manager
Transport	ITSP Proxy Address	Enter the IP Address of the Session Manager.
SIP URI	Local URI	Select "Use Internal Data" option from the drop down list.
	Contact	Select "Use Internal Data" option from the drop down list.
	Display Name	Select "Use Internal Data" option from the drop down list.
	Registration	Select "0: <None>" option from the drop down list.
	Incoming Group	Enter the line number assigned.
	Outgoing Group	Enter the line number assigned.

Table 3: SIP Line Configuration Parameters

SIP Line - Line 17

SIP Line
Transport
SIP URI
VoIP
T38 Fax
SIP Credentials

Line Number 17

ITSP Domain Name bwvdev.com

Prefix

National Prefix 0

Country Code

International Prefix 00

Send Caller ID None

Association Method By Source IP address

In Service ☒

Use Tel URI ☐

Check OOS ☒

Call Routing Method Request URI

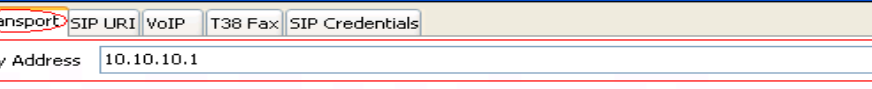
Originator number for forwarded and twinning calls

☒ REFER Support

Incoming Auto

Outgoing Auto

Figure 7: Line Configuration: SIP Line



SIP Line - Line 17*

SIP Line **Transport** SIP URI VoIP T38 Fax SIP Credentials

ITSP Proxy Address 10.10.10.1

Network Configuration

Layer 4 Protocol TCP Send Port 5060

Use Network Topology Info None Listen Port 5060

Explicit DNS Server(s) 0 . 0 . 0 . 0 0 . 0 . 0 . 0

Calls Route via Registrar ☐

Separate Registrar ☐

Figure 8: Line Configuration: Transport

SIP Line - Line 17*

SIP Line | Transport | **SIP URI** | VoIP | T38 Fax | SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	17 17	<...>				N...	0: <Non...	10

[Edit Channel](#)

Via: <None>

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: None

Registration: 0: <None>

Incoming Group: 17

Outgoing Group: 17

Max Calls per Channel: 10

Figure 9: Line Configuration: SIP URI

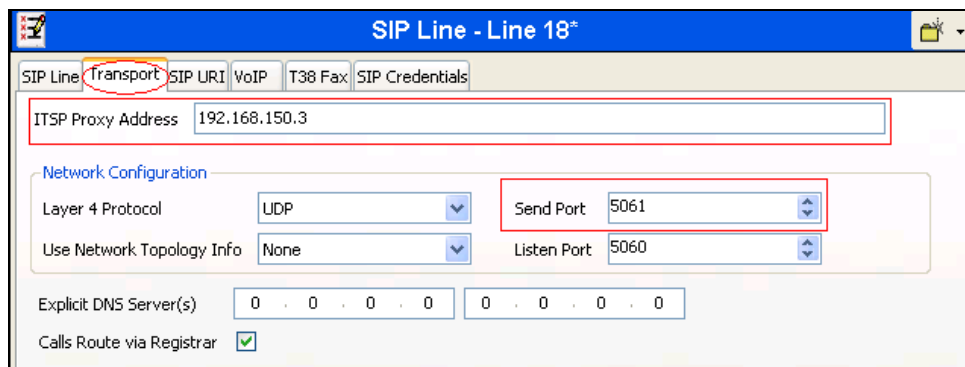
This section also contains a description of the configuration of the SIP trunk between IP Office and Empix evolve Connect2 Server. Right click the **Line** icon shown in **Figure 2**, select **SIP Line** (not shown) to be configured, and enter the parameters shown in **Table 4**. Refer to **Figures 10, 11, 12 and 13**.

Item	Parameter	Usage
SIP Line	Line Number	Assign an available Line from the drop down box.
	ITSP Domain Name	Enter the IP address of the Empix evolve Connect2 Server.
Transport	ITSP Proxy Address	Enter the IP address of the Empix evolve Connect2 Server.
	Send Port	Enter the port number to communicate with the Empix evolve Connect2 Server.
SIP URI	Local URI	Enter “*”.
	Contact	Select “Use Internal Data” from the drop-down menu.
	Display Name	Select “Use Internal Data” from the drop-down menu.
	Registration	Select “None” from the drop-down menu.
	Incoming Group	Enter the line number assigned.
	Outgoing Group	Enter the line number assigned.
VoIP	Compression Mode	Check the “G.711 ALAW” box, and uncheck the others.

Table 4: SIP Line Configuration Parameters

The screenshot shows the 'SIP Line - Line 18*' configuration window. The 'SIP Line' tab is active. The 'Line Number' is set to 18, and the 'ITSP Domain Name' is 192.168.150.3. Other fields include Prefix, National Prefix (0), Country Code, International Prefix (00), Send Caller ID (None), and Association Method (By Source IP address). Checkboxes for 'In Service' and 'Check OOS' are checked. The 'Call Routing Method' is set to 'Request URI'.

Figure 10: SIP Line Configuration Screen



SIP Line - Line 18*

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

ITSP Proxy Address 192.168.150.3

Network Configuration

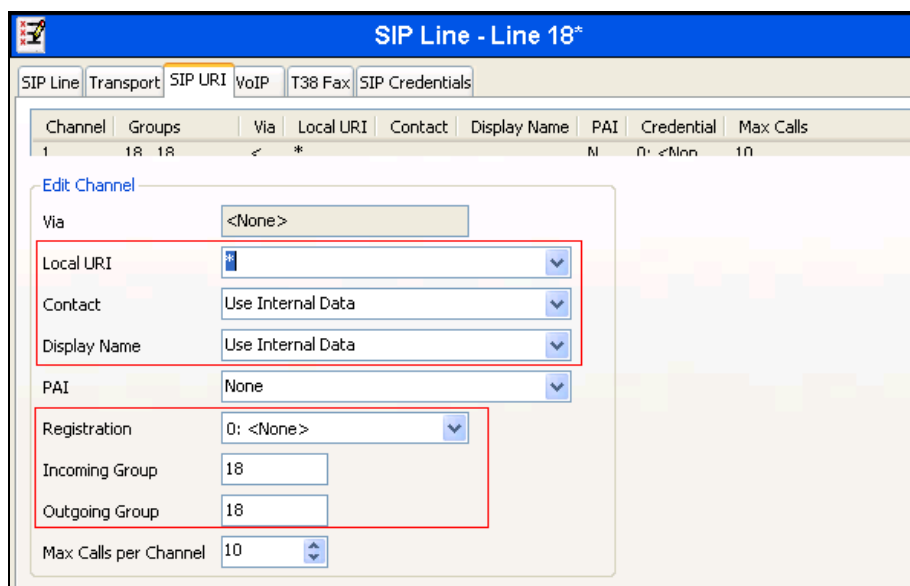
Layer 4 Protocol UDP Send Port 5061

Use Network Topology Info None Listen Port 5060

Explicit DNS Server(s) 0 . 0 . 0 . 0 0 . 0 . 0 . 0

Calls Route via Registrar ☒

Figure 11: Line Configuration: Transport



SIP Line - Line 18*

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	18 18	<	*			M	0: <None>	10

Edit Channel

Via <None>

Local URI

Contact Use Internal Data

Display Name Use Internal Data

PAI None

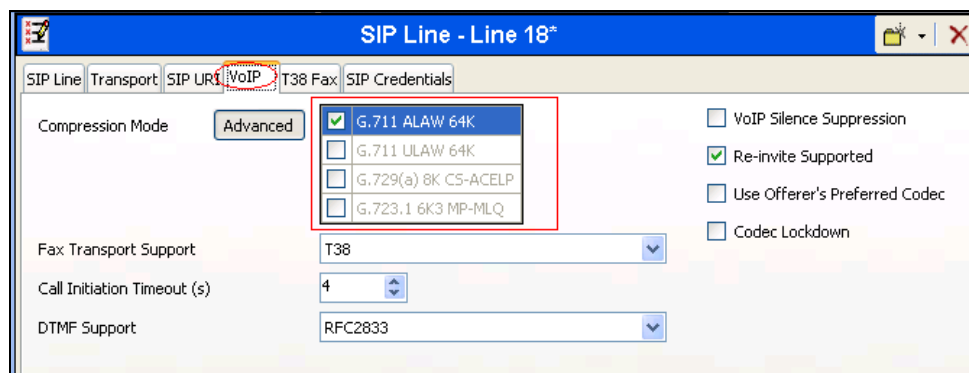
Registration 0: <None>

Incoming Group 18

Outgoing Group 18

Max Calls per Channel 10

Figure 12: SIP: SIP URI Configuration Screen



SIP Line - Line 18*

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

Compression Mode Advanced

☒ G.711 ALAW 64K

☐ G.711 ULAW 64K

☐ G.729(a) 8K CS-ACELP

☐ G.723.1 6K3 MP-MLQ

☐ VoIP Silence Suppression

☒ Re-invite Supported

☐ Use Offerer's Preferred Codec

☐ Codec Lockdown

Fax Transport Support T38

Call Initiation Timeout (s) 4

DTMF Support RFC2833

Figure 13: SIP: VoIP Configuration Screen

5.6. Short Codes

Right-click the **Short Code** icon shown in **Figure 2** and click **New** (not shown) to allocate a short code to provide access to the PSTN via the SIP interface, using parameters as shown in **Table 5**. Refer to **Figure 14**.

Item	Parameter	Usage
Short Code	Code	Enter 54xxx ; Since DN of client on Avaya CS1000 used during compliance testing is 54000.
	Telephone Number	Enter 54N"@10.10.10.1" where 10.10.10.1 is the IP address of Session Manager.
	Line Group ID	Enter the line number which was assigned to the PSTN interface in Figure 7 .

Table 5: Shortcode Configuration Parameters: PSTN

The screenshot shows a configuration window titled "54xxx: Dial*". The "Short Code" tab is active. The following fields are visible:

- Code:** 54xxx
- Feature:** Dial (dropdown menu)
- Telephone Number:** 54N"@10.10.10.1"
- Line Group Id:** 17 (dropdown menu)
- Locale:** (dropdown menu)
- Force Account Code:** ☐

Figure 14: PSTN Access Short Code

Right-click the **Short Code** icon shown in **Figure 2** and click **New** (not shown) to allocate a short code to provide access to the Empix evolve Connect2 Server via the SIP interface, using parameters as shown in **Table 6**. Refer to **Figure 15**.

Item	Parameter	Usage
Short Code	Code	Enter 66N ; Since DN assigned on Empix evolve Connect2 Server used during compliance testing is 66xxx.
	Telephone Number	Enter 66N"@192.168.150.3" where 192.168.150.3 is the IP address of the Empix evolve Connect2 Server.
	Line Group ID	Enter the line number which was assigned to the interface in Figure 10 .

Table 6: Shortcode Configuration Parameters: Empix evolve Connect2 Server

The screenshot shows a web-based configuration form for 'Short Code'. The form has a blue header bar with a small icon on the left. Below the header, the title 'Short Code' is displayed in a light beige bar. The form fields are as follows:

Code	66N;
Feature	Dial
Telephone Number	66N"@192.168.150.3"
Line Group Id	18
Locale	[Blue bar]
Force Account Code	<input type="checkbox"/>

Figure 15: Empix evolve Connect2 Server Access Short Code

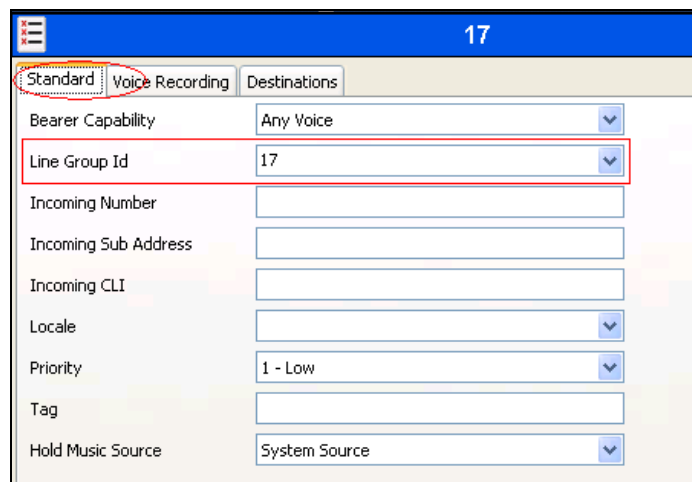
5.7. Incoming Call Route

PSTN Incoming Call Routes

Right-click the **Incoming Call Route** icon shown in **Figure 2** and click **New** (not shown) to create an Incoming Call Route for routing calls from the PSTN to local extensions. Assign parameters to this call route as shown in **Table 7**. Refer to **Figures 16** and **17**.

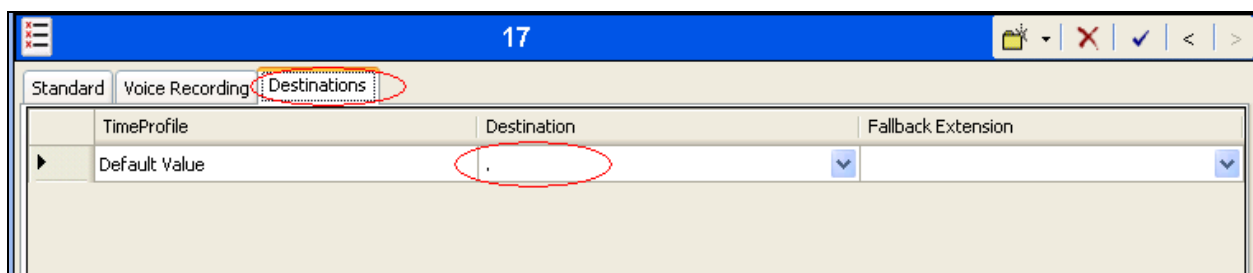
Item	Parameter	Usage
Standard	Line Group Id	Enter the Group Id of the SIP line, as shown in Figure 7 .
Destinations	Destination	Configure the local extension of the called party number by entering a value of “.”.

Table 7: Parameters for Incoming Call Routes



The screenshot shows the 'Standard' tab of the Incoming Call Route configuration. The 'Line Group Id' dropdown is highlighted with a red box and set to '17'. Other fields include Bearer Capability (Any Voice), Incoming Number, Incoming Sub Address, Incoming CLI, Locale, Priority (1 - Low), Tag, and Hold Music Source (System Source).

Figure 16: Incoming Call Route - Standard Parameters



The screenshot shows the 'Destinations' tab of the Incoming Call Route configuration. The 'Destination' dropdown is highlighted with a red box and set to '.'. Other fields include TimeProfile (Default Value) and Fallback Extension.

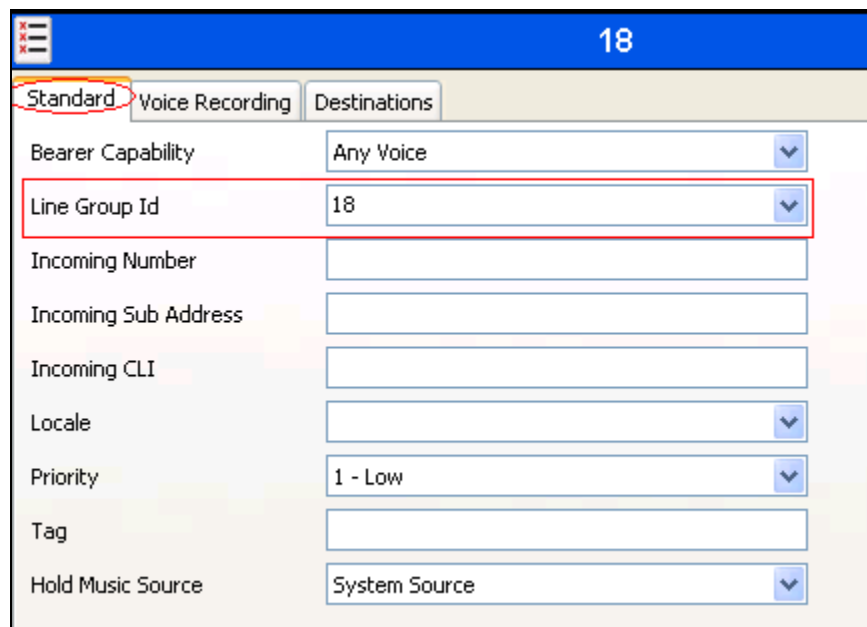
Figure 17: Incoming Call Route - Destinations Screen

SIP Trunk Incoming Call Route

Right-click the **Incoming Call Route** icon shown in **Figure 2** and click **New** (not shown) to create an Incoming Call Route for the SIP trunk to local extensions. Assign parameters to this call route as shown in **Table 8**. Refer to **Figures 18** and **19**.

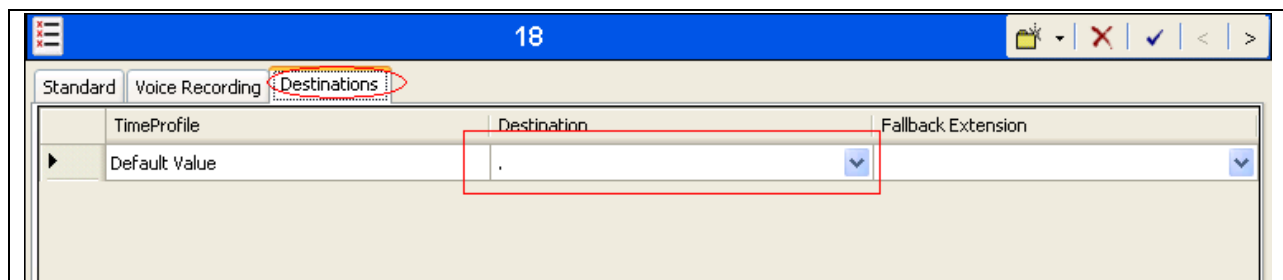
Tab	Parameter	Usage
Standard	Line Group Id	Enter the Group Id of the SIP line, as shown in Figure 10 .
Destinations	Destination	Enter “.”.

Table 8: Parameters for Incoming Call Routes



The screenshot shows the 'Standard' tab of the SIP Incoming Call Route configuration. The 'Line Group Id' field is highlighted with a red box and contains the value '18'. Other fields include Bearer Capability (Any Voice), Incoming Number, Incoming Sub Address, Incoming CLI, Locale, Priority (1 - Low), Tag, and Hold Music Source (System Source).

Figure 18: SIP Incoming Call Route - Standard Parameters



The screenshot shows the 'Destinations' tab of the SIP Incoming Call Route configuration. The 'Destination' field is highlighted with a red box and contains the value '.'. Other fields include TimeProfile (Default Value) and Fallback Extension.

Figure 19: SIP Incoming Call Route - Destinations Screen

6. Configure Empix evolve Connect2 Server

The Empix evolve Connect2 Server installation process also installs MySQL and the Apache server. The installation procedure is beyond the scope of this document. Refer to **Section 9[2]** for further information on complete installation and configurations of the Empix evolve Connect2 Server.

Empix evolve Connect2 Server uses a Web-based tool for configuration, which can be accessed from the server via <http://localhost:20080/login.php>. Log in using the initial login credentials as shown in **Figure 20** below.

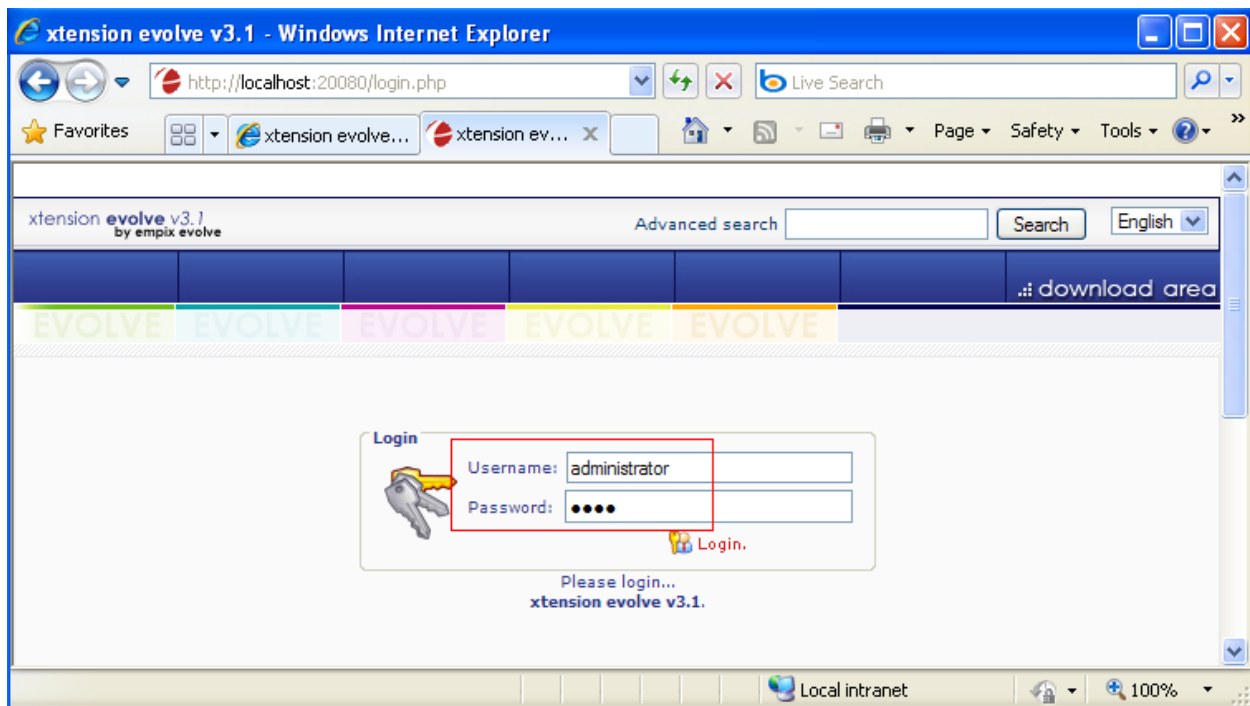


Figure 20: Empix evolve Connect2 Server Login Screen

This section explains the configuration of the following components of Empix evolve Connect2 Server that is required for the compliance testing:

- Installing Updates.
- Installing License.
- Configure Routing Table.
- Configure System Parameters.
- Configure iGate.
- Adding Users.
- Starting Service.

6.1. Install Updates

Navigate to **Administrative tools > Updates**. Click the **Check Updates** button to check for new updates, and the **Download & install** button to install needed updates as shown in **Figure 21** below.

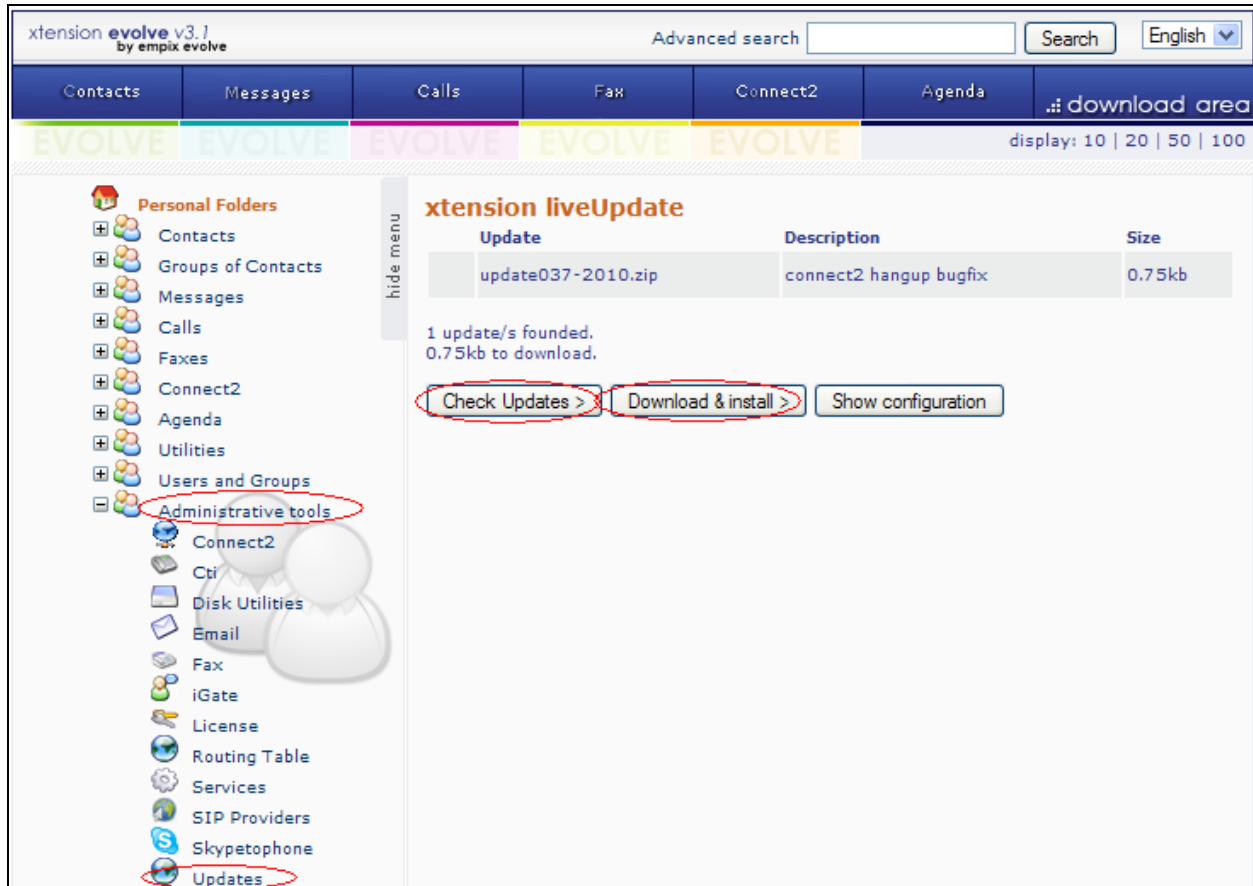


Figure 21: Empix evolve Connect2 Server Update Screen

6.2. Install License

Navigate to **Administrative tools > License** and enter all the appropriate identification information. If a permanent license is available, enter the license code into the **Serial Number** field and click the **Activate** button. If no license is available, a thirty-day trial license can be obtained without charge by clicking the **Request try'n buy** button. Refer to **Figure 22** below.

The screenshot displays the 'xtension evolve v3.1 by empix evolve' web interface. The top navigation bar includes tabs for 'Contacts', 'Messages', 'Calls', 'Fax', 'Connect2', and 'Agenda', along with a 'download area' link. A search bar and language selector are also present. The left sidebar lists 'Personal Folders' and 'Administrative tools', with 'License' selected. The main content area is titled 'License' and features a 'Simple' radio button selected. The form contains the following fields: 'Name' (filled with 'DevConnect'), 'Address', 'Postal Code', 'City', 'Province', 'Country', 'empixevolve Login' (filled with '@avaya.com'), and 'Serial Number'. At the bottom, the 'Activate' and 'Request try'n buy' buttons are circled in red.

Figure 22: License Activation Screen

6.3. Configure Routing Table

Navigate to **Administrative tools > Routing Table** (not shown) and enter the parameters shown in the following table. Refer to **Figure 23** below.

Parameter	Usage
connect2	Enter the leading digit which is used to route connect calls.
faxserver skypetophone SIP gateway Recorder	If any of these services are used, this field should be set appropriately. If the service is not used, a digit should be entered which does not correspond to any numbers in the IP Office dialing plan, as none of these fields can be left blank.
Digits number	Enter the number of digits to be used for Empix evolve Connect2 Server extensions.

Table 9: Routing Table Configuration Parameters

Advanced search Search

extension evolve v3.1

Contacts Messages Calls Fax Connect2 Agenda download area

EVOLVE EVOLVE EVOLVE EVOLVE EVOLVE Journal ☐ display: 10 | 20 | 50 | 100

Personal Folders

- Contacts
- Groups of Contacts
- Messages
- Calls
- Faxes
- Connect2
- Agenda
- Utilities
- Users and Groups
- Administrative tools**
- Connect2
- Cti
- Disk Utilities
- Email
- Fax
- iGate
- License
- Routing Table**
- Services

Routing Table

☒ Simple ☐ Advanced

Calls Handler

Select calls handler

Use GnuGK to handle calls

System prefix

skypetophone 56

connect2 66

faxserver 76

SIP gateway 9

Recorder 8

Digits number: 5

Address PABX:

Figure 23: Xtension evolve Routing Table

6.4. Configure System Parameters

Navigate to **Administrative tools > Connect2** and enter the parameters shown in **Table 10**. Refer to **Figure 24** below.

Parameter	Usage
Country code	Enter the country dialing code preceded by a “+” character.
Operator	Enter the number of the extension to which calls are to be routed if a user is unable to answer.

Table 10: Empix evolve Connect2 Server Configuration Parameters

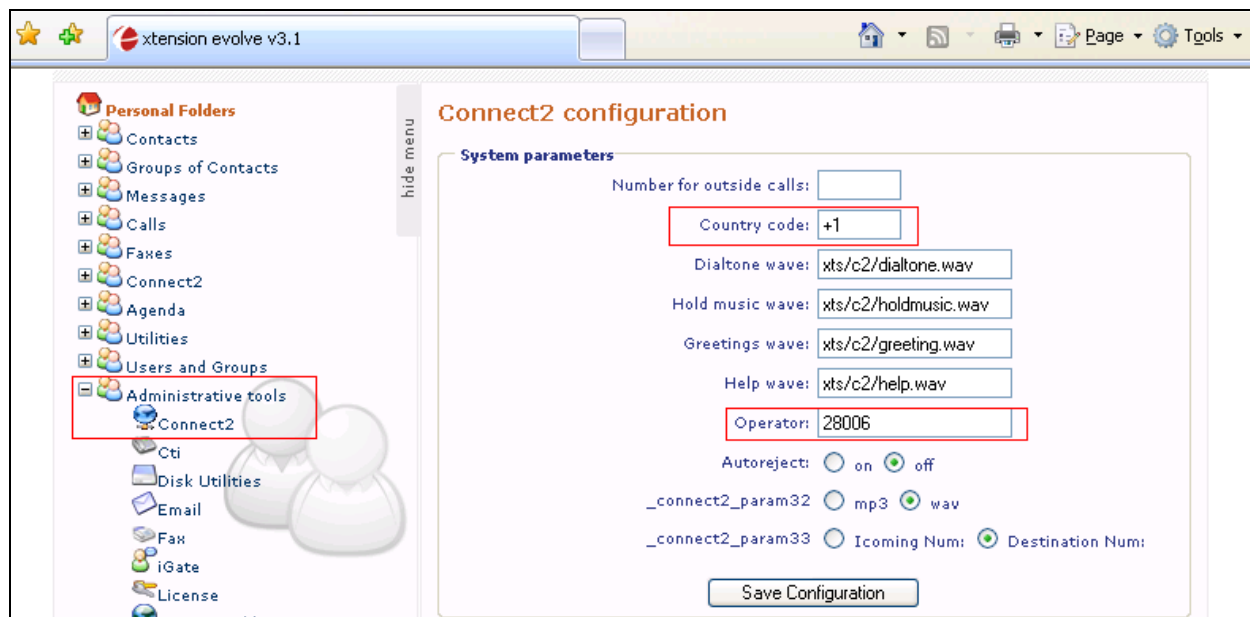


Figure 24: Empix evolve Connect2 Server Configuration Screen

6.5. Configure iGate

Navigate to **Administrative tools > iGate** and enter the parameters as shown in **Table 11**. Refer to **Figure 25**.

Parameter	Usage
Gateway	Enter the IP address of IP Office, and click the SIP radio button.
Interface	Enter “*:” followed by the port number used to receive SIP messages. This should match the Send Port value configured for Figure 11 .

Table 2: iGate Configuration Parameters

The screenshot displays the 'iGate Configuration' window within the 'xtension evolve v3.1' application. The left-hand navigation pane lists various system components, with 'Administrative tools' and 'iGate' highlighted with red circles. The main configuration area is titled 'iGate Configuration' and features two tabs: 'Simple' and 'Advanced'. The 'Gateway' field is populated with the IP address '192.168.150.109', and the 'SIP' radio button is selected. Below this, the 'H.323' section contains fields for 'Local Username' (iGate), 'Interface' (*:1721), 'Tos' (0x010), 'Codecs' (G.711-Alaw-64k), and 'Gatekeeper' (localhost). The 'SIP' section includes fields for 'Local Username' (iGate), 'Interface' (*:5061), 'Tos' (0x010), 'Codecs' (G.711-Alaw-64k), 'Register', 'Register User', and 'Register Password'. The 'Interface' field in the SIP section is circled in red.

Figure 8: iGate Configuration Screen

6.6. Add Users

Perform the procedures described in this section for each of the client endpoints that were created in **Section 5.4**. Navigate to **Users and Groups > New User**, select the **Simple** radio button and enter the parameters shown in **Table 13**. Refer to **Figure 24**.

Parameter	Usage
Username / Password	Enter authorization credentials to be used by the user.
Name	Enter the first name of the user.

Table 13: User Info Parameters

The screenshot shows the 'Insert new user' form in the Evolve v3.1 interface. The 'Simple' radio button is selected and circled in red. The 'Visible as contact' checkbox is checked. The 'Login' section has 'Username: Extn28004' and 'Password:'. The 'Personal Informations' section has 'Name: Extn28004'. The 'Phone numbers' section has empty fields for Telephone, Office fax, and Mobile. The left sidebar shows 'Users and Groups' > 'New User' highlighted with a red box.

Figure 24: User Info Screen

Under the **Phone numbers** section, enter the **Mobile** number of the user. During compliance testing a DN **54002** on the Avaya CS1000 was emulated for a mobile phone. Refer to **Figure 25** below.

The screenshot shows the 'User Phone Numbers Screen' in the Evolve v3.1 interface. The 'Mobile' field is filled with '54002' and highlighted with a red box. The 'Telephone' and 'Office fax' fields are empty. The left sidebar shows 'Users and Groups' > 'New User' highlighted with a red box.

Figure 9: User Phone Numbers Screen

Under the **Cti** section enter the parameters as shown in **Table 14**. Refer to **Figure 26**.

Parameter	Usage
Line ID	Select the IP Office device name for the user from the drop-down menu.
Line numeric id	Enter the user's extension number.

Table 14: User Cti Parameters

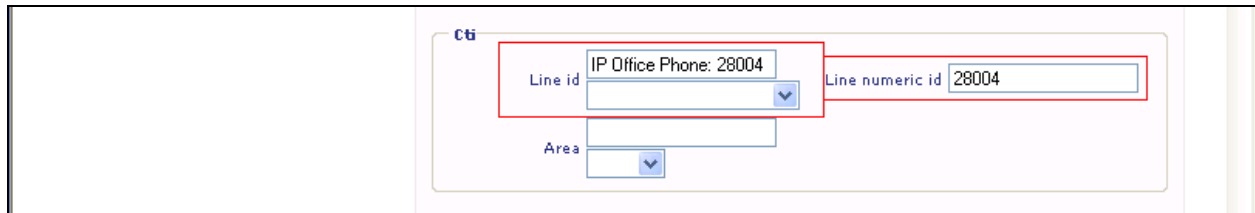
The screenshot shows a web interface for configuring user Cti parameters. A red box highlights the 'Cti' section. Inside this section, there are three fields: 'Line id' with a dropdown menu showing 'IP Office Phone: 28004', 'Line numeric id' with a text input field containing '28004', and 'Area' with a dropdown menu. The 'Line id' and 'Line numeric id' fields are also highlighted by a red box.

Figure 26: User Cti Screen

Note here that if Presence Server application is not installed on the system then the **Line ID** drop down is not provided. User then just needs to populate the **Line numeric id** field. During compliance testing Presence Server application was installed and therefore the **Line id** drop down option is seen as shown in **Figure 26** above.

Under the **Connect2 configuration** section, enter the parameters shown in **Table 16**. Refer to **Figure 27**.

Parameter	Usage
Connect2 Number	Enter the virtual extension to be assigned to the user. This must match the short code created in Figure 15 .
Number or id	Add the user's mobile phone number to the list.
Twin calls	Check this box.
Destination 1	Enter the user's mobile phone number.
Destination 2	Enter the user's local extension.
Extension Number	Enter the user's local extension.

Table 3: Empix evolve Connect2 Server User Parameters

The screenshot displays the 'xtension evolve v3.1' web interface. On the left is a navigation menu with options like Messages, Calls, Faxes, Connect2, Agenda, Utilities, Users and Groups, and Administrative tools. The main content area has tabs for Info, Phone numbers, Addresses, Groups, Cti, Fax, Connect2, and SIP. The 'Connect2' tab is active, showing three configuration sections: 'Connect2 configuration', 'Incoming phone numbers', and 'Outgoing phone numbers - Twin calls'. Each section contains specific input fields and checkboxes, with red boxes highlighting the values specified in the table above.

Figure 27: User Connect2 Configuration Screen

6.7. Start Service

Navigate to **Administrative tools > Services**, check the **connect2** box, and click the **Start xtension evolve service** button as shown in **Figure 28** below.

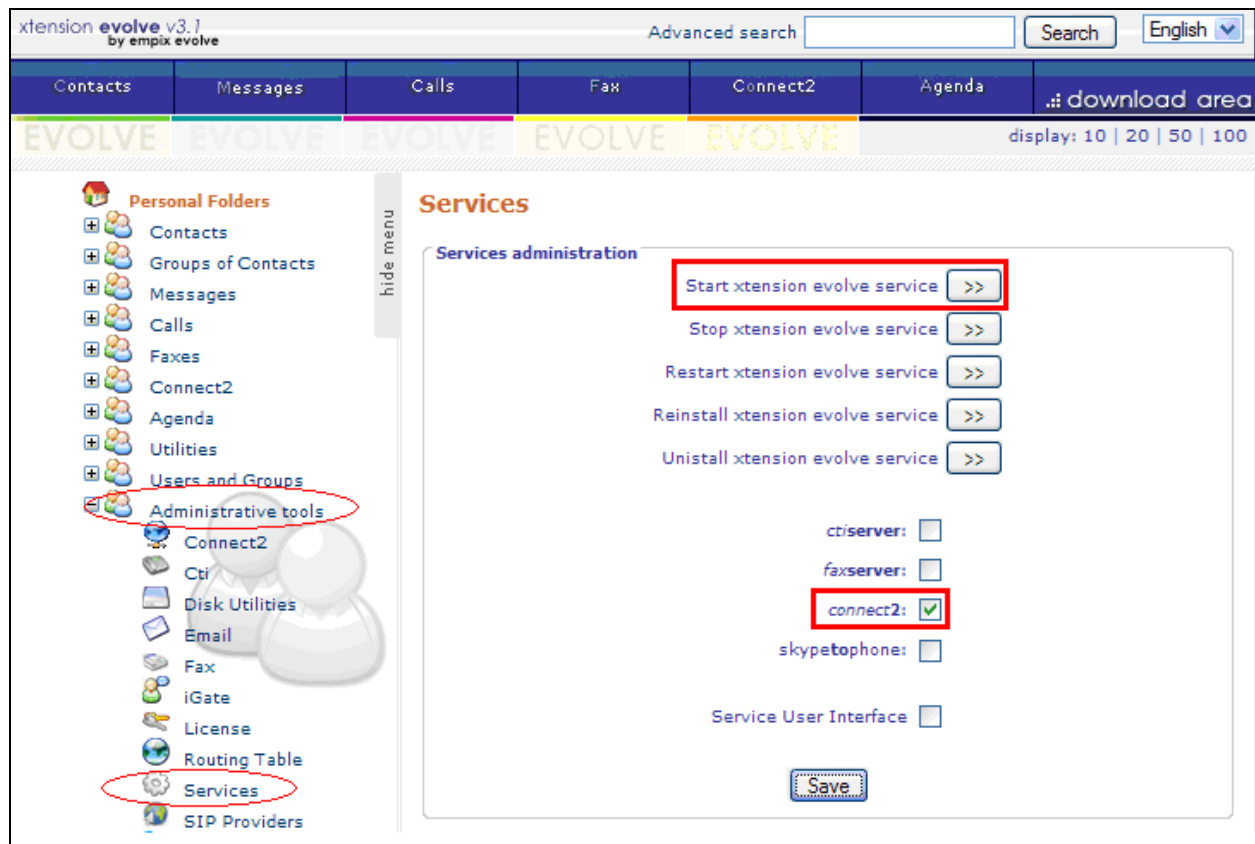


Figure 10: Services Configuration Screen

7. Verification Steps

Use the following steps to verify that evolve Connect2 Server and Avaya IP Office are each configured correctly.

- Click on the “iGate” icon in the Windows shortcut tray in the low right corner of the server display (not shown).
- The xtension iGate status screen is displayed (not shown).
- Make a call from a PSTN endpoint to a local IP Office extension which is “twinned” to cell phone.
- The call progress can be seen on the iGate screen (not shown).

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 2**. The Empix evolve Connect2 Server application is considered compliant with Avaya IP Office Release 7.0.

9. Additional References

[1] Product documentation for Avaya products may be found at:

<https://support.avaya.com/css/Products/>

[2] Product documentation for Empix evolve Connect2 Server may be found at:

<http://www.empixevolve.com/downloads/>

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