



## IP Office Technical Tip

Tip no: 188

Release Date: September 27, 2007

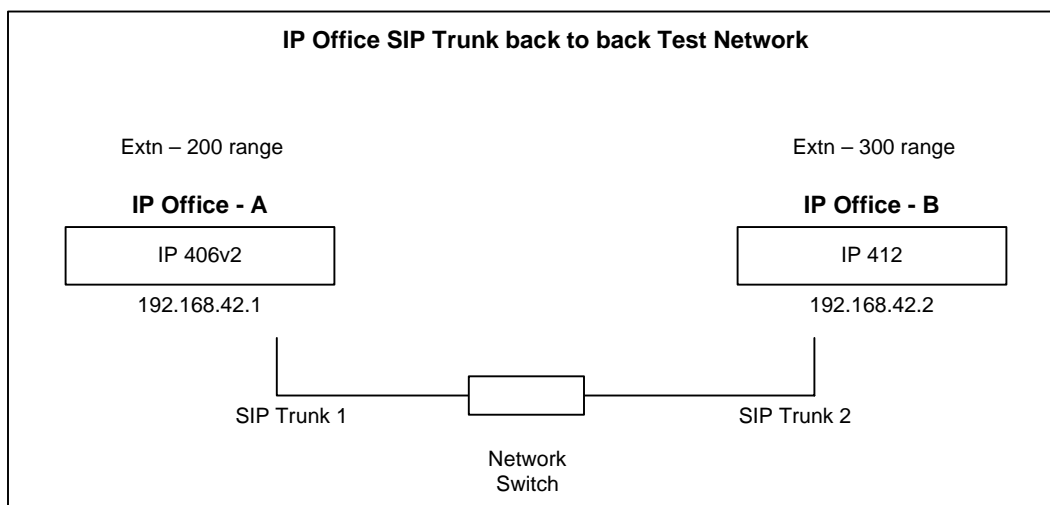
Region: GLOBAL

---

### Verifying IP Office SIP Trunk Operation

#### IP Office back-to-back SIP Line testing

IP Office Release 4.0 supports SIP trunking. Many Internet Telephony Service Providers (ITSP) offer SIP service for connectivity but the programming and operation of SIP trunks can vary greatly between ITSP's. Improper configuration is the leading cause of reported issues. To eliminate the IP Office as a source of issues it is recommended to test the IP Office SIP Line functionality back-to-back, ensuring correct operation of the IP Office hardware and software. This document details the configuration and programming to verify SIP Line functionality in back-to-back operation between two IP Offices.



Verifying IP Office SIP Trunk Operation.....	1
IP Office back-to-back SIP Line testing.....	1
Equipment Required .....	3
Configuration of IP Office – A (IP406v2) .....	3
IP Office SIP Trunk Channels License – IP Office - A.....	3
IP Office System > LAN > Gatekeeper – IP Office - A .....	3
IP Office System > LAN > Network Topology – IP Office - A .....	4
Create a new SIP Line – IP Office - A.....	4
Configure the SIP Line tab – IP Office - A.....	5
Configure the SIP URI tab – IP Office – A (URI - Uniform Resource Identifier) ....	7
Configure the ARS (outbound) – IP Office - A .....	9
Define a System Short Code – IP Office - A .....	10
User Data – SIP tab.....	11
Define an Incoming Call Route – IP Office - A .....	11
Configuration of IP Office – B (IP412).....	12
IP412 – SIP Channel License .....	12
IP412 – Gatekeeper Tab.....	13
IP412 – Network Topology Tab .....	13
IP412 – SIP Line Tab.....	14
IP412 – SIP URI Tab .....	14
IP412 – SIP ARS Tab .....	15
IP412 SIP Short Code Tab .....	15
User Data – SIP tab.....	15
IP412 – SIP Incoming Call Route Tab .....	16
Testing and Verifying Operation.....	16

## **Equipment Required**

Two (2) IP Office's with Release 4.0.5 (minimum) installed.  
Two (2) SIP Trunk Channels License Keys (valid) – one for each IP Office.  
One LAN switch.

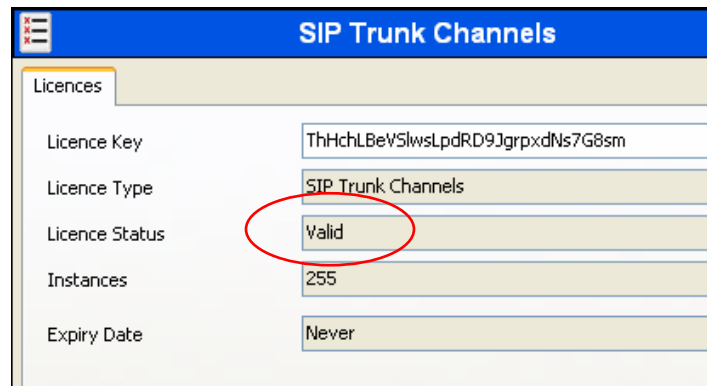
### ***NOTE 1: IP Office Small Community Network (SCN) operation***

*Before proceeding, configure the two IP Offices for typical SCN operation using the IP addresses and extension numbers listed in the diagram. When configured initiate test calls end to end to verify operation. This step is important to ensure proper configuration and operation of basic IP networking between the IP Offices necessary before SIP trunk testing. When successful with this step continue to the SIP configuration section. Additional detail of SCN configuration is available in the IP Office documentation.*

## **Configuration of IP Office – A (IP406v2):**

### **IP Office SIP Trunk Channels License – IP Office - A**

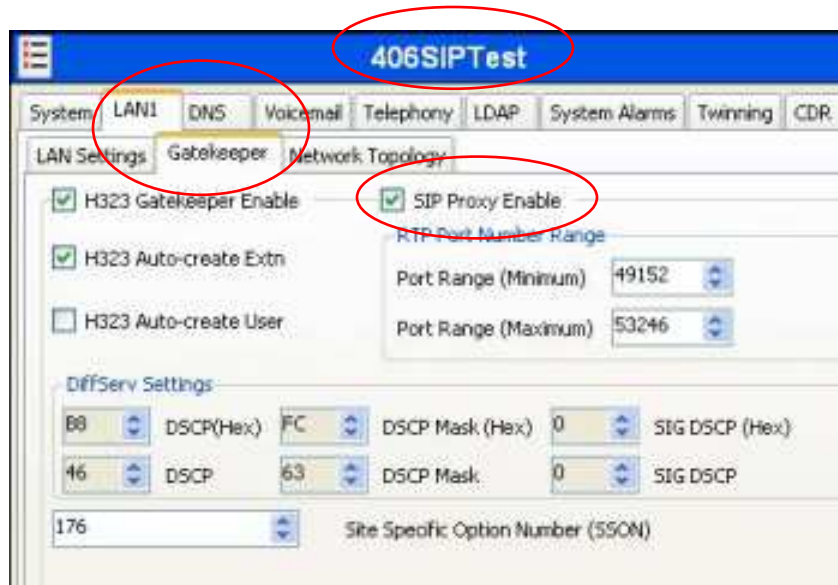
Check that each IP Office License Key for SIP Trunk Channels is VALID. Without a valid SIP license access to the SIP Line configuration function will not be available.



SIP Trunk Channels	
Licences	
Licence Key	ThHchLBeVSlwsLpdRD9JgrpxdNs7G8sm
Licence Type	SIP Trunk Channels
Licence Status	Valid
Instances	255
Expiry Date	Never

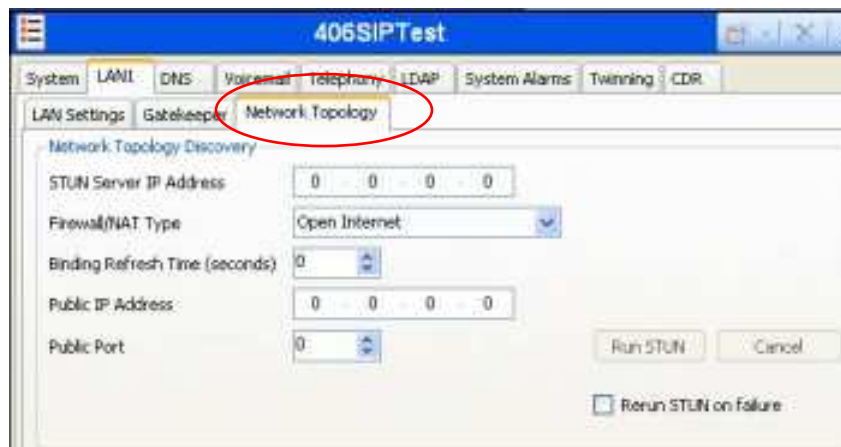
## **IP Office System > LAN > Gatekeeper – IP Office - A**

Display the IP Office > System > LAN > Gatekeeper tab and ensure the “SIP Proxy Enable” function is checked and enabled. The System tab displayed below is that of the IP 406V2 (IP Office – A in the network diagram). Leave, or set as displayed, all other settings (default shown).



**IP Office System > LAN > Network Topology – IP Office - A**

Check the tab immediately to the right of the “Gatekeeper” tab – the Network Topology settings. This tab may be necessary in the field for the STUN discovery functionality. For the back-to-back tests configure the tab as displayed below. Additional information regarding the STUN function is available in the documentation and HELP files for use in the field.



**Create a new SIP Line – IP Office - A**

With the SIP Channel licenses valid go to the IP Office configuration tree “Line” section. Within the Line section right button click and select “New” > “SIP Line”. The IP Office will automatically assign a Line Number for the new SIP Line. It is recommended if the default Line number is below 16 this be changed to a Line number high than 16. The below example has been changed to 16. The change is done in the SIP Line tab displayed on the next page. In field deployments enter the Line number (if provided) by the ITSP. The Line number must be a unique line identifier within this IP Office.

Line	
Line Number	Line Type
1	IP
5	Analogue Trunk
6	Analogue Trunk
7	Analogue Trunk
8	Analogue Trunk
16	SIP Line

### **Configure the SIP Line tab – IP Office - A**

Creating a SIP Line will display the frame on the next page. This tab sets the basic SIP protocol settings used to communicate with the SIP service. The settings of these tabs in the field must be coordinated closely with your ITSP. Continue configuration of the individual fields as displayed for the back-to-back tests. Additional information is available in the documentation.

#### **Line Number**

Automatically defined by the IP Office when created. This has been changed to “16” per previous recommendation. This can be changed to match the Line number provided by the ITSP in field use. The Line number must be unique for this IP Office.

#### **ITSP Domain Name**

Internet Telephony Service Provider (ITSP). For the test network “example.com” is entered as the domain name. In field use you will enter the domain name provided by the ITSP.

### **ITSP IP Address**

Enter the IP address of the remote SIP device – 192.168.42.2 - this is the IP address of IP Office – B (IP412) in the test network. In the field this address will be provided by the ITSP and represent the SIP device the IP Office will exchange SIP messaging/protocol with in the ITSP.

The following fields are not used in the tests, they are not required. The fields should be blank by default, if not, delete the information leaving the fields as displayed above. The ITSP may require (and will then provide) this information for their SIP service if necessary. The fields and their use are self explanatory and additional information is available in the documentation and HELP files.

Registration Required – checked if any of the following fields are required

Primary Authentication Name

Primary Authentication Password

Primary Registration Expiry – by default set to 60

Secondary Authentication Name

Secondary Authentication Password

Secondary Registration Expiry – by default set for 60

### **In Service**

Check that the SIP Line is “in-service” (Default setting).

### **Use Tel URI**

Not used in the tests. This setting is dependant on the ITSP and the SIP service setup. If required by the ITSP check this field and then change the ITSP Domain Name field from the Domain Name (example.com) to the telephone number associated with the SIP service (example 7325551212) received from the provider.

**NOTE 2:** *Be aware that some SIP provider equipment may not currently support the IETF RFC 3966 Tel URI format standard.*

### **VoIP Silence Suppression**

Unchecked by default and unchecked for the test. In the field verify operation and use with the ITSP.

### **RE-INVITE Supported**

Unchecked by default and unchecked for the test. In the field verify operation and use with the ITSP and enable if necessary.

### **Voice Packet Size and Compression Mode**

For the test network the compression mode is left at “Automatic select”. The IP Office will negotiate the codec for voice compression with the remote SIP service. This field automatically sets the voice packet size to “0” in the display. If the ITSP has defined a specific codec for use change the “Compression Mode” to the codec specified. Changing the “Compression Mode” to a specific codec will change the “Voice Packet Size” to the correct packet size accordingly.

## **Network Configuration settings**

### **Layer 4 Protocol**

By default set to “UDP”. The other setting available through the pull down menu is “TCP”. In the field verify either “UDP” or “TCP” port selection and operation with the ITSP and set accordingly.

### **Send Port**

By default set to “5060”. Several other ports are available through the pull down menu if necessary and used in coordination with the “Layer 4 Protocol” setting. Generally UDP port 5060 and TCP port 5061 is used for SIP protocol operation per IETF RFC 3261. NAT, firewalls and other factors in the ITSP network may dictate use of other ports. In the field set as necessary in coordination with the ITSP information supplied.

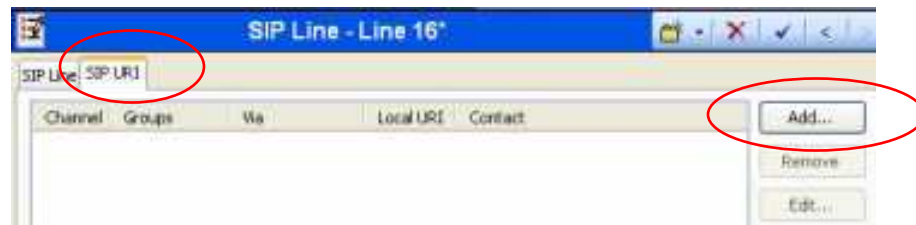
### **Use Network Topology Info**

Select the LAN interface of the IP Office used for SIP communications. In the IP406v2 the interface available is LAN1.

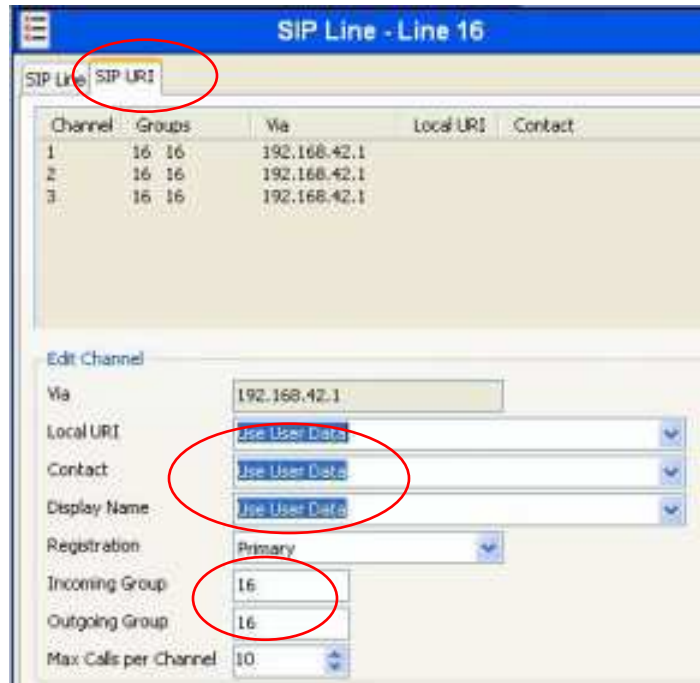
### **Configure the SIP URI tab – IP Office – A** (URI - Uniform Resource Identifier)

This tab is available and used with the SIP Line configuration. In the IP Office this configuration can act as ‘trunks’ across the SIP Line. Additional information regarding the SIP URI tab is available in documentation and the HELP files.

The basic display provides an overview of the channel configuration (empty by default). The detail configuration tabs become available when selecting “ADD” (defining a new entry) or “EDIT” to change an existing entry highlighted.



In the SIP URI tab display select and click “**ADD**” (displayed next page...)



**Via = “IP Address”** (example 192.168.42.1)

Automatically assigned by the IPOffice. The address is assigned from the LAN interface selected in the “SIP Line” definition “Use Network Topology Info”. It identifies the LAN interface and the IP address of the SIP Line in the IP Office.

**NOTE 3:** The below marked (\*) entries must be configured in coordination with the ITSP for field use. For the test network set as below. These three (3) fields: Local URI, Contact, and Display Name, have pull down menu settings and can be modified directly by the Administrator to a specific entry. If administratively edited the text entered will be used on all outbound calls using this SIP URI. Check with the ITSP regarding names and naming conventions used in your deployment.

**\* Local URI**

Change from the default “Use Authentication Name”, for the test network use the pull down menu and select “Use User Data”. With SIP enabled each User now has a “SIP” data tab the programming and use of which is describe in detail later in the document. With the menu set to ‘Use User Data’ the data in the User SIP tab will be used to populate this field for the outgoing call.

**\* Contact**

Change from the default “Use Authentication Name”, use the pull down menu to select “Use User Data”.

**\* Display Name**

Change from the default “Use Authentication Name”, use the pull down menu to select “Use User Data”.



### Registration

Default to Primary – leave at default - registration is not used in the test network. Check with the ITSP to verify if Primary or Secondary registration is used and used per call.

### Incoming Group

Default is zero (0). The SIP URI trunk group number assigned by the System Administrator for this SIP URI. Best practice is to set the Group number to that used / assigned to the SIP Line, in this instance SIP Line 16.

### Outgoing Group

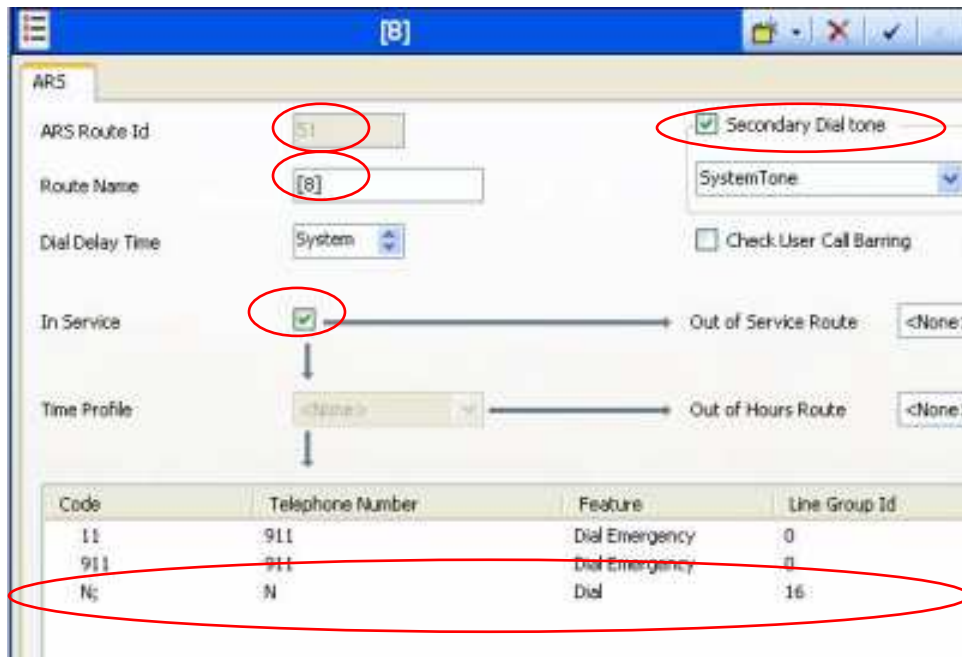
Default is zero (0). This is the SIP URI trunk group number assigned by the System Administrator for this SIP URI Outgoing Group. Best practice is to set the Group number to that used / assigned to the SIP Line, in this instance SIP Line 16.

### Max Calls per channel

Default to 10. Leave the default for use in the test network. This value can be changed in the field in coordination with the ITSP service settings.

### Configure the ARS (outbound) – IP Office - A

After configuration of the SIP Line and SIP URI define the ARS for directing the outbound calls to the SIP Line. For the test network the intent is to 'dial 8' (receive secondary dial tone) and then dial a number defined on the remote system, directing the call to the remote system via the SIP Line. Select the ARS section from the configuration tree and right button click "New". A display similar to the next page will be shown.



**ARS Route Id**

Defined by the IP Office by default – not administrator selectable. In this example “51” has been assigned by the system.

**Route Name**

In the test network example defined as “[8]” in an effort to keep naming and numbering convention similar. The Route Name could also be defined with an alpha character name (such as “SIP”).

**Dial Delay Time**

Each ARS route can be administratively set for a specific Dial Delay Time. The test network will use the default “System” dial delay time.

**In Service**

Checked and enabled by default with the creation of a SIP Line.

**Out of Service Route**

In the event this Route/Line combination is out of service an alternate route can be selected. In the test network there is no alternate route selected ( <None> ).

**Time Profile**

As no Time Profiles are defined in the System (not used for the test network) none are available for selection.

**Code, Telephone Number, Feature, Line Group**

Each ARS can be configured with a series of Short Codes to match dialled numbers. These are specific to the ARS and can vary with each ARS. By default the two “Dial Emergency” codes shown are populated. For the test network define the third “N;” code as listed. Note the Line Group ID the call is directed to is Line Number of “16” defined in the SIP Line and SIP URI sections of this document.

**Secondary Dial Tone – System Tone**

Check (enable) the Secondary Dial Tone setting. The System will provide secondary dial tone when this ARS is selected through the Short Code defined to access the route (the Short Code will be programmed in the next step). When checked the system makes available the “Tone” pull down menu and defaults to “System Tone”. Leave or set as displayed for the test network.

**Check User Call Barring**

Unchecked for the test network. This can be used in the field as needed by the customer system requirements.

**Define a System Short Code – IP Office - A**

For the test network the intent is to ‘dial 8’ (receive secondary dial tone) and then dial a number defined on the remote system, directing the call to the remote system through the ARS to the SIP Line. Define a System Short Code to direct calls to the ARS Route which will select the Line Group to direct the call to – in this case SIP Line 16. Note the pull down menu “Line Group Id” used to select the ARS Route “51 : [8]” created in the previous step.

8N: Dial	
Short Code	
Code	8N
Feature	Dial
Telephone Number	N
Line Group Id	51: [8]
Locale	
Force Account Code	<input type="checkbox"/>

### **User Data – SIP tab**

With the introduction of SIP, each User will have a new tab for a SIP “User Data” entry. By default this is populated with the “Name” entry of the User. These entries can be individually programmed and can then be used to populate the SIP URI fields; Local URI, Contact, Display Name if ‘Use User Data’ is selected in the SIP URI programming (described earlier in this document). The example below displays User Extn203 being identified in SIP as a global E.164 number – 17328522203. This number will later be programmed in the ‘Incoming Call Route’ to direct any incoming SIP calls – addressed to this E.164 number – to this User’s extension.

Extn203: 203	
User	Voicemail
DND	ShortCodes
Source Numbers	Button Programming
Menu Programming	Twinning
T3 Op	Announcements
SIP	
SIP Name	17328522203
SIP Display Name (Alias)	17328522203
Contact	17328522203
Anonymous	<input type="checkbox"/>

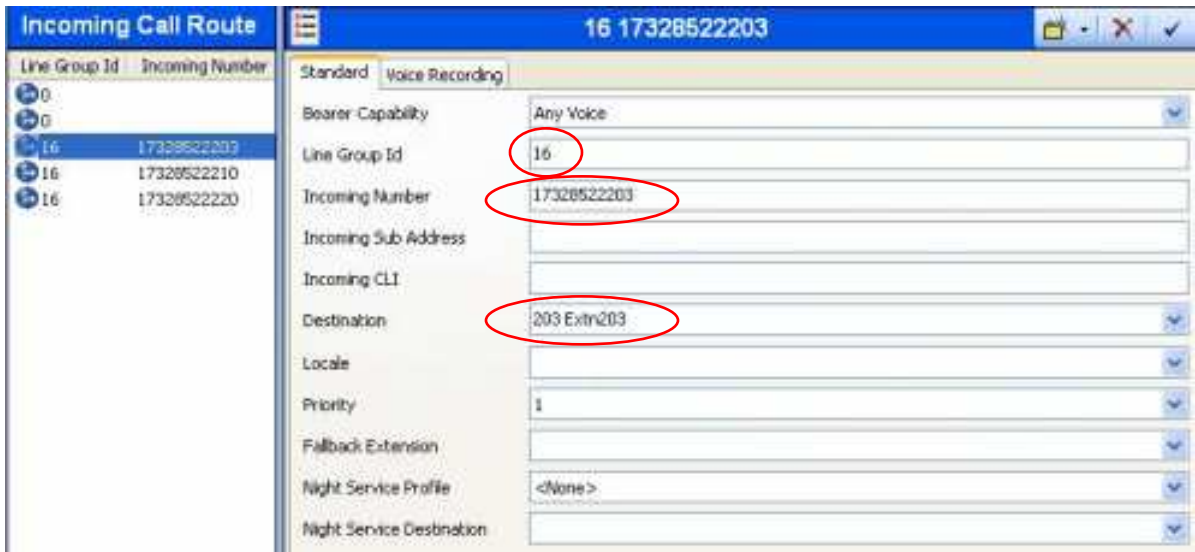
### **Define an Incoming Call Route – IP Office - A**

Define an “Incoming Call Route” for each Line, Incoming Number and the Destination. In the field you can define as many entries as necessary for the customer network.

Line Group Id = 16 (the SIP Line and SIP URI Incoming/Outgoing group defined).

Incoming number = 17328522203 - This will be the E.164 number ‘called’ by the remote and received in SIP INVITE from the remote site. This number also matches that defined in Extn203’s SIP data tab. Both must be defined and match to be identified in the IP Office and the call delivered successfully.

Destination = Use the pull down menu to select Extn203.



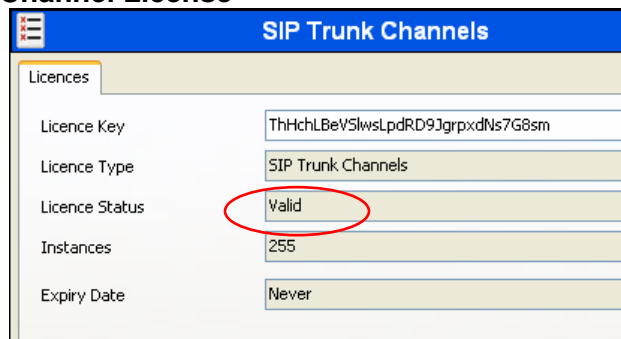
## COMPLETED

This completes the configuration and setup of **IP Office – A** (IP406v2) in the test network. The following configuration screens of **IP Office – B** (the IP412) have been provided without the detail explanations. If necessary refer back to the **IP Office A** (IP406v2) configuration details.

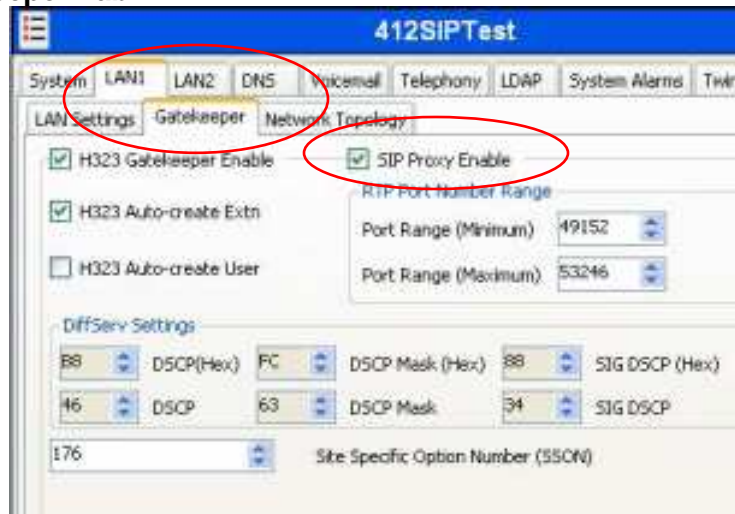
### Configuration of IP Office – B (IP412):

Set the **IP Office – B** (IP412) configuration tabs as per the following screen shots. Check the settings when completed to ensure correct configuration and operation with **IP Office – A** in the test network.

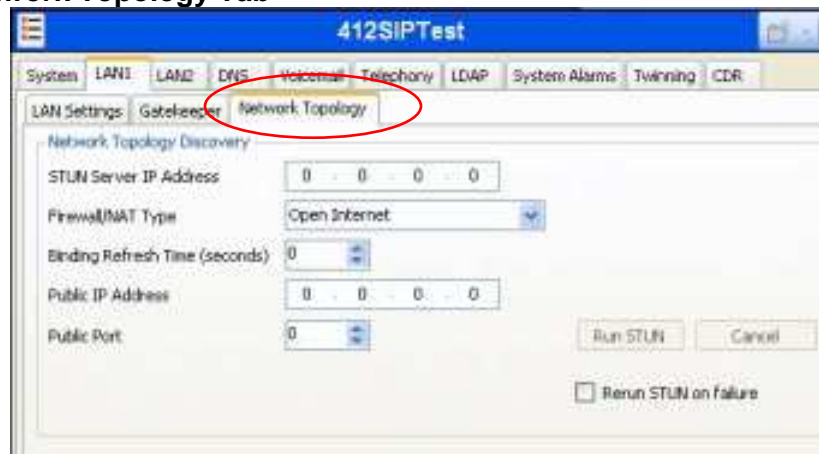
#### IP412 – SIP Trunk Channel License



### IP412 – Gatekeeper Tab



### IP412 – Network Topology Tab



## IP412 – SIP Line Tab

SIP Line - Line 16\*

SIP Line | SIP URI

Line Number	16	Registration Required	<input type="checkbox"/>
ITSP Domain Name	example.com	In Service	<input checked="" type="checkbox"/>
ITSP IP Address	192 . 168 . 42 . 1	Use Tel URI	<input type="checkbox"/>
Primary Authentication Name		VoIP Silence Suppression	<input type="checkbox"/>
Primary Authentication Password		Out Of Band DTMF	<input type="checkbox"/>
Primary Registration Expiry	60	Local Tones	<input checked="" type="checkbox"/>
Secondary Authentication Name		Fax T38	<input type="checkbox"/>
Secondary Authentication Password		RE-INVITE Supported	<input checked="" type="checkbox"/>
Secondary Registration Expiry	60	Voice Packet Size	0
		Compression Mode	Automatic Select

Network Configuration

Layer 4 Protocol	UDP	Send Port	5060
Use Network Topology Info	LAN 1	Listen Port	5060

## IP412 – SIP URI Tab

SIP Line - Line 16

SIP Line | SIP URI

Channel	Groups	Via	Local URI	Contact
1	16 16	192.168.42.2		
2	16 16	192.168.42.2		

Edit Channel

Via	192.168.42.2
Local URI	Use User Data
Contact	Use User Data
Display Name	Use User Data
Registration	Primary
Incoming Group	16
Outgoing Group	16
Max Calls per Channel	10

**IP412 – SIP ARS Tab**

Code	Telephone Number	Features	Line Group Id
11	911	Dial Emergency	0
911	911	Dial Emergency	0
N	N	Dial	16

**IP412 SIP Short Code Tab**

**User Data – SIP tab**

## IP412 – SIP Incoming Call Route Tab

Line Group Id	Incoming Number
0	
0	
16	17328523302
16	17328523326

Bearer Capability	Any Voice
Line Group Id	16
Incoming Number	17328523302
Incoming Sub Address	
Incoming CLI	
Destination	302 Extn302
Locale	
Priority	1
Fallback Extension	
Night Service Profile	<None>
Night Service Destination	

### COMPLETED

This completes the configuration and setup of **IP Office – B (IP412)** in the test network. With both IP406v2 and IP412 properly configured as per the above screen shot examples the IP Offices will use the SIP Line (back-to-back) to initiate and connect calls using the SIP Line.

### Testing and Verifying Operation

For the test network the intent is to 'dial 8' (receive secondary dial tone) and dial a number on the remote system (one that has been configured through the above examples), directing the call to the remote system via the back-to-back SIP Line.

Initiate a call from IP Office – A (IP406v2) extension 203. Dial "8", receive secondary dial tone, then finish dialling "17328523302" (the extension number of the first expansion module port 1 digital phone) on IP Office – B (IP412).

After the System Dial Delay Time IP Office – A will initiate a SIP INVITE across the SIP Line to the remote system (IP Office – B – IP412). This can be viewed using the IP Office Monitor application and checking the display boxes in the SIP filter tab. Extension 302 on the remote system will begin ringing and the phone display should show "17328522203@example.com". This verifies the SIP Line in operation, as the incoming call is identified as being received from "17328522203" (Extn203) and the "example.com" Domain name defined in the IP406v2 SIP Line.

Initiate a call from IP Office – B (IP412) extension 302. Dial "8", receive secondary dial tone, then finish dialling "17328522203" (the extension number of the first DS port on the front of the IP Office – A (IP406v2).

After the System Dial Delay Time IP Office – B will initiate a SIP INVITE across the SIP Line to the remote system (IP Office – A – IP406v2). This can be viewed using the IP Office Monitor application and checking the display boxes in the SIP filter tab. Extension 203 on the remote system will begin ringing and the phone display should show "17328523302@ example.com". This verifies the SIP Line in operation, as the incoming



call is identified as being received from “17328523302” (Extn302) and the “example.com” Domain name defined in the IP412 SIP Line.

*Issued by:*  
*Avaya GSS Tier 4 Support*  
*Contact details:-*  
*EMEA/APAC*  
*Tel: +44 1707 392200*  
*Fax: +44 (0) 1707 376933*  
*Email: [gsstier4@avaya.com](mailto:gsstier4@avaya.com)*

*NA/CALA*  
*Tel: +1 732 852 1955*  
*Fax: +1 732 852 1943*  
*Email: [IPOUST4ENG@Avaya.com](mailto:IPOUST4ENG@Avaya.com)*

*Internet: <http://www.avaya.com>*  
© 2007 Avaya Inc. All rights reserved.