

Avaya IP Office Standard Trusted IP / SIP Gateway Set Up Telquest Tech Support

We will be using LAN2 in this example.

This screenshot shows the 'LAN2' configuration tab in the Avaya IP Office software. The 'LAN Settings' sub-tab is active. The 'IP Address' field is set to '192 . 168 . 150 . 123' and the 'IP Mask' is '255 . 255 . 255 . 0'. The 'Primary Trans. IP Address' is '0 . 0 . 0 . 0'. The 'Firewall Profile' is set to '<None>' and the 'RIP Mode' is 'None'. The 'Enable NAT' checkbox is unchecked. The 'Number Of DHCP IP Addresses' is set to '1'. The 'DHCP Mode' is set to 'Disabled'. A yellow callout box points to the 'LAN2' tab with the text '1. Click Here...'. Another yellow callout box points to the 'IP Address' field with the text '2. KSU's LAN IP Address.'. A third yellow callout box points to the 'IP Mask' field with the text '3. Set the IP Mask'. A fourth yellow callout box points to the 'Disabled' radio button with the text '4. Click Here...'. A fifth yellow callout box points to the 'Advanced' button with the text 'This is NOT the Router Use IP Route area...'. A sixth yellow callout box points to the 'Set the DNS' text with the text 'Set the DNS'.

1. Click Here...

2. KSU's LAN IP Address.

3. Set the IP Mask

Set the DNS

4. Click Here...

This is NOT the Router Use IP Route area...

Next....

This screenshot shows the 'Network Topology Discovery' sub-tab in the Avaya IP Office software. The 'STUN Server IP Address' is set to '107 . 23 . 150 . 92' and the 'STUN Port' is '3478'. The 'Firewall/NAT Type' is set to 'Open Internet'. The 'Binding Refresh Time (seconds)' is '10'. The 'Public IP Address' is '0 . 0 . 0 . 0' and the 'Public Port' is '5060'. The 'Run STUN' button is highlighted. A yellow callout box points to the 'Network Topology Discovery' sub-tab with the text '1. Click Here...'. Another yellow callout box points to the 'STUN Server IP Address' field with the text '2. Set like this...'. A third yellow callout box points to the 'Run STUN' button with the text '3. Set like this...'. A fourth yellow callout box points to the 'Run STUN on startup' checkbox with the text 'These are populated by Run STUN'.

1. Click Here...

2. Set like this...

3. Set like this...

These are populated by Run STUN

1. Right Click Here and create a new SIP Line

2. Then, Click Here...

1. Click Here...

2. Click Here...

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 | Add...

Edit Channel

1. Put a * (STAR) here...

2. Set each like this...

3. Set to correct Groups...

4. Set to correct Value...

Via: <None>

Local URI: *

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 17

Outgoing Group: 17

Max Calls per Channel: 1

All 3 not used.....

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

Index | UserName | Authentication Name | Contact | Expiry | Register

The image shows two screenshots of the Asterisk SIP configuration interface. The top screenshot is the 'Edit Channel' form, and the bottom screenshot is the 'SIP' tab for a specific user/extension.

Top Screenshot: Edit Channel

- Via:** <None>
- Local URI:** * (Annotated: "This allows all DID Numbers to be routed by the Incoming Call Route...")
- Contact:** Use Internal Data (Annotated: "See this too...")
- Display Name:** Use Internal Data
- PAI:** Use Internal Data
- Registration:** 0: <None>
- Incoming Group:** 17
- Outgoing Group:** 17
- Max Calls per Channel:** 1

Bottom Screenshot: SIP Configuration for EXT207: 207

- 1. Click Here...** points to the 'SIP' tab.
- SIP Name:** Enter the DID Number used by this User/Ext. (Annotated: "This Sets this")
- SIP Display Name (Alias):** EXT207
- Contact:** (Annotated: "This is what is sent out as the Caller ID If your SIP Provider allows it....")
- Annotation:** "If you use DID Numbers for Users, then enter it here... This is the 'Internal Data'. If it isn't here, calls will fail." points to the SIP Name field.

1. Click Here...

**2. Set this...
This allows STUN to work.....**

Enter the URL or the IP Address of the SIP Provider here...

1. Click Here...

SIP Line - Line 17*

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

ITSP Proxy Address

Network Configuration

Layer 4 Protocol UDP Send Port 5060

Use Network Topology Info LAN 2 Listen Port 5060

SIP Line - Line 17*

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials

Line Number 17

ITSP Domain Name

Create an Incoming Call Route for each DID Number:

Here are examples with 3 different DID Numbers, each with a different destination

Incoming Call Route		
Line Group Id	Destination	Incoming Number
17	207 EXT207	5618323801
17	201 ext201	5618323802
17	200 Main	5618323803

17 5618323801	
Standard	Voice Recording
Destinations	
Bearer Capability	Any Voice
Line Group Id	17
Incoming Number	5618323801
Incoming Sub Address	

Incoming Call Route		
Line Group Id	Destination	Incoming Number
17	207 EXT207	5618323801
17	201 ext201	5618323802
17	200 Main	5618323803

17 5618323802	
Standard	Voice Recording
Destinations	
Bearer Capability	Any Voice
Line Group Id	17
Incoming Number	5618323802
Incoming Sub Address	

Incoming Call Route		
Line Group Id	Destination	Incoming Number
17	207 EXT207	5618323801
17	201 ext201	5618323802
17	200 Main	5618323803

17 5618323803	
Standard	Voice Recording
Destinations	
Bearer Capability	Any Voice
Line Group Id	17
Incoming Number	5618323803
Incoming Sub Address	

Inbound Call - SIP Call Flow

When a call comes in on a SIP Line, the Called Number (DID) is passed to the Local URI.

Sample:

INVITE sip:2012351234@192.168.111.248 SIP/2.0

If that Called Number (DID) is not there, the Monitor Program will report “Not Found”.

If that Called Number (DID) is there, it is passed on to the Incoming Call Route.

Make sure that you have
an Inc. Call Route for
EACH DID Number

If you don't
the call will fail

Where it is routed to its destination.

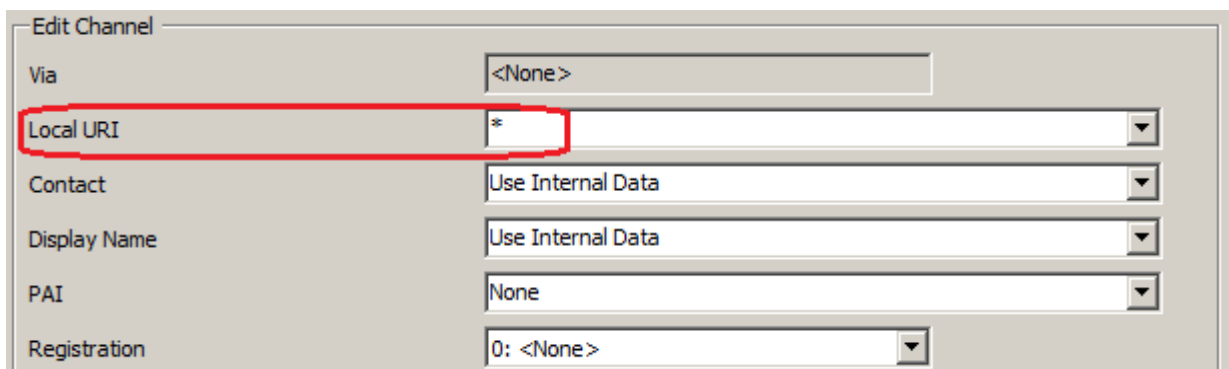
This is an optional method for the Local URI setting on Page 6.

You can also use a * (STAR) as a wild card in the Local URI area.

Then you will not need to create a separate Local URI for each DID Number.

The * will accept all DID Numbers.

You will still need to create an Incoming Call Route for each DID Number.



The image shows a screenshot of a web-based configuration interface titled "Edit Channel". It contains several fields for configuring a channel. The "Local URI" field is highlighted with a red rectangular box and contains an asterisk (*). The other fields are as follows:

Field	Value
Via	<None>
Local URI	*
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	None
Registration	0: <None>