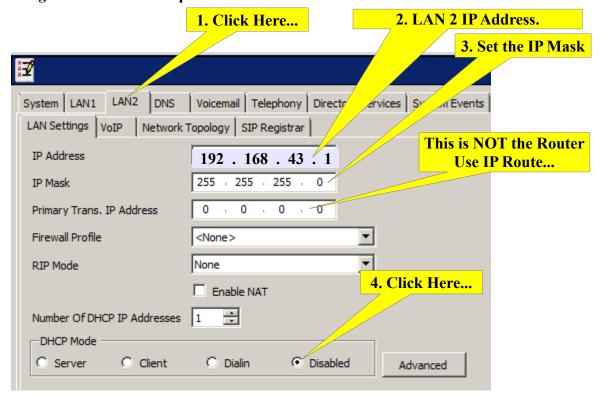
Avaya IP Office Standard SIP Gateway Set Up Telquest Tech Support

We will be using LAN2 in this example. You could also use LAN1 Instead.



Note:

I am using the default IP Address of LAN2.

You may need to change this to something different in order to work with the SIP Providers Local Gateway.

They should be able to tell you their Gateways' Default IP Address and you can just use an IP Address that is one higher or lower.

Example:

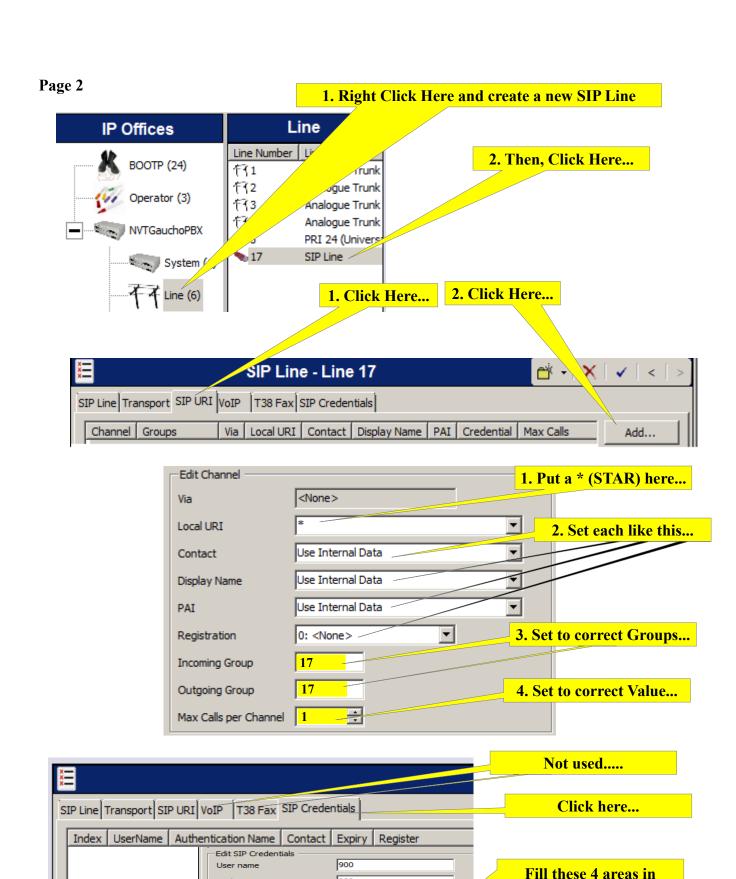
They use 192.168.1.200

You use 192.168.1.199 or 192.168.1.201

Don't go above 254 as the last 3 digits in the IP Address.

I am using 192.168.43.2 as the SIP Providers Local Gateway in this Help Sheet.

Yours may be different...



900

900

60

÷

If needed...

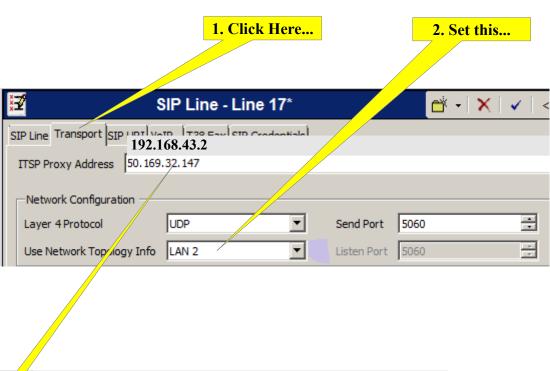
Uncheck...

Authentication Name

Contact

Expiry (mins)
Registration required

		This allows all DID Numbers to be routed by the Incoming Call Route						
Edit Channel			See this too					
Via	None>							
Local URI	*	•						
Contact	Use Internal Data	▼						
Display Name	Use Internal Data	▼						
PAI	Use Internal Data	▼						
Registration	0: <none></none>	<u> </u>						
Incoming Grup	17							
Outgoin Group	17							
May Calls per Channel	1							
User Voicemail VD ShortCodes Source Numbers relephony Forwarding Dial In Voice Recording Buth n Programming Menu Programming Mobility Phone Manager Options Hunt Group Membership Announcements SIP Personal Directory Enter the DID Number used by this User/Ext. SIP Name								
SIP Display Name (Alias) Contact	EXT207							
			ers, then enter it here sn't here, calls will fail.					
This is what is sent out as the Caller ID								
If your SIP Provider allows it								



I. Click Here...

SIP Line Transport SIP URI VOIP T38 Fax SIP Creventials

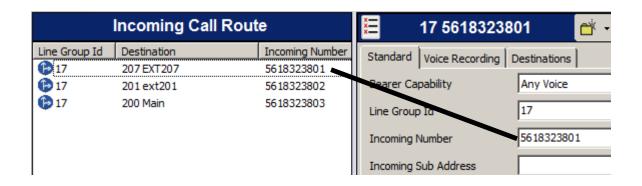
Line Number 17

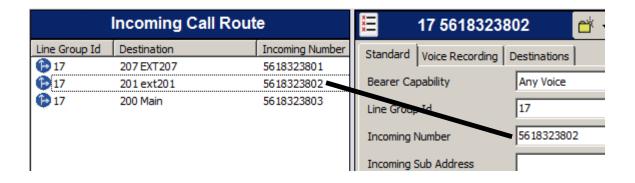
ITSP Domain Name 192.168.43.2

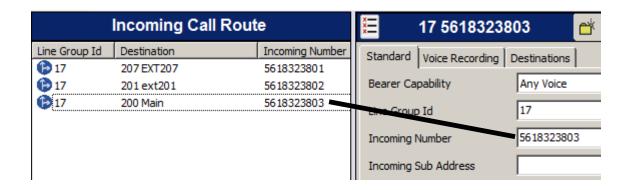
Page 5

Create an Incoming Call Route for each DID Number:

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







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

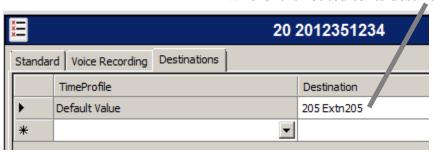
If that Called Number (DID) is not there, the Monitor Program will report "Not Found".

Edit Channel			
Via	<none></none>		
Local URI	2012351234		
Contact	Use Internal Data		
Display Name	Use Internal Data		
PAI	None		
Registration	0: <none></none>		

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

			20 201235123 4		re that you have Call Route for
	Standard Voice Recording	Destinations		EACH	DID Number
	Bearer Capability	Any Voice	V		If you don't the call will fail
l	Line Group ID	20			
l	Incoming Number	2012351234			

Where it is routed to its destination.



Page 7

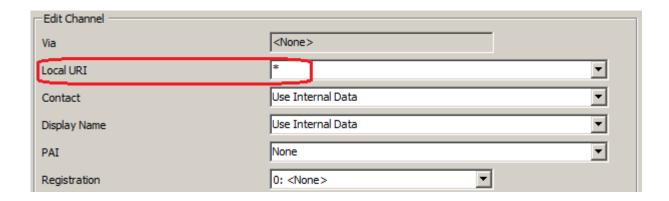
This is an optional method for the Local URI setting on Page 6.

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.



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Add a new IP Route:

This is equal to a "Router" address.

