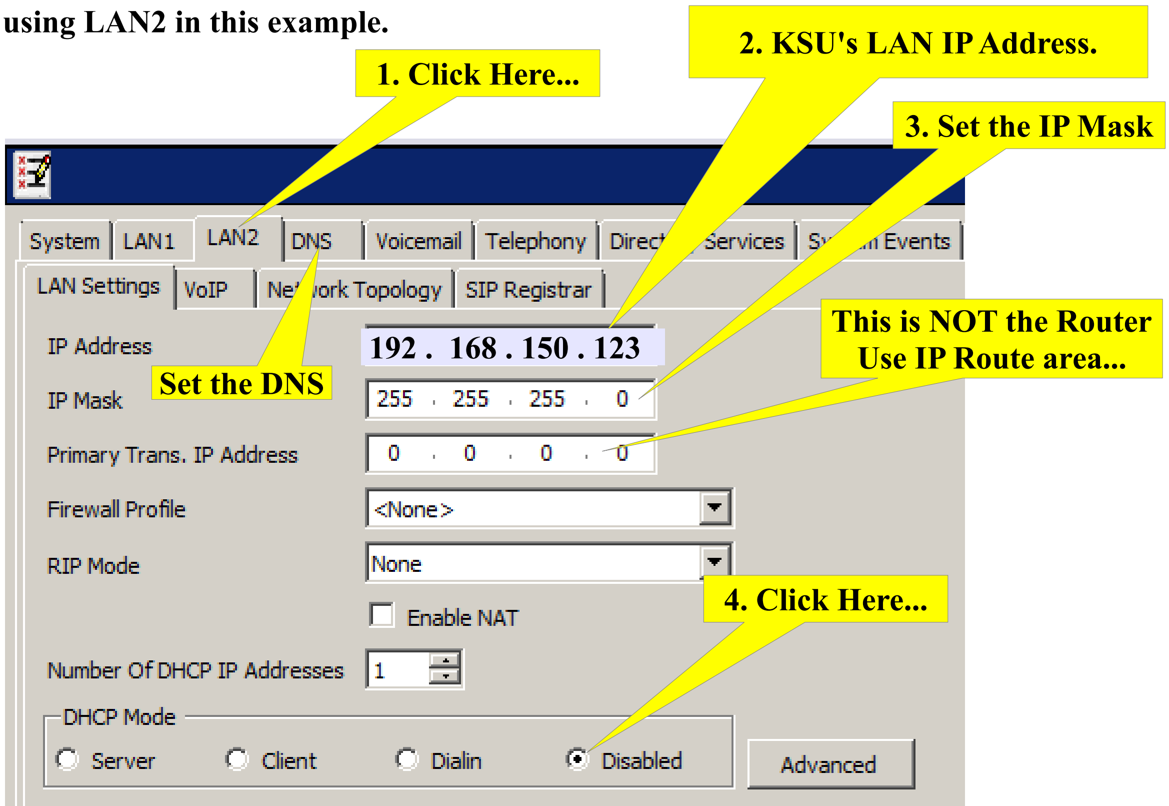
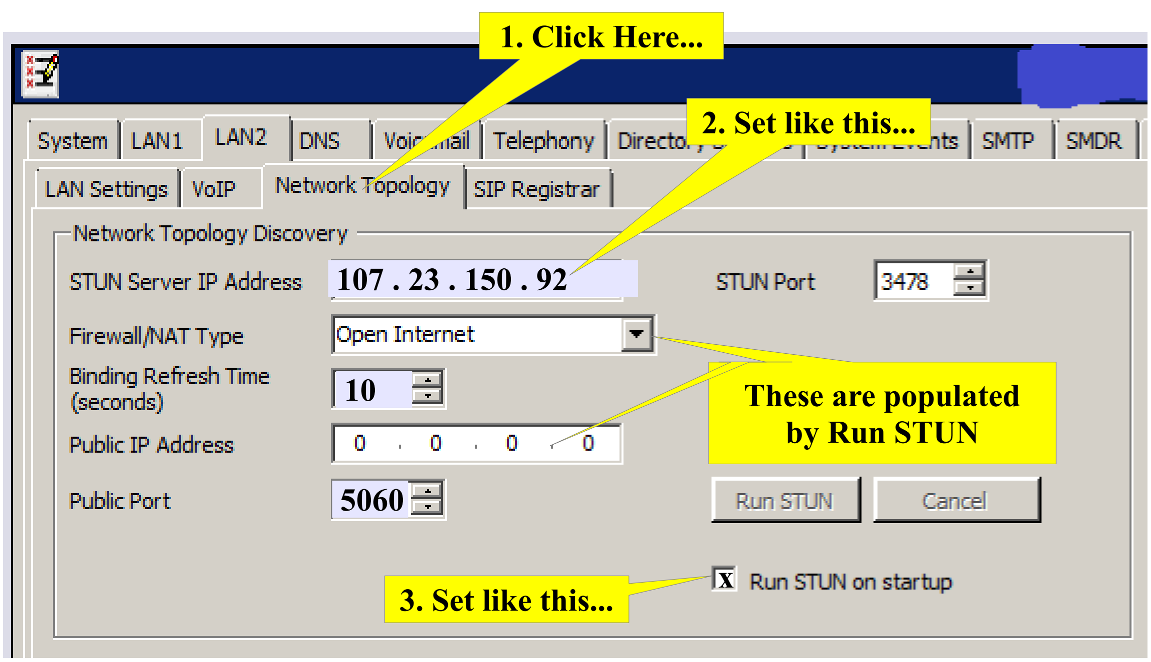
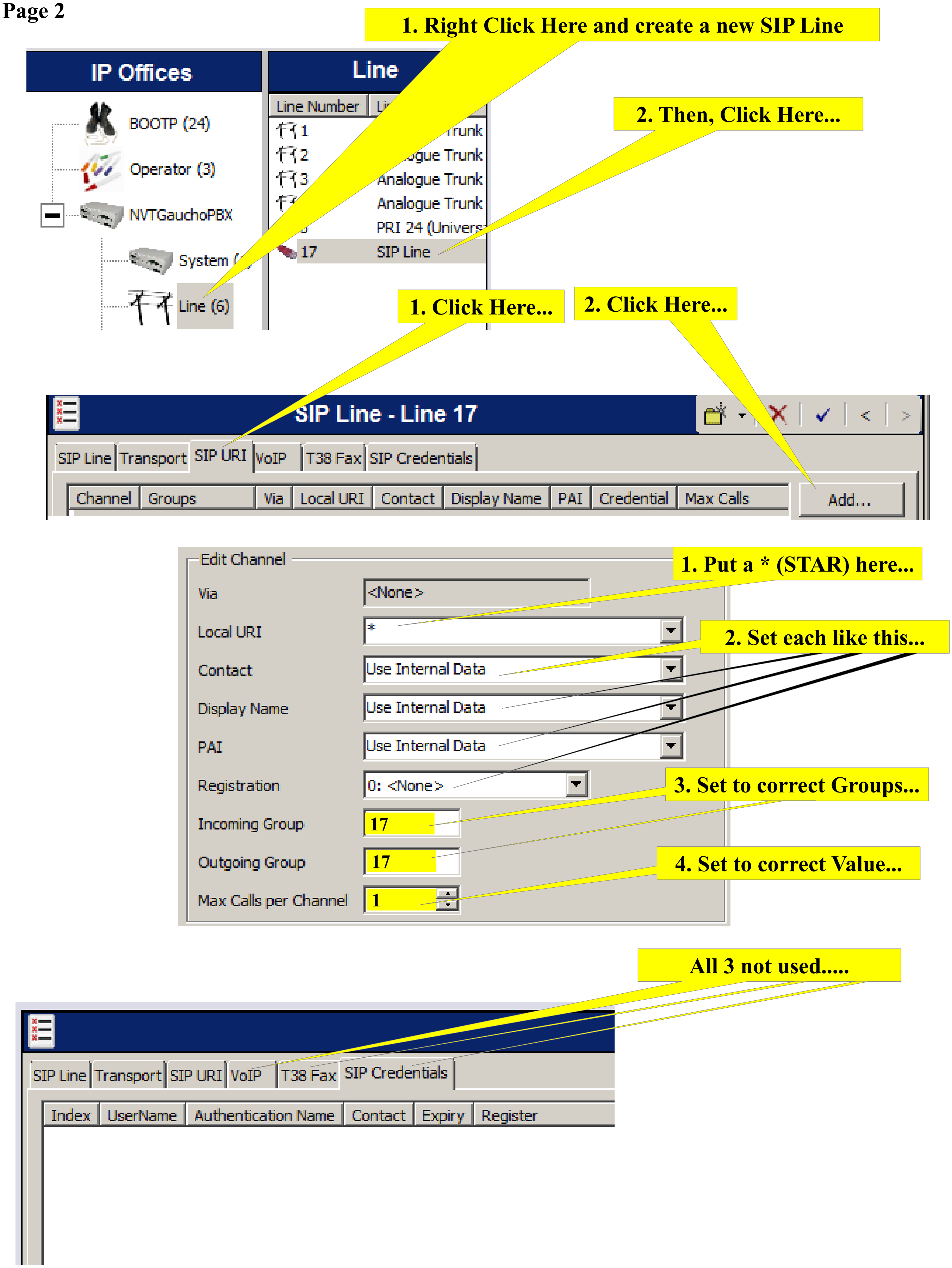
Avaya IP Office Standard
Trusted IP / SIP Gateway Set Up
Telquest Tech Support

We will be using LAN2 in this example.



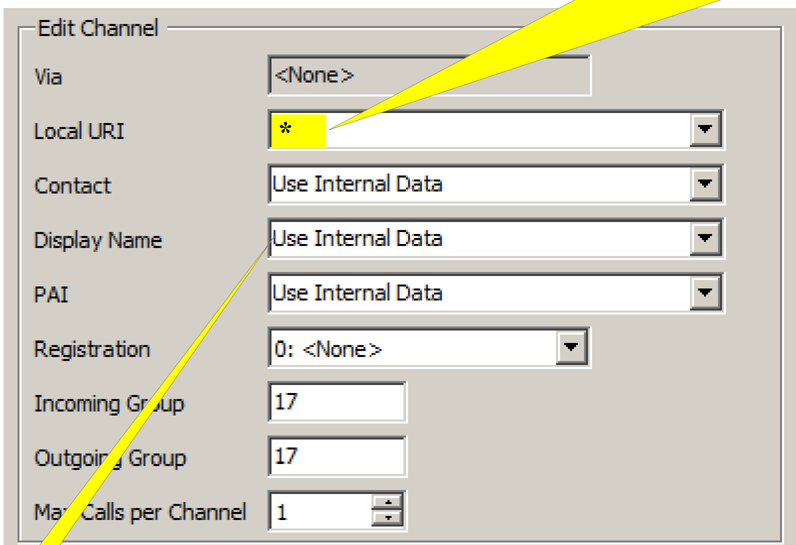
Next....




1. Right Click Here and create a new SIP Line
IP Offices
BOOTP (24)
Operator (3)
NVTGauchoPBX
System
Line (6)
Line
Line Number
Analogue Trunk
Analogue Trunk
PRI 24 (Univers
17
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...
Via
<None>
Local URI
*
2. Set each like this...
Contact
Use Internal Data
Display Name
Use Internal Data
PAI
Use Internal Data
Registration
0: <None>
3. Set to correct Groups...
Incoming Group
17
Outgoing Group
17
4. Set to correct Value...
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

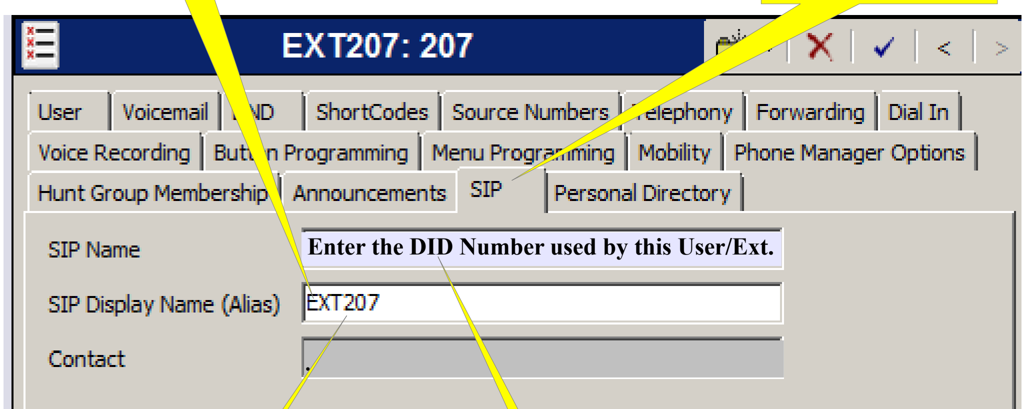
This allows all DID Numbers to be routed by the Incoming Call Route...

See this too...



This **Sets this**

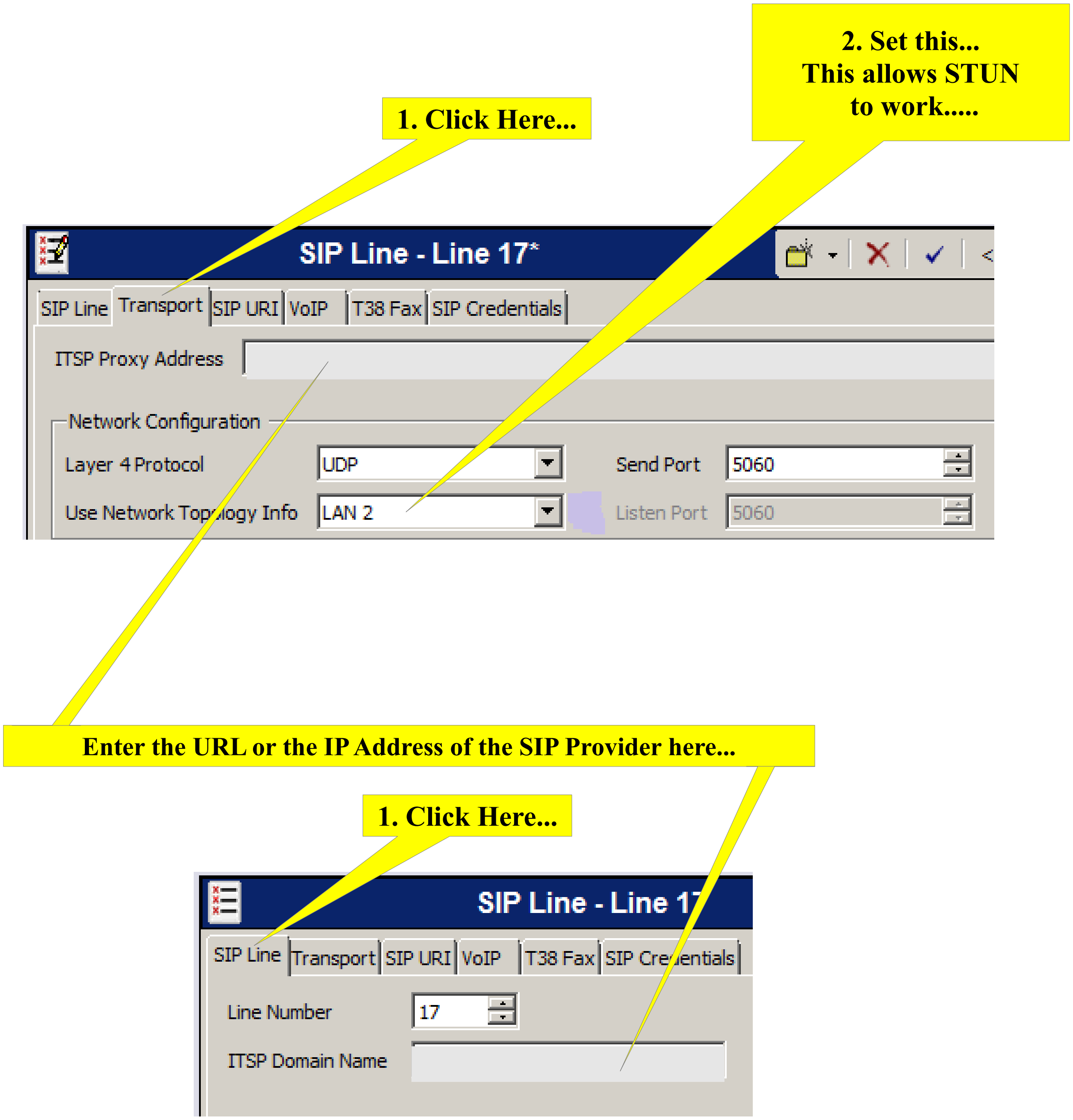
1. Click Here...



If you use DID Numbers for Users, then enter it here...
This is the "Internal Data". If it isn't here, calls will fail.

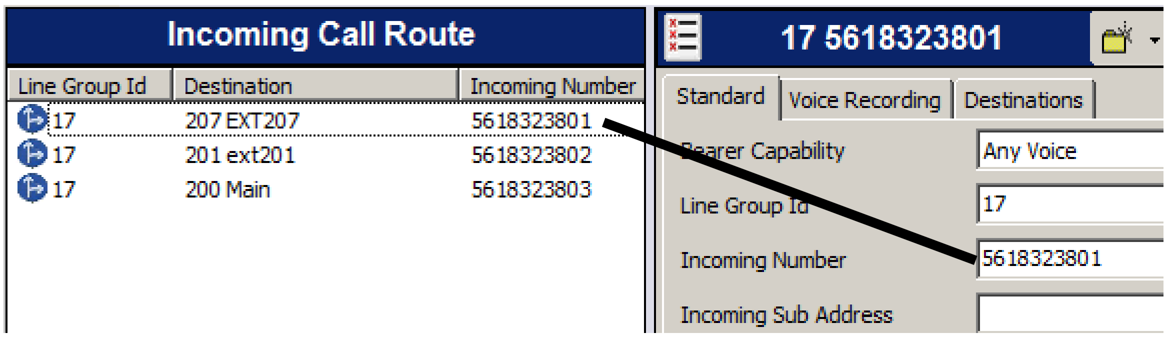
This is what is sent out as the Caller ID

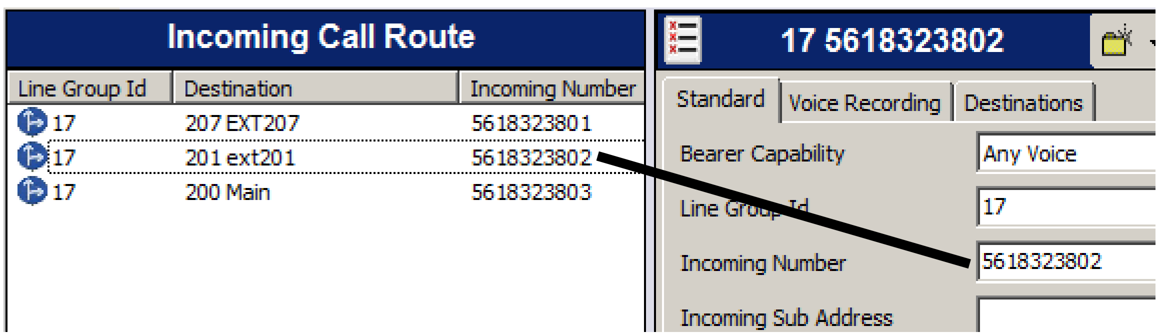
If your SIP Provider allows it....

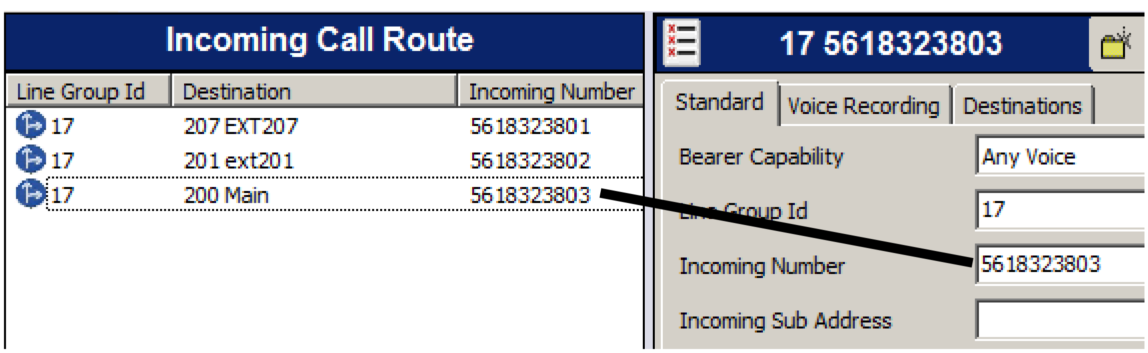

2. Set this...
This allows STUN
to work.....
1. Click Here...
SIP Line - Line 17*
SIP Line
Transport
SIP URI
VoIP
T38 Fax
SIP Credentials
ITSP Proxy Address
Network Configuration
Layer 4 Protocol
UDP
Send Port
5060
Use Network Topology Info
LAN 2
Listen Port
5060
Enter the URL or the IP Address of the SIP Provider here...
1. Click Here...
SIP Line - Line 17
SIP Line
Transport
SIP URI
VoIP
T38 Fax
SIP Credentials
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







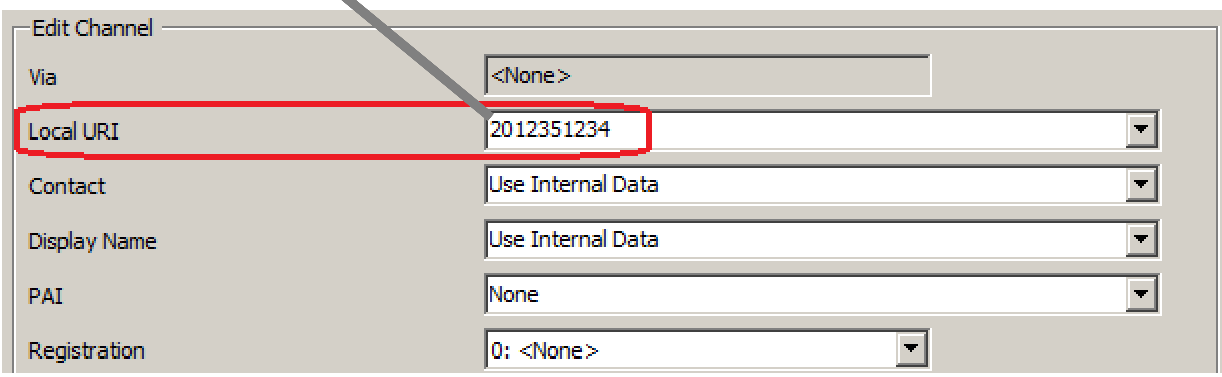
Inbound Call - SIP Call Flow

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

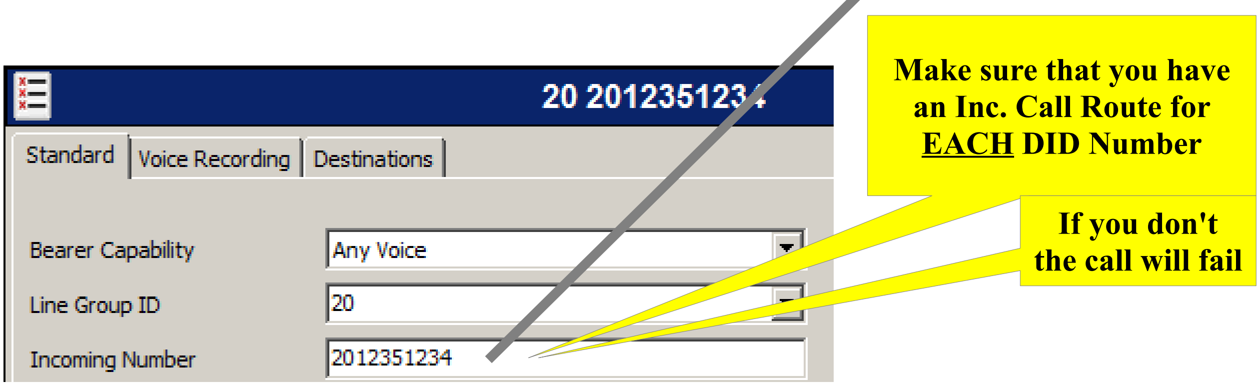
Sample:

```
INVITE sip:2012351234@192.168.111.248 SIP/2.0
```

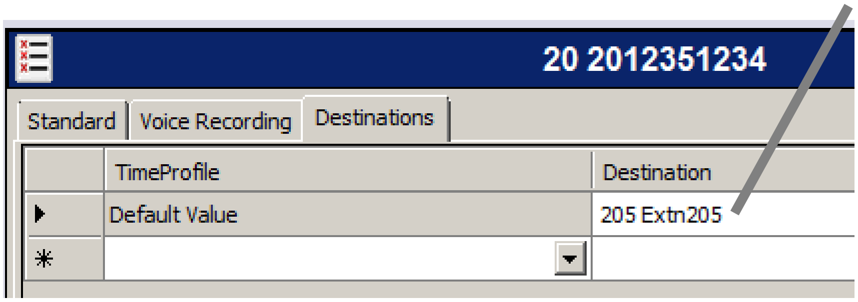
If that Called Number (DID) is not there, the Monitor Program will report "Not Found".



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



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.