



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

---

# **Application Notes for Biscom FAXCOM Server with Avaya IP Office Server Edition R10 – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for Biscom FAXCOM to interoperate with Avaya IP Office Server Edition R10. Biscom FAXCOM is a fax server application that uses the SIP trunk interface of Avaya IP Office to send and receive fax.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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 Biscom FAXCOM (FAXCOM) to interoperate with Avaya IP Office 10 Server Edition. FAXCOM is a fax server application that uses the SIP trunk interface of Avaya IP Office to send and receive fax with Error Correction Mode enabled.

The Avaya IP Office Server Edition configuration consisted of two Avaya IP Office systems, a primary Linux server at the Main site and an expansion IP500V2 at the Remote site that were connected via Small Community Network (SCN) trunks.

For testing, FAXCOM Server application was used to verify outgoing fax can be sent. Incoming fax is routed by Avaya IP Office to Biscom FAXCOM via an available SIP channel. The received incoming fax is stored in a folder on FAXCOM server.

## 2. General Test Approach and Test Results

The feature test cases were performed manually. Fax calls to and from FAXCOM were made. The faxes were sent using the FAXCOM test fax application and an analog fax machine at the PSTN.

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

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.

### 2.1. Interoperability Compliance Testing

The compliance testing included feature and serviceability areas.

The feature testing focused on verifying the following on FAXCOM:

- Proper handling of faxes via SIP trunk (UDP) with T.38: send/receive, internal fax, external fax over SIP trunk, simultaneous bi-directional faxes, and miscellaneous failure scenarios.
- Proper handling of faxes with different pages and data rates.
- No adverse impact on the internal and external VoIP calls during fax transmission.

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

## 2.2. Test Results

All test cases were executed and verified as successful. However, the following observation is noted:

- SIP trunk from IP Office to FAXCOM must have direct media disabled, FAXCOM does not support shuffling.
- FAXCOM only supports MULAW (G.711u). If a different codec is used FAXCOM will reject that fax call.
- Outbound PSTN calls that require an account code were not tested with this release of FAXCOM.

## 2.3. Support

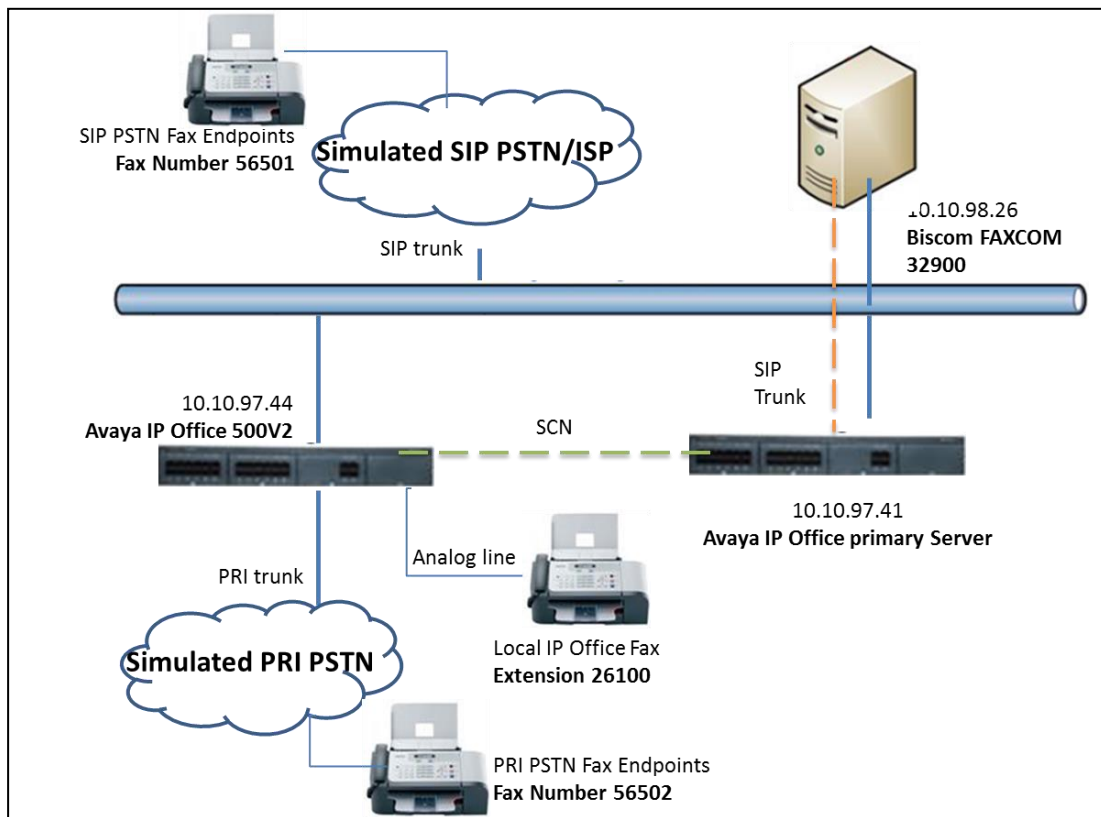
Technical support on FAXCOM can be obtained through the following:

- **Phone:** 1 (800) 477-2472
- **Email:** support@biscom.com
- **Web:** <http://www.biscom.com>

### 3. Reference Configuration

The configuration used for the compliance testing is shown below. FAXCOM is connected to IP Office Server Edition via SIP Trunk. IP Office connected to simulated SIP PSTN and PRI trunks. Numbers in range of 32xxx were used to route faxes to FAXCOM. The FAXCOM server sent and received fax calls to/from a fax machine in the IP Office network as well as from PSTN.

All incoming calls to the fax numbers were routed by IP Office over the SIP trunks to FAXCOM, and all outgoing faxes were routed by FAXCOM over the SIP trunks to IP Office.



**Figure 1: Compliance Testing Configuration**

## 4. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office primary Linux server	10.0.0.0.550
Avaya IP Office 500 V2 Expansion	10.0.0.0.550
Biscom FAXCOM on Microsoft Windows Server 2008 R2 Enterprise SP1 64-bit	6.5.5.13 Dialogic Edition

## 5. Configure Avaya IP Office

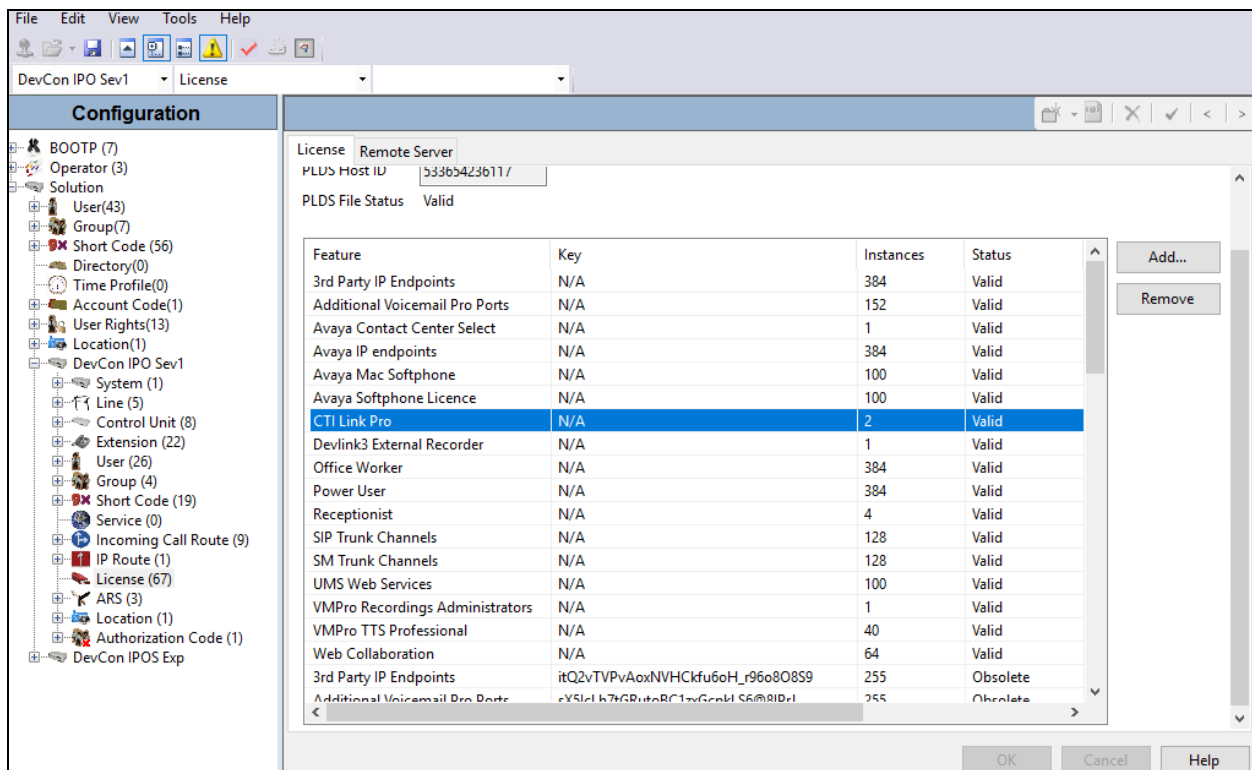
This section provides the procedures for configuring IP Office, assuming it has been installed and licensed. The procedures include the following areas:

- Verify IP Office license
- Obtain LAN IP address
- Enable SIP trunks
- Administer SIP line
- Administer incoming call route
- Administer short code

### 5.1. Verify IP Office License

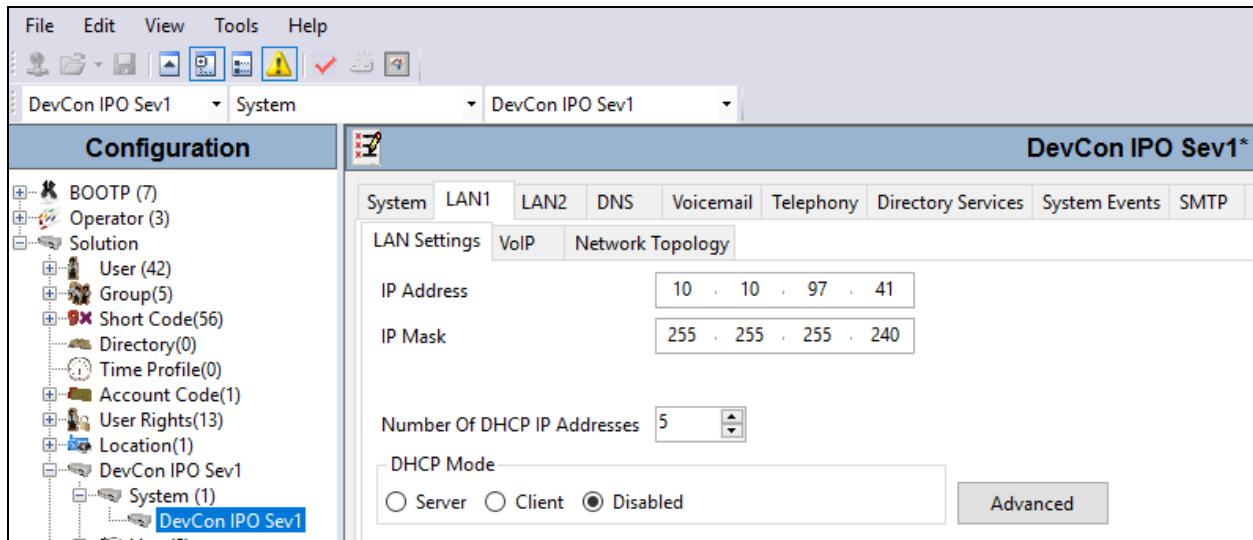
From a PC running the IP Office Manager application, select **Start → All Programs → IP Office → Manager** to launch the Manager application. Select the proper IP Office system, and log in using the appropriate credentials.

The **Avaya IP Office R10 Manager** screen is displayed. From the configuration tree in the left pane, select **License** to display the **License** screen in the right pane. Verify that the **License Status** for **SIP Trunk Channels** is “Valid”, and that the **Instances** value is sufficient for the desired maximum number of simultaneous faxes. If there is insufficient capacity of SIP Trunks, contact an Avaya representative to make the appropriate changes.



## 5.2. Obtain LAN IP Address

From the configuration tree in the left pane, under **Solution** select **Primary Server** → **System** for example, **DevCon IPO Sev1** → **System** tab to display the **DevCon IPO Sev1** screen in the right pane, where **DevCon IPO Sev1** is the name of IP Office Primary Server. 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 FAXCOM in **Section 6.3**. Note that IP Office can support SIP trunks on the LAN1 and/or LAN2 interfaces, and the compliance testing used the LAN1 interface.



### 5.3. Enable SIP Trunks

Continuing from above, select the **VoIP** sub-tab. Make certain that **SIP Trunks Enable** is checked, as shown below.

The screenshot shows the 'DevCon IPO Sev1' configuration window with the 'VoIP' sub-tab selected. The 'SIP Trunks Enable' checkbox is checked. Other visible settings include 'SIP Registrar Enable' checked, 'SIP Domain Name' set to 'bvwdev.com', and 'SIP Registrar FQDN' set to 'bvwdev.com'. The 'Layer 4 Protocol' section shows 'UDP' and 'TCP' checked, with ports 5060 and 5061 configured. The 'Challenge Expiration Time (sec)' is set to 10. The 'RTP' section shows a port number range from 40750 to 50750.

### 5.4. Administer SIP Line from IP Office to FAXCOM Server

From the configuration tree in the left pane, right-click on **Line** and select **New → SIP Line** from the pop-up list to add a new SIP line. Select the **Transport** tab in the right pane. For **ITSP Proxy Address**, enter the IP address of FAXCOM. Set the **Layer 4 Protocol** field to “UDP”. Retain the default values for the remaining fields.

The screenshot shows the 'SIP Line - Line 11\*' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '10.10.98.26'. The 'Network Configuration' section shows 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to 5060, and 'Listen Port' set to 5060. The 'Use Network Topology Info' is set to 'None'. The 'Explicit DNS Server(s)' are set to 0.0.0.0. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.



Select the **SIP URI** tab, and click **Add** (not shown) to display the **New Channel** section. Enter the wildcard character “\*” for **Local URI**, **Contact**, and **Display Name**. Enter an unused group number such as “11” for **Incoming Group** and **Outgoing Group**. Set **Max Calls per Channel** to the maximum number of simultaneous faxes allowed by the FAXCOM license, in this case default value 10 is used. Retain the default values in the remaining fields. Click **OK** (not shown).

The screenshot displays the 'SIP Line - Line 11\*' configuration window. On the left is a 'Configuration' tree with categories like User Rights, Location, DevCon IPO, System, Line, Control Unit, Extension, User, Group, and Short Code. The main panel shows the 'SIP URI' tab selected. The configuration fields are as follows:

Field	Value
Local URI	*
Contact	*
Display Name	*
Identity	None
Header	P Asserted ID
Forwarding And Twinning	
Originator Number	
Send Caller ID	Diversion Header
Diversion Header	None
Registration	0: <None>
Incoming Group	11
Outgoing Group	11
Max Sessions	10

Select the **VoIP** tab. Select only codec “G.711 ULAW 64K” as configured on FAXCOM. Check **Re-invite Supported**. Make sure **Allow Direct Media Path** is unchecked. For **Fax Transport Support**, select “T38” from the drop-down list. Retain the default values in the remaining fields.

The screenshot shows the 'SIP Line - Line 11\*' configuration window with the 'VoIP' tab selected. The window has a title bar with standard icons and a tabbed interface with 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'SIP Credentials', 'SIP Advanced', and 'Engineering'. The 'VoIP' tab is active, displaying codec selection options and various support checkboxes. The 'Codec Selection' section features a 'Custom' dropdown, an 'Unused' list with 'G.711 ALAW 64K', 'G.722 64K', and 'G.729(a) 8K CS-ACELP', and a 'Selected' list with 'G.711 ULAW 64K'. Navigation buttons (right arrow, up arrow, left arrow, down arrow, and right arrow) are between the lists. On the right, checkboxes for 'Local Hold Music' (checked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (unchecked), 'Force direct media with phones' (unchecked), and 'PRACK/100rel Supported' (unchecked) are shown. Below these are dropdowns for 'Fax Transport Support' (set to 'T38'), 'DTMF Support' (set to 'RFC2833/RFC4733'), and 'Media Security' (set to 'Disabled'). At the bottom right are 'OK', 'Cancel', and 'Help' buttons.

Tab	Value
SIP Line	SIP Line
Transport	Transport
SIP URI	SIP URI
VoIP	VoIP
SIP Credentials	SIP Credentials
SIP Advanced	SIP Advanced
Engineering	Engineering

Codec Selection: Custom

Unused:

- G.711 ALAW 64K
- G.722 64K
- G.729(a) 8K CS-ACELP

Selected:

- G.711 ULAW 64K

Local Hold Music: ☒

Re-invite Supported: ☒

Codec Lockdown: ☐

Allow Direct Media Path: ☐

Force direct media with phones: ☐

PRACK/100rel Supported: ☐

Fax Transport Support: T38

DTMF Support: RFC2833/RFC4733

Media Security: Disabled

OK Cancel Help

## 5.5. Small Network Community

A Small Network Community (SCN) trunk was pre-configured on the IP Office Server Edition for connectivity between IP Office Linux Primary to IP500V2 Expansion. Since it is an integral part of the test configuration, a screenshot is included in this section for informational purposes. Below is screen shot of SCN on Linux primary, in this case it is **Line 1**.

The screenshot displays the 'IP Office Line - Line 1\*' configuration window. The 'Line' tab is selected, showing various settings for Line 1. The 'Line Number' is set to 1. The 'Transport Type' is 'WebSocket Server'. The 'Networking Level' is 'SCN'. The 'Security' is 'Medium'. The 'Telephone Number' is empty. The 'Prefix' is empty. The 'Outgoing Group ID' is 99001. The 'Number of Channels' is 250. The 'Outgoing Channels' is 250. The 'Gateway' section shows the 'Address' as 10.10.97.44, 'Location' as 2: Belleville Primary, 'Password' as a masked field, and 'Confirm Password' as a masked field. The 'SCN Resiliency Options' section includes checkboxes for 'Supports Resiliency' (checked), 'Backs up my IP phones' (checked), 'Backs up my hunt groups' (unchecked), and 'Backs up my IP DECT phones' (unchecked). The 'Description' field is empty. The window has 'OK', 'Cancel', and 'Help' buttons at the bottom right.

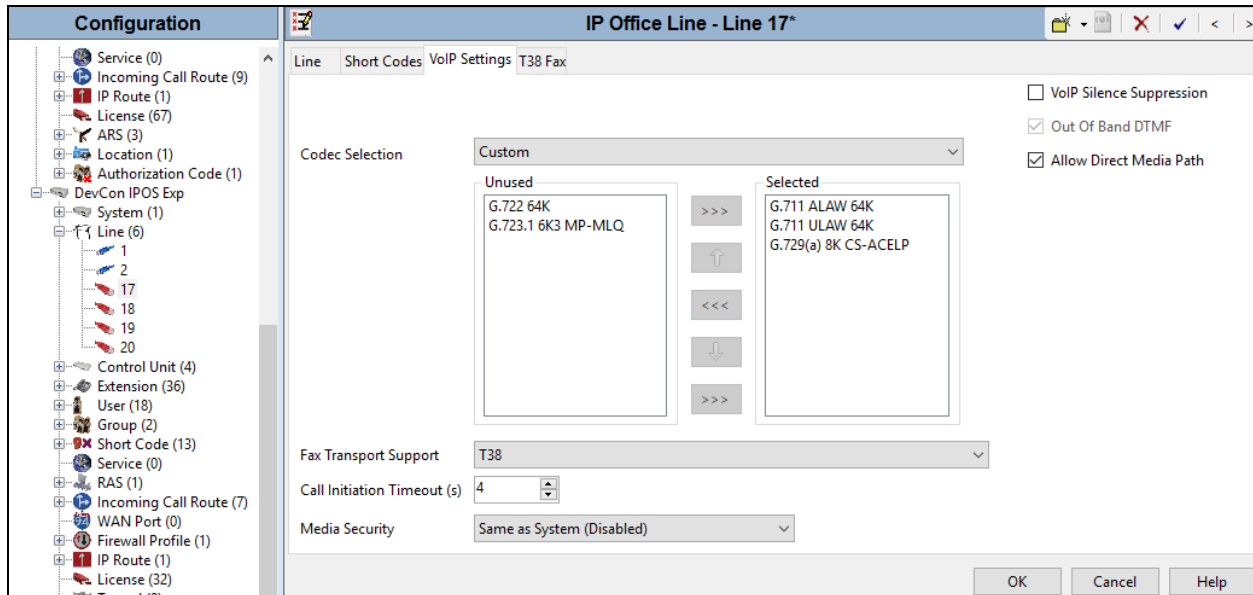
Click on **VoIP Settings** tab to configure **Codec Selection**. Below are the codes used during compliance test for SCN on IP Office Server Edition for **Line 1**.

The screenshot shows the 'IP Office Line - Line 1\*' configuration window with the 'VoIP Settings' tab selected. The 'Codec Selection' dropdown is set to 'Custom'. Below it, there are two lists: 'Unused' containing 'G.722 64K' and 'Selected' containing 'G.711 ALAW 64K', 'G.711 ULAW 64K', and 'G.729(a) 8K CS-ACELP'. Navigation buttons (right arrow, up arrow, left arrow, down arrow, and double right arrow) are between the lists. Other settings include 'Out Of Band DTMF' (checked), 'Allow Direct Media Path' (checked), 'Fax Transport Support' set to 'T38', 'Call Initiation Timeout (s)' set to '4', and 'Media Security' set to 'Same as System (Disabled)'. At the bottom are 'OK', 'Cancel', and 'Help' buttons.

Select the line configured for IP Office Expansion system, in this case, **Line 17**.

The screenshot shows the 'IP Office Line - Line 17\*' configuration window with the 'T38 Fax' tab selected. On the left is a 'Configuration' tree showing a hierarchy of system components, with 'Line (6)' expanded to show 'Line 17'. The main configuration area includes: 'Line Number' (17), 'Transport Type' (WebSocket Client), 'Networking Level' (SCN), 'Security' (Medium), 'Telephone Number' (empty), 'Prefix' (empty), 'Outgoing Group ID' (99999), 'Number of Channels' (250), and 'Outgoing Channels' (250). The 'Gateway' section includes 'Address' (10.10.97.41), 'Port' (443), 'Location' (2: Belleville Primary), 'Password' (masked), and 'Confirm Password' (masked). The 'SCN Resiliency Options' section has 'Supports Resiliency' checked, with sub-options 'Backs up my IP phones' (checked), 'Backs up my hunt groups' (unchecked), and 'Backs up my IP DECT phones' (unchecked). At the bottom are 'OK', 'Cancel', and 'Help' buttons.

Click on **VoIP Settings** tab to configure Codec Selection. Below are the codecs used during compliance test for IP Office Expansion System.



## 5.6. Simulated PSTN/ISP SIP Trunk Configuration

A SIP trunk was pre-configured on the IP Office for connectivity to the simulated PSTN/ISP. Since it is an integral part of the test configuration, a screenshot is included in this section for informational purposes.

On Primary IP Office, **Line 12** was used to connect IP Office to simulated PSTN/ISP.

The screenshot shows the 'SIP Line - Line 12' configuration window. The window has a title bar with standard icons and a tabbed interface with the following tabs: SIP Line, Transport, SIP URI, VoIP, SIP Credentials, SIP Advanced, and Engineering. The 'SIP Line' tab is active. The configuration is organized into two main columns. The left column contains fields for: Line Number (12), ITSP Domain Name (bvwdev.com), Local Domain Name (empty), URI Type (SIP), Location (Cloud), Prefix (empty), National Prefix (0), International Prefix (00), Country Code (empty), Name Priority (System Default), and Description (empty). The right column contains: In Service (checked), Check OOS (checked), Session Timers (Refresh Method: Auto, Timer (sec): On Demand), and Redirect and Transfer (Incoming Supervised REFER: Always, Outgoing Supervised REFER: Always, Send 302 Moved Temporarily: unchecked, Outgoing Blind REFER: checked). At the bottom right are buttons for OK, Cancel, and Help.

Field	Value
Line Number	12
ITSP Domain Name	bvwdev.com
Local Domain Name	
URI Type	SIP
Location	Cloud
Prefix	
National Prefix	0
International Prefix	00
Country Code	
Name Priority	System Default
Description	
In Service	<input checked="" type="checkbox"/>
Check OOS	<input checked="" type="checkbox"/>
Session Timers - Refresh Method	Auto
Session Timers - Timer (sec)	On Demand
Redirect and Transfer - Incoming Supervised REFER	Always
Redirect and Transfer - Outgoing Supervised REFER	Always
Redirect and Transfer - Send 302 Moved Temporarily	<input type="checkbox"/>
Redirect and Transfer - Outgoing Blind REFER	<input checked="" type="checkbox"/>

Under the **Transport** tab, for **ITSP Proxy Address** the IP address of the simulated PSTN/ISP was configured.

**SIP Line - Line 12\***

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

ITSP Proxy Address 10.10.97.228

Network Configuration

Layer 4 Protocol UDP 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

In the **SIP URI** tab, SIP URI record created as shown below:

**SIP Line - Line 12\***

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header
1	12 12	Auto	Auto	Auto	None	PAI		None	None
2	12 13	Auto	Auto	Auto	None	PAI		None	None

Add... Remove Edit...

OK Cancel Help

In **VoIP** tab, default values were used as shown below:

The screenshot shows the 'SIP Line - Line 12\*' configuration window with the 'VoIP' tab selected. The window has a title bar with standard icons and a tabbed interface with the following tabs: SIP Line, Transport, SIP URI, VoIP, SIP Credentials, SIP Advanced, and Engineering. The 'VoIP' tab is active, displaying the following settings:

- Codec Selection:** A dropdown menu set to 'Custom'. Below it are two lists: 'Unused' and 'Selected'.
  - Unused:** Contains 'G.722 64K'.
  - Selected:** Contains 'G.711 ALAW 64K', 'G.711 ULAW 64K', and 'G.729(a) 8K CS-ACELP'.
  - Between the lists are buttons: '>>>', '<<<', and '<<<'.
- Fax Transport Support:** A dropdown menu set to 'T38'.
- DTMF Support:** A dropdown menu set to 'Inband'.
- Media Security:** A dropdown menu set to 'Same as System (Disabled)'.
- Checkboxes on the right:**
  - ☐ Local Hold Music
  - ☒ Re-invite Supported
  - ☐ Codec Lockdown
  - ☒ Allow Direct Media Path
    - ☐ Force direct media with phor
  - ☐ PRACK/100rel Supported

At the bottom right are buttons for 'OK', 'Cancel', and 'Help'.



## 5.7. Simulated PSTN PRI on IP 500V2 Expansion

On the IP Office Expansion system, PRI trunk is used to connect to simulated PSTN, in this case it is **Line 1**.

**Configuration**

- User Rights(13)
- Location(1)
- DevCon IPO Sev1
  - System (1)
    - DevCon IPO Sev1
      - Line (5)
        - 1
        - 2
        - 10
        - 11
        - 12
- Control Unit (8)
- Extension (22)
- User (26)
- Group (3)
- Short Code (18)
- Service (0)
- Incoming Call Route (9)
- IP Route (1)
- License (67)
- ARS (3)
- Location (1)
- Authorization Code (1)
- DevCon IPOS Exp
  - System (1)
    - Line (6)
      - 2
      - 17
      - 18

**PRI 24 (Universal) - Line 1**

Channels

Line Number	01	Line SubType	PRI
Card	1		
Port	9	Admin	In Service
Switch Type	NI2	Provider	Local Telco
Send Service Messages	<input type="checkbox"/>		
Channel Allocation	1 -> 23		
Prefix			
Add 'Not end-to-end ISDN' Information Element	Never		
Progress Replacement	None		
Send Redirecting Number	<input type="checkbox"/>		
Test Number			
Clock Quality	Network	Framing	ESF
CRC Checking	<input checked="" type="checkbox"/>	Zero Suppression	B8ZS
CSU Operation	<input type="checkbox"/>	Line Signaling	CPE

OK Cancel Help

## 5.8. Administer Incoming Call Route

From the configuration tree in the left pane, right-click on **Incoming Call Route**, and select **New** from the pop-up list to add a new route (not shown). For **Line Group Id**, select the incoming group number from **Section 5.4**, in this case “11”. Click **OK**.

**Configuration**

- BOOTP (7)
- Operator (3)
- Solution
  - User (42)
  - Group(5)
  - Short Code(56)
  - Directory(0)
  - Time Profile(0)
  - Account Code(1)
  - User Rights(13)
  - Location(1)
  - DevCon IPO Sev1
    - System (1)
      - Line (5)
        - Control Unit (8)
        - Extension (22)
        - User (26)
        - Group (3)
        - Short Code (18)
        - Service (0)
        - Incoming Call Route (9)
        - IP Route (1)
        - License (67)
        - ARS (3)
        - Location (1)
        - Authorization Code (1)

**Incoming Call Route**

Line Group ID	Incoming Number	Destination
2	.	.
10	.	.
11	.	.
12	19088426999	6219675
12	4048511333	26008 2
12	4048511332	26006 Z
12	4048511331	26002 E
12	190884xxxxx	#
12	.	.

**11**

Standard Voice Recording Destinations

Bearer Capability	Any Voice
Line Group ID	11
Incoming Number	
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	holdmusic
Ring Tone Override	None

Select the **Destinations** tab. For **Destination**, enter “.” to route the call to the dialed number received from FAXCOM without modification.

The screenshot shows a software window titled "11" with a menu icon on the left and a toolbar on the right containing icons for help, save, delete, confirm, and navigation. Below the title bar are three tabs: "Standard", "Voice Recording", and "Destinations". The "Destinations" tab is active and contains a table with three columns: "TimeProfile", "Destination", and "Fallback Extension". The first row of the table has the values "Default Value", ".", and "." respectively, each followed by a dropdown arrow. Below the table is a large empty rectangular area. At the bottom right of the window are three buttons: "OK", "Cancel", and "Help".

TimeProfile	Destination	Fallback Extension
Default Value	.	.

## 5.9. Administer Short Code

From the configuration tree in the left pane, right-click on **Short Code** and select **New** from the pop-up list to add a new short code for fax calls to FAXCOM. In the compliance testing, users on IP Office are designated with fax numbers 320xx, and faxes are routed over the SIP trunks to FAXCOM. Users on IP Office Expansion system used 3320xx to reach FAXCOM on Primary IP Office.

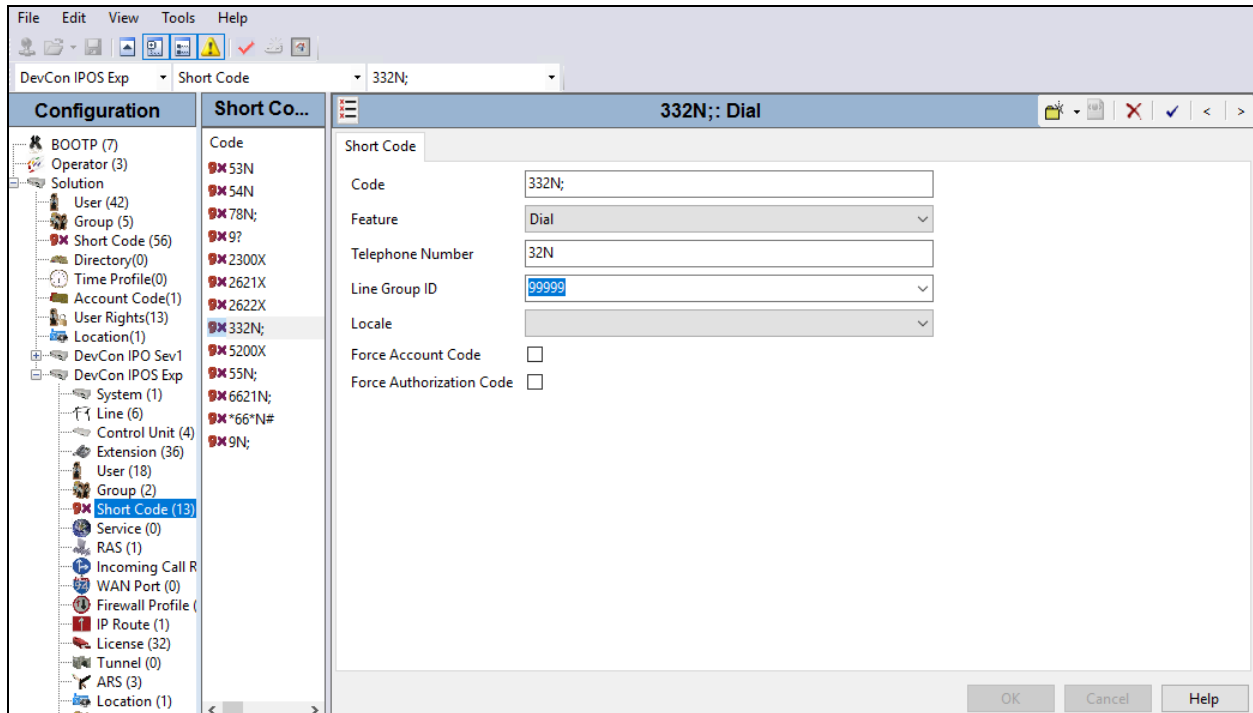
For **Code**, enter “32900”. For **Feature**, select “Dial” from the drop-down list. For **Telephone Number**, enter “32900”@10.10.98.26” where “32900” corresponds to the short code and “10.10.98.26” is the IP address of FAXCOM. For **Line Group ID**, enter the outgoing group number from **Section 5.4**, example 11. Click **OK**.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'Configuration' tree shows a hierarchy starting with 'Location(1)', followed by 'DevCon IPO Sev1', 'System (1)', 'Line (5)', 'Control Unit (8)', 'Extension (22)', 'User (26)', 'Group (4)', and 'Short Code (19)'. The 'Short Code (19)' list includes various codes like \*66\*N#, 2300X, 2621X, 2622X, 26503, 27N;, 32900 (highlighted in blue), 33000, 33001, 5200X, 53N;, 54N;, 55N;, 621N;, 65N;, 721N;, and 78N;. The main window is titled '32900: Dial' and contains the following fields:

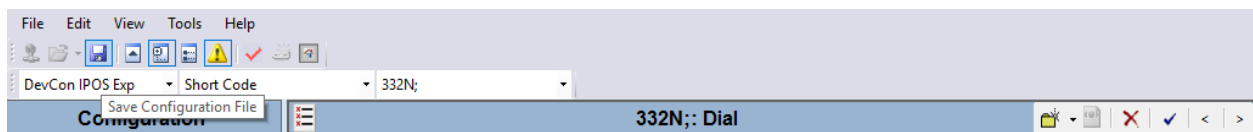
- Short Code**: 32900
- Feature**: Dial (selected from a dropdown)
- Telephone Number**: 32900"@135.10.98.26"
- Line Group ID**: 11 (selected from a dropdown)
- Locale**: (empty dropdown)
- Force Account Code**: ☐
- Force Authorization Code**: ☐

At the bottom right of the window are three buttons: 'OK', 'Cancel', and 'Help'.

Repeat same step to create a Short Code on the IP Office Expansion system to reach FAXCOM, displayed below is the Short Code used during compliance test. For **Code**, enter “332N”. For **Feature**, select “Dial” from the drop-down list. For **Telephone Number**, enter “32N”. For **Line Group ID**, enter the outgoing group ID from **Section 5.5**, example 99999. Click **OK**



Click on **Save** icon to save all changes.



Follow instructions on the screens to save all changes.

## 6. Configure Biscom FAXCOM

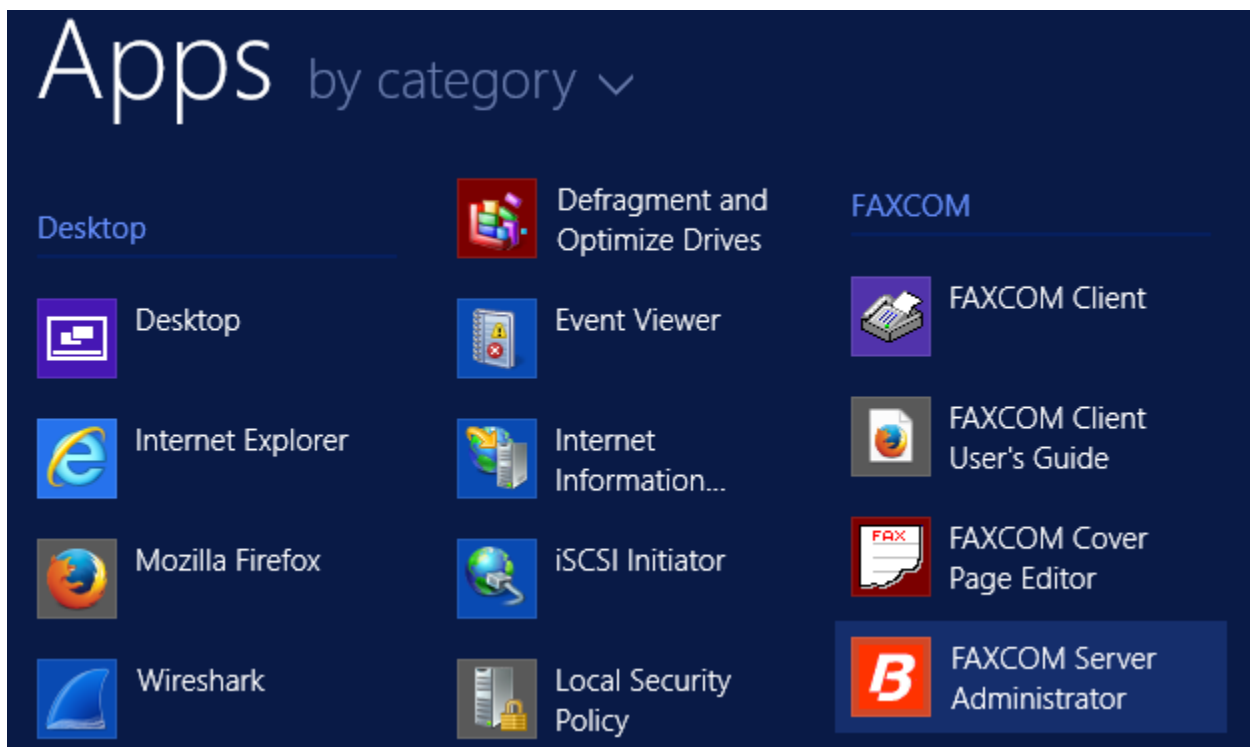
This section provides the procedures for configuring FAXCOM. The procedures include the following areas:

- Launch FAXCOM Server Administrator Program
- Administer Fax Ports
- Administer Server Settings
- Administer incoming routing table
- Restart service

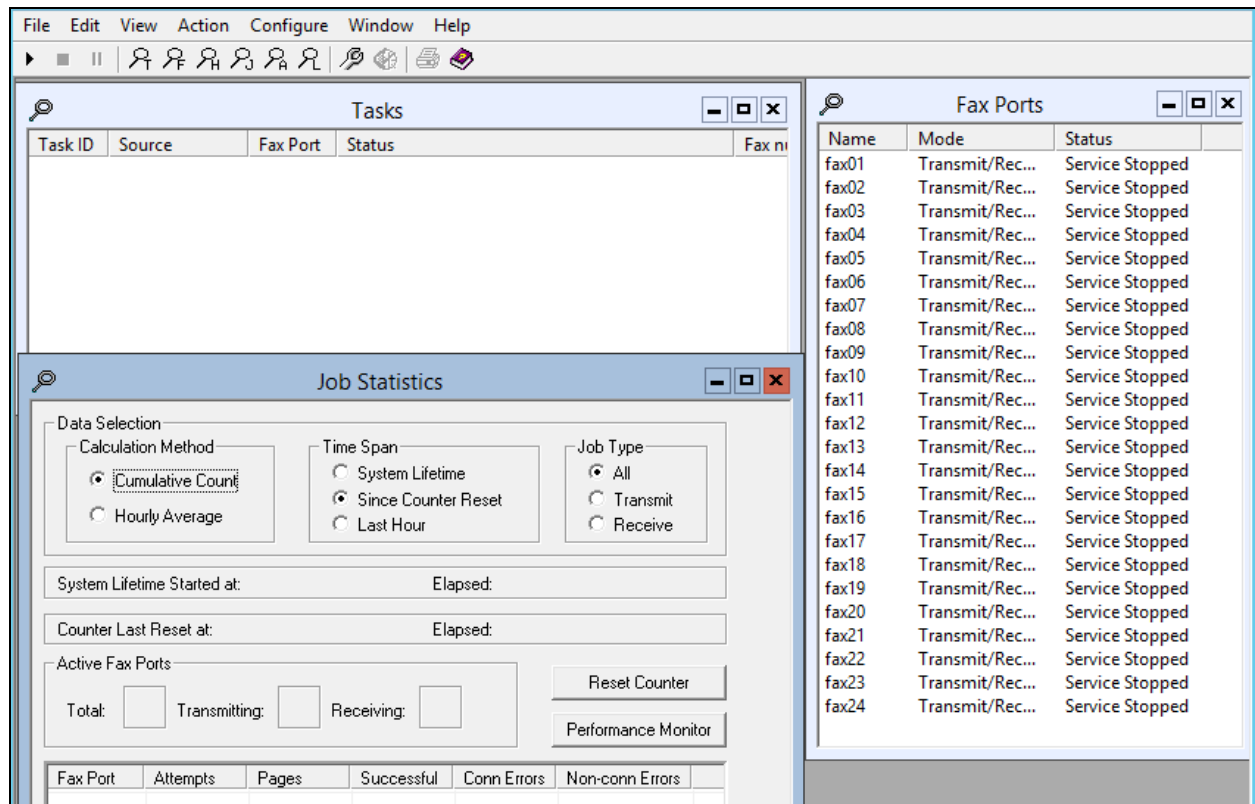
Note that as part of the FAXCOM installation, the IP Office IP address was entered, and a site name and the Basic user profile were created.

### 6.1. Launch FAXCOM Server Administrator Program

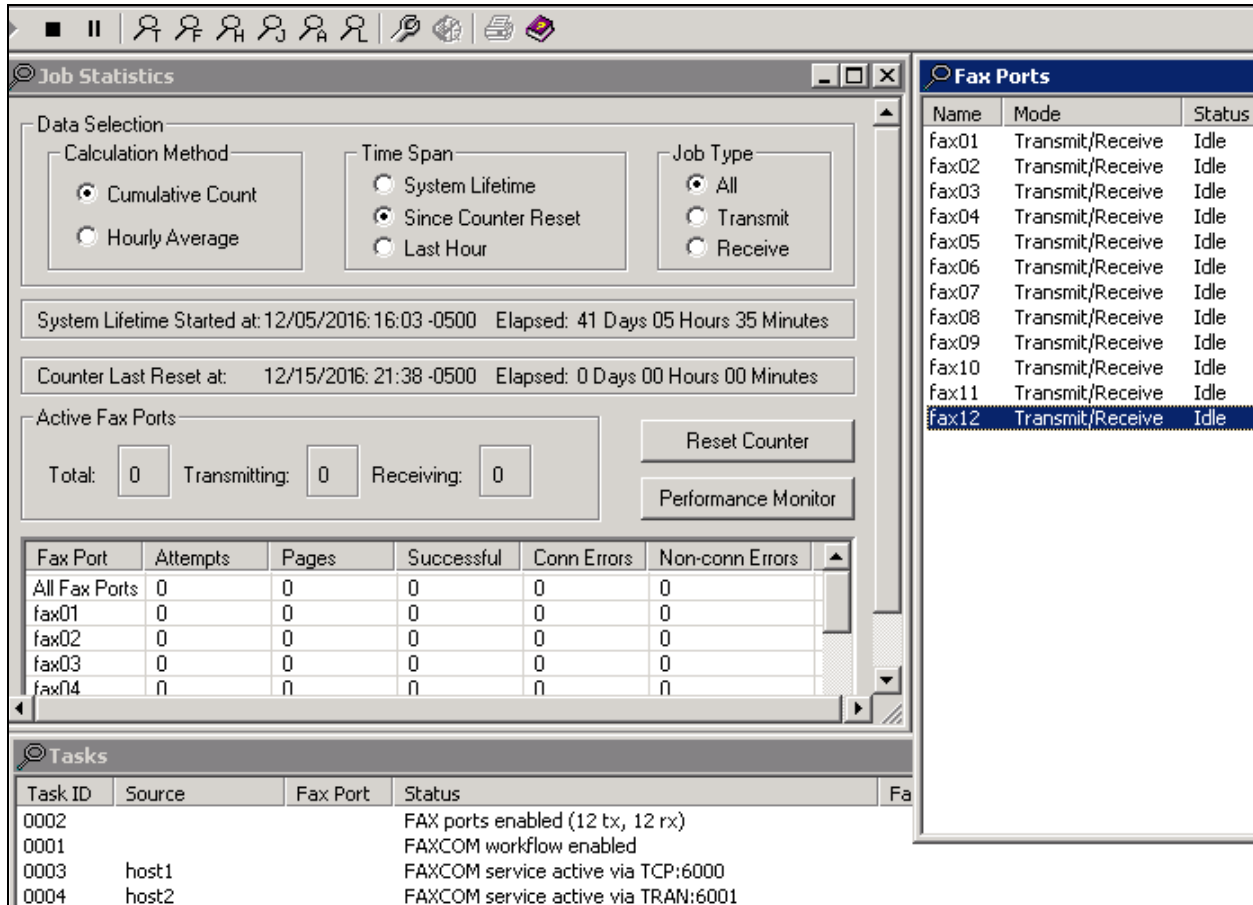
From the FAXCOM server, select **Start → Apps by category → FAXCOM Server Administrator** to launch the application.



**FAXCOM Server Administrator** window opens and a set of three windows are displayed on the FAXCOM desktop –Job Statistics, Tasks, and Fax Ports.



Click on  icon to start fax service. Verify all the ports are “Idle” as shown below:



The screenshot displays the FAXCOM application interface. The main window is titled "Job Statistics" and contains several sections:

- Data Selection:**
  - Calculation Method:** ☒ Cumulative Count, ☐ Hourly Average
  - Time Span:** ☐ System Lifetime, ☒ Since Counter Reset, ☐ Last Hour
  - Job Type:** ☒ All, ☐ Transmit, ☐ Receive
- System Lifetime:** Started at: 12/05/2016: 16:03 -0500 Elapsed: 41 Days 05 Hours 35 Minutes
- Counter Last Reset at:** 12/15/2016: 21:38 -0500 Elapsed: 0 Days 00 Hours 00 Minutes
- Active Fax Ports:**
  - Total: 0 Transmitting: 0 Receiving: 0
  - Buttons: Reset Counter, Performance Monitor
- Fax Port Statistics Table:**

Fax Port	Attempts	Pages	Successful	Conn Errors	Non-conn Errors
All Fax Ports	0	0	0	0	0
fax01	0	0	0	0	0
fax02	0	0	0	0	0
fax03	0	0	0	0	0
fax04	0	0	0	0	0
- Tasks Table:**

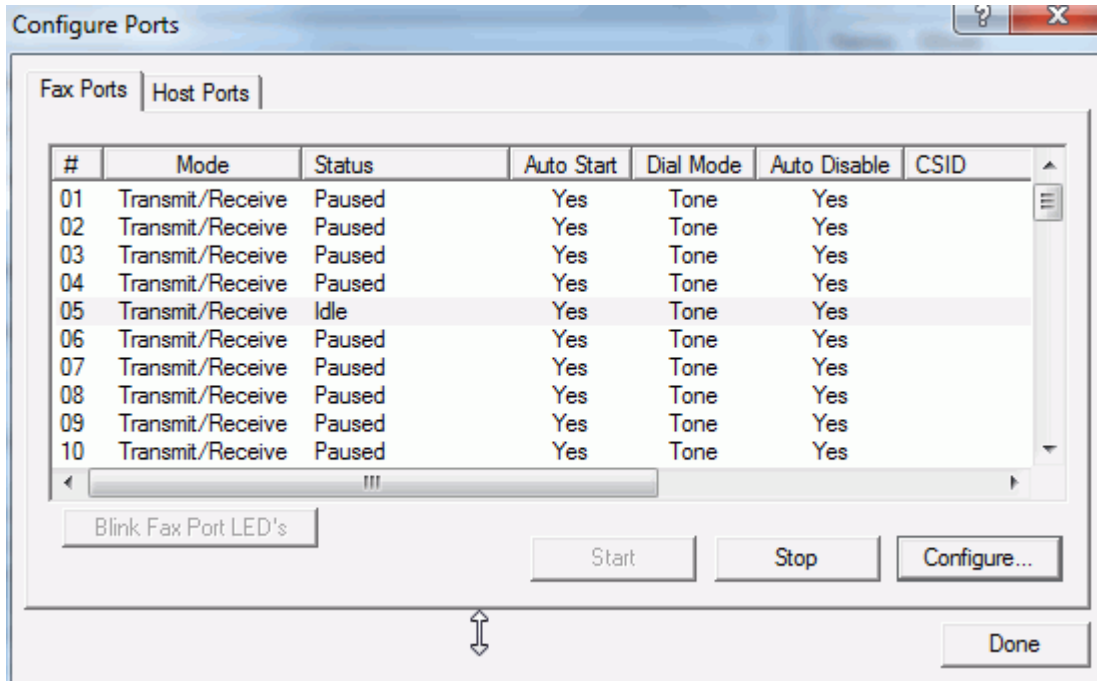
Task ID	Source	Fax Port	Status
0002			FAX ports enabled (12 tx, 12 rx)
0001			FAXCOM workflow enabled
0003	host1		FAXCOM service active via TCP:6000
0004	host2		FAXCOM service active via TRAN:6001

On the right side, there is a separate window titled "Fax Ports" with a table showing the status of 12 fax ports:

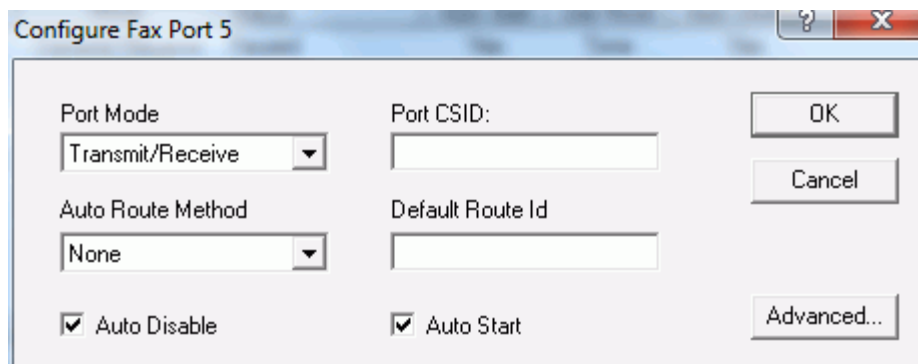
Name	Mode	Status
fax01	Transmit/Receive	Idle
fax02	Transmit/Receive	Idle
fax03	Transmit/Receive	Idle
fax04	Transmit/Receive	Idle
fax05	Transmit/Receive	Idle
fax06	Transmit/Receive	Idle
fax07	Transmit/Receive	Idle
fax08	Transmit/Receive	Idle
fax09	Transmit/Receive	Idle
fax10	Transmit/Receive	Idle
fax11	Transmit/Receive	Idle
fax12	Transmit/Receive	Idle

## 6.2. Administer Fax Ports

Navigate to **Configure → Ports → Fax Ports** (not shown). Configure **Fax Ports** tab shown as below:



To configure the port, highlight the port and click the **Configure** button to display the Fax Port dialog. (If the port is not stopped, you are prompted whether to stop the port since you cannot configure a port unless you first stop it. You can, however, view the configuration in read-only mode without stopping the port.). Specify the appropriate information as follows, clicking **OK** when done. During compliance testing, **Port Mode** “Transmit/Receive” was selected.





### 6.3. Administer Server Setting

From the FAXCOM Server Administrator window, select **Configure → Server Setting**. Select **SR140 Settings**; select “T.38” for **Mode**. In **Call Control** box, select “SIP” and “Avaya” for **Call Control Variant**. **T.38 Version** is “0”. **Local IP address** is FAXCOM IP address; in this case it is “10.10.98.26”. **Gateway IP Address** is the IP Office IP address; in this case it is “10.10.97.41”. Click **Done** to save changes.

The screenshot shows the 'Configure All Settings' window with the 'SR140 Settings' tab selected. The window has a blue title bar with a question mark and a close button. The main area is divided into several sections:

- Navigation Tabs:** Dialing, Local Exchanges, Internal Numbers, LCR Routes, LCR Rules, Translation Server, Data Archive, Alarm Events, Alarm Notifications, Fax Ports, Host Ports, Server Settings (selected), SR140 Settings, Options, Inbound Routes.
- Licensed Channels:** 24
- License Manager:** Button
- Debug logging:** ☒
- V.34 Mode:** ☐
- Mode:** ☒ T.38, ☐ T.38 + G.711, ☐ G.711
- Call Control:** ☐ H.323, ☒ SIP
- Call Control Variant:** SIP UDP only (dropdown)
- Round Robin:** ☒
- T.38 Version:** 0 (dropdown)
- IP Preference:** IPV4 Only (dropdown)
- Local IP Address:** 10.10.98.26 (text field)
- Gateway IP Address:** List containing 10.10.97.41. Buttons: Add, Remove, Move Up, Move Down.
- Done:** Button at the bottom right.

Select **Configure** → **All Settings**, select **Options** tab. Verify maximum number of ports setup. Click **Done** to close window.

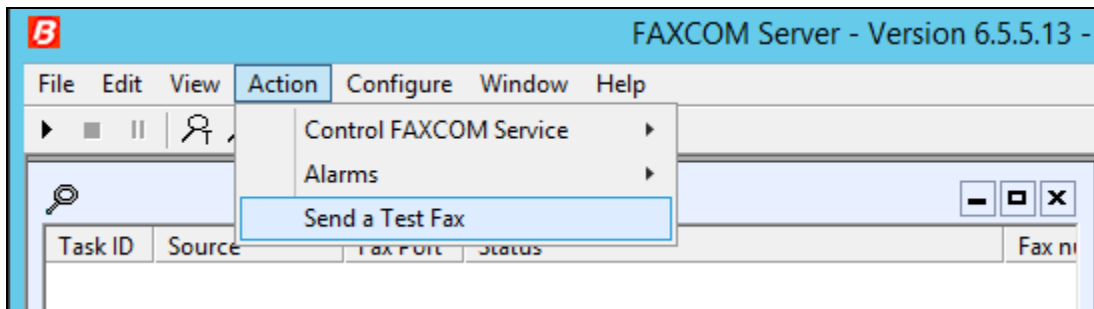
The screenshot shows a software configuration window titled "Configure All Settings". It has a blue header bar with a question mark icon and a red close button (X). Below the header is a tabbed interface with the following tabs: Dialing, Local Exchanges, Internal Numbers, LCR Routes, LCR Rules, Translation Server, Data Archive, Alarm Events, Alarm Notifications, Fax Ports, Host Ports, Server Settings, SR140 Settings, Options (selected), and Inbound Routes. The "Options" tab is active, displaying the following settings:

- Fax Server Model: **FAXCOM Server** (highlighted in blue)
- FAXCOM Server Software Serial Number:
- Maximum Fax Ports:
- ☐ OCR Option

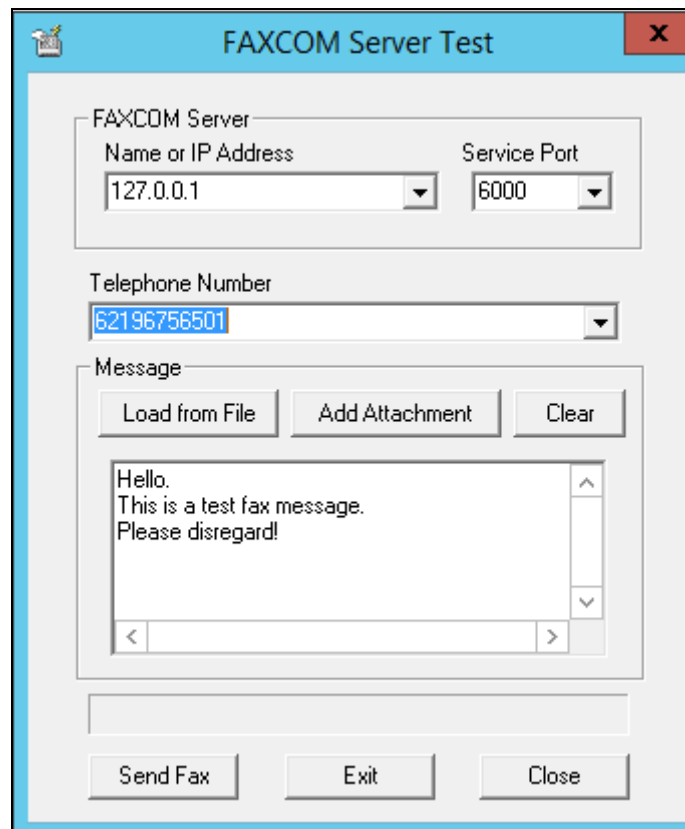
A "Done" button is located at the bottom right of the window.

## 6.4. Send a Test Fax

From the **Action** menu, select **Send a Test Fax**.



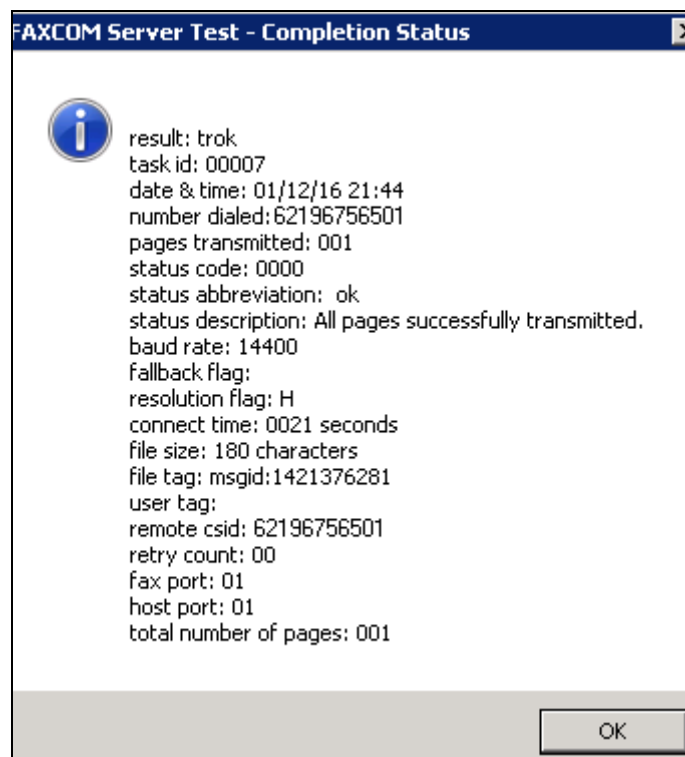
The FAXCOM Server Test dialog box is displayed, with the name or internal IP Address of the FAXCOM Server itself and the default TCP port number prefilled. In the **Telephone Number** box, specify the fax number of a nearby fax machine. Click the **Send Fax** button.



Verify in the log window job is created and sending fax.

0003	host1			FAXCOM service active via TCP:6000	
0002				FAX ports enabled (24 tx, 24 rx)	
0001				FAXCOM workflow enabled	
0046	2012PC2_TESTQUEUE_tx1 0050	fax07		Fax delivery transmitting (3 of 13) to: 62196756501, remote CSID: ...	62196
0047	2012PC2_TESTQUEUE_tx1 0051	fax15		Fax delivery transmitting (2 of 13) to: 26100, remote CSID: Fax	26100

Upon completion, the Completion Status window is displayed. Confirm the **result** is “trok” (for transmit ok). Click **OK**. When returned to the FAXCOM Server Test dialog, close the dialog.



## 7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of IP Office and FAXCOM. Prior to verification, start sending a fax from the PSTN to a fax user on IP Office.

### 7.1. Verify Avaya IP Office

Send a fax from FAXCOM to a simulated PSTN fax machine. Verify the connection status of a trunk on IP Office by navigate to **Avaya IP Office R10 Manager** screen shown in **Section 5.1**, select **File → Advanced → System Status** to launch the System Status application, and log in using the appropriate credentials (not shown).

The **IP Office System Status** screen is displayed. Expand **Trunks** in the left pane and select the SIP line in this case “11” SIP trunk to FAXCOM.

Verify that the **SIP Trunk Summary** screen shows an active channel with **Current State** of “Connected”. Also verify that the **Other Party on Call** contains the proper information for the trunk “Line: 17 SIP bvwddev.com” to simulated PSTN, as shown below

The screenshot displays the Avaya IP Office System Status application. The left-hand navigation pane shows a tree structure with 'System' expanded, and 'Trunks (5)' selected. Under 'Trunks', 'Line: 11' is highlighted. The main content area shows the 'SIP Trunk Summary' for Line 11. The summary includes the following details:

- Line Service State: In Service
- Peer Domain Name: 10.10.98.26
- Resolved Address: 10.10.98.26
- Line Number: 11
- Number of Administered Channels: 0
- 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: 128
- SIP Trunk Channel Licenses in Use: 0
- SIP Device Features: (indicated by a green circle and 0%)

Below the summary is a table with columns: Cha..., U., Call Curr..., Time Remote C..., Con..., Caller Other, Dire..., Rou..., Rec..., Rec..., Tra..., Tra... The table contains one row of data. At the bottom of the application, there is a row of buttons: Trace, Trace All, Pause, Ping, Call Details, Graceful Shutdown, and Force Out of Service.

## 7.2. Verify Biscom FAXCOM

In the **FAXCOM** screen, verify all the ports in **Fax Ports** window shows that all ports are idle.

The screenshot displays the FAXCOM software interface. The main window is titled "Job Statistics" and contains several sections:

- Data Selection:** Includes "Calculation Method" (Cumulative Count selected, Hourly Average unselected), "Time Span" (System Lifetime unselected, Since Counter Reset selected, Last Hour unselected), and "Job Type" (All selected, Transmit unselected, Receive unselected).
- System Lifetime:** Started at: 12/05/2016 16:03 -0500 Elapsed: 41 Days 05 Hours 35 Minutes
- Counter Last Reset at:** 12/05/2016 21:38 -0500 Elapsed: 0 Days 00 Hours 00 Minutes
- Active Fax Ports:** Total: 0 Transmitting: 0 Receiving: 0. Includes "Reset Counter" and "Performance Monitor" buttons.
- Fax Port Table:**

Fax Port	Attempts	Pages	Successful	Conn Errors	Non-conn Errors
All Fax Ports	0	0	0	0	0
fax01	0	0	0	0	0
fax02	0	0	0	0	0
fax03	0	0	0	0	0
fax04	0	0	0	0	0

The "Tasks" window at the bottom shows a list of tasks:

Task ID	Source	Fax Port	Status
0002			FAX ports enabled (12 tx, 12 rx)
0001			FAXCOM workflow enabled
0003	host1		FAXCOM service active via TCP:6000
0004	host2		FAXCOM service active via TRAN:6001

The "Fax Ports" window on the right shows a list of ports and their status:

Name	Mode	Status
fax01	Transmit/Receive	Idle
fax02	Transmit/Receive	Idle
fax03	Transmit/Receive	Idle
fax04	Transmit/Receive	Idle
fax05	Transmit/Receive	Idle
fax06	Transmit/Receive	Idle
fax07	Transmit/Receive	Idle
fax08	Transmit/Receive	Idle
fax09	Transmit/Receive	Idle
fax10	Transmit/Receive	Idle
fax11	Transmit/Receive	Idle
fax12	Transmit/Receive	Idle

## 8. Conclusion

These Application Notes describe the configuration steps required for Biscom FAXCOM 6.5.5.13 to successfully interoperate with Avaya IP Office R10. All feature and serviceability test cases were completed with an observation noted in **Section 2.2**.

## 9. Additional References

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

1. *IP Office Manager R10*, Document 15-601011, available at <http://support.avaya.com>.
2. *FAXCOM\_Server\_Administrator's\_Guide*.
3. *FAXCOM-Fax-Server-Data-Sheet*.

---

**©2017 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect +Program at [devconnect@avaya.com](mailto:devconnect@avaya.com).