Avaya IP Office Basic Mode
Connecting Two Basic Mode Systems via SIP Trunks
Telquest Tech Support

Telquest has developed a configuration that will allow 2 Avaya IP Office Basic Mode KSU's to make and receive calls between each other over the Internet.

Tested on Release 9.0 and 9.1.

This is done by using the 3 Free built in SIP Trunk Channel Licenses. (Voip Link)
This allows up to 3 simultaneous calls between the two systems.
You can add additional SIP Trunk Channel Licenses to increase the amount of simultaneous calls.
A Combo Card is also required to provide the VCM Channels needed for VOIP Communications.

A Router to Router VPN Tunnel is required for Secure Communications between the systems.
You can try using the Open Internet but your connections will be unsecured and open to Hackers.
Telquest does not provide any Tech Support for the Open Internet method.

Due to the Basic Mode system, there are some limitations on what can and cannot be done.
This is a free configuration that is intended to be used to call and transfer callers between two systems.
If it falls short of your needs, then you need to consider the Standard Mode.

You get:

Direct Calling from Extension to Extension

Extension to Calling Group (ring only)

VM Access to and from each system

Auto Attendant Access to and from each system

You can keep the existing Extension Numbers on both ends (No Renumbering required)

Transfer an outside caller from one system to the other but caller cannot be transferred back.

You cannot:

Make outside calls via the Remote system

Page from one system to the other

Use any Feature Codes across the link.

There may be other features that will not work that we are not aware of or have not been tested.

You <u>MUST</u> put the KSU in the PBX Mode.

Note:

When you use the PBX Mode (and you must) it will place 3 Intercom Buttons on each phone.

They are the a=, b= and c= buttons and CANNOT be removed or have any added.

A 1408, 8 button telephone will only allow 5 CO Lines to appear on the phone.

The Voip Link only requires 1 button.

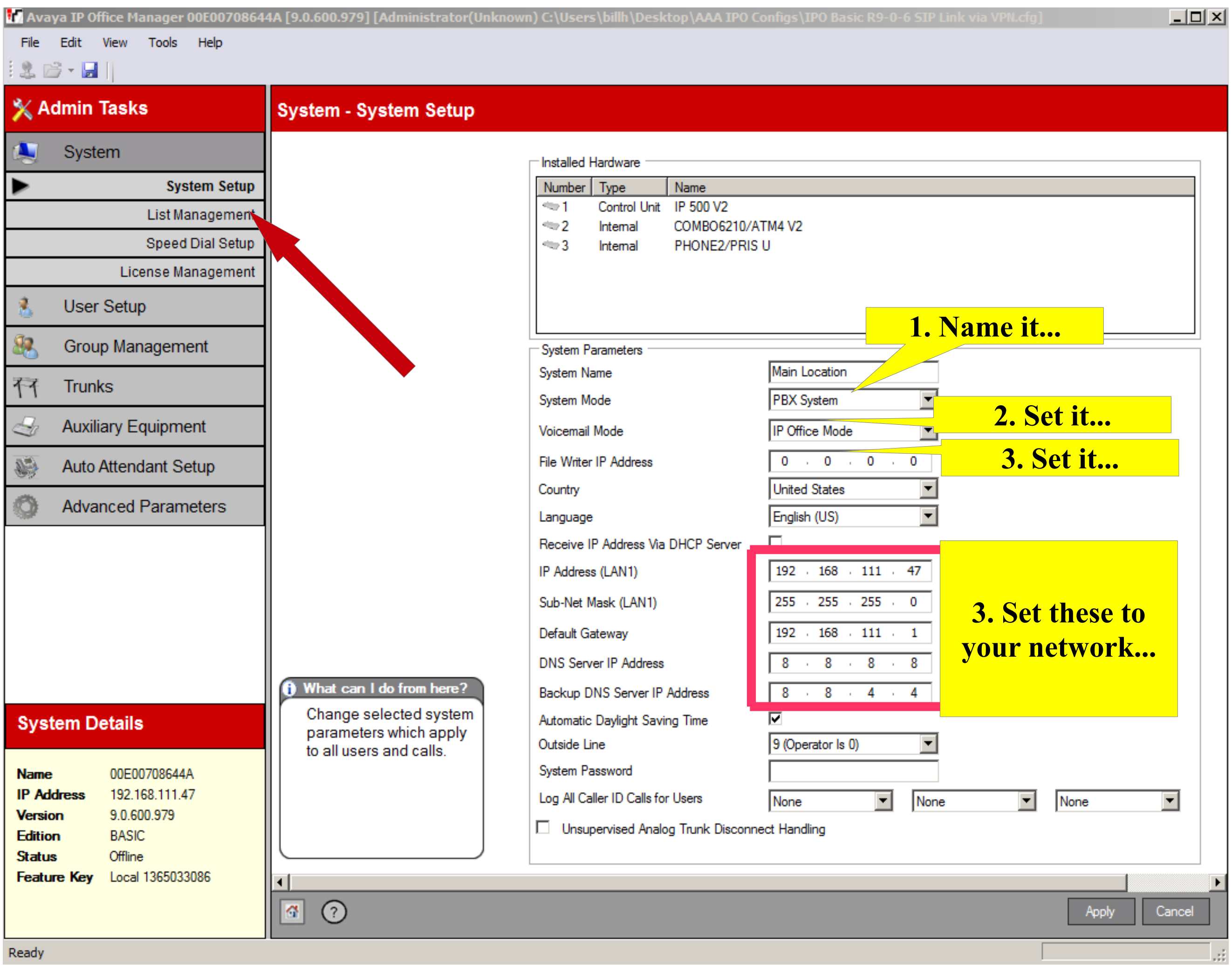
So, watch the type and button capacity of the phones that you use.

1416 telephones would be the best.

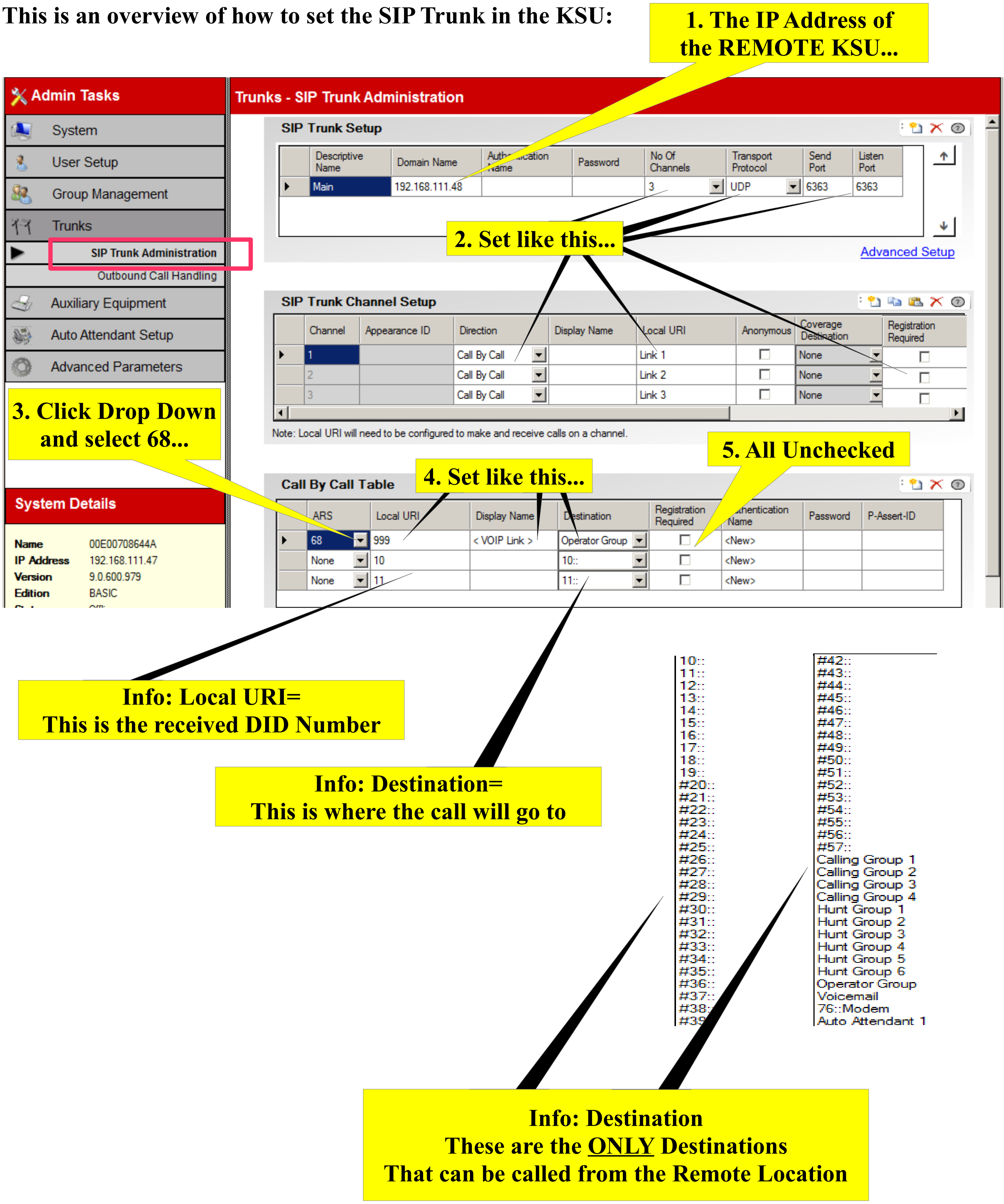
You must give the KSU a Static LAN IP Address.

Set the Default gateway

Set the DNS Server IP Addresses



This is an overview of how to set the SIP Trunk in the KSU:

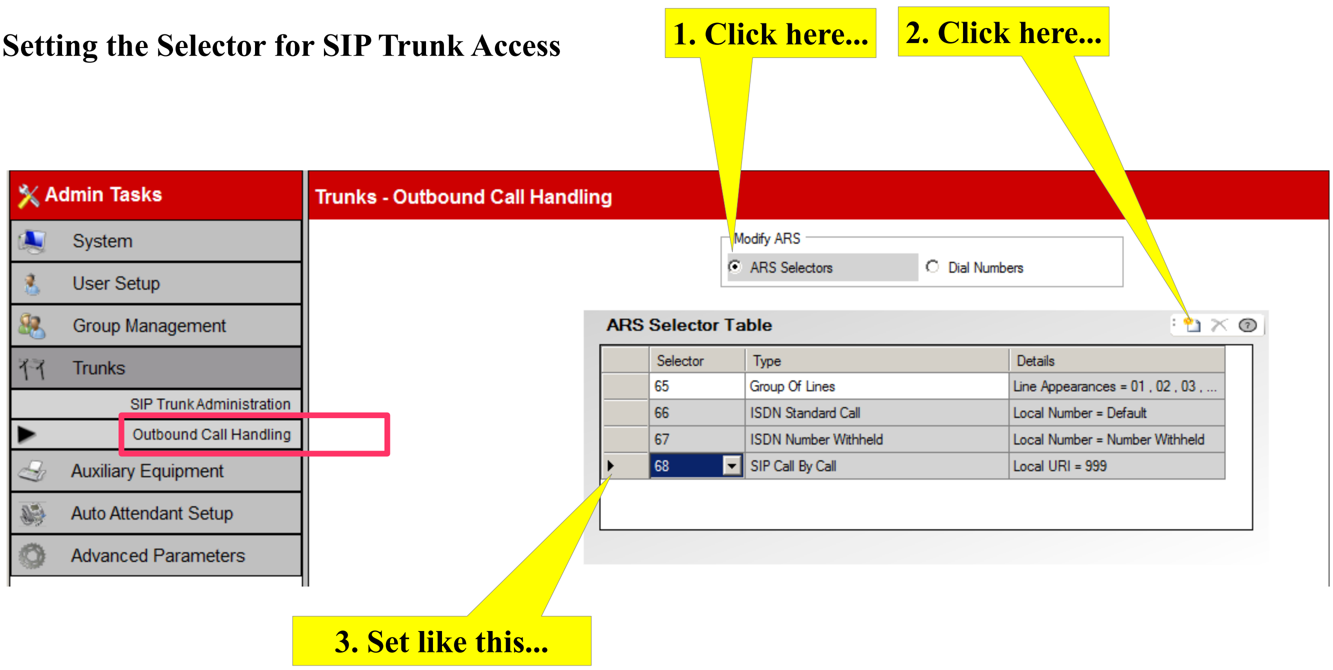


Making calls to the remote location

You must use an ARS Selector to use the Voip Link.

You cannot dial the Selector number
You must access it via a button

Setting the Selector for SIP Trunk Access



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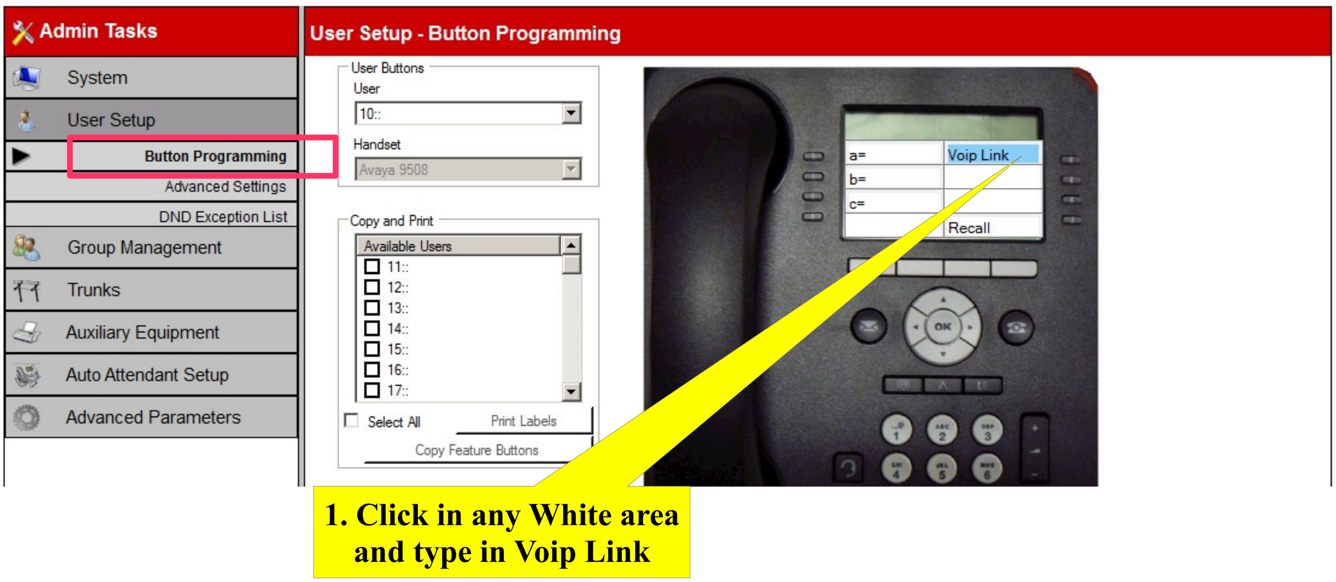
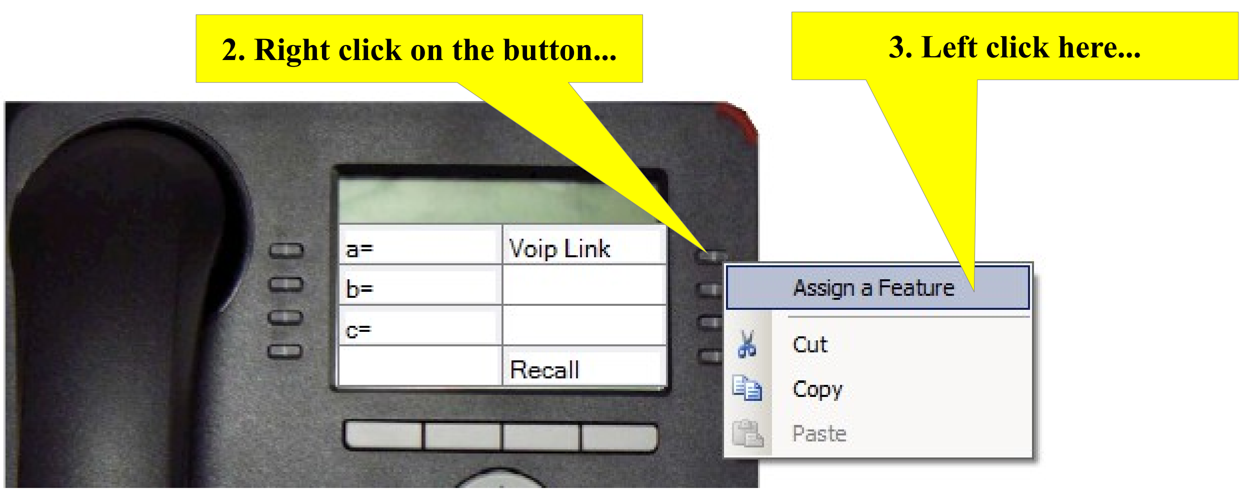
Put Selector number 68 on a button

There are 2 different types of phones that can be used on the Basic Mode, 95xx and 14xx type.

The method is the same, the pictures are different.





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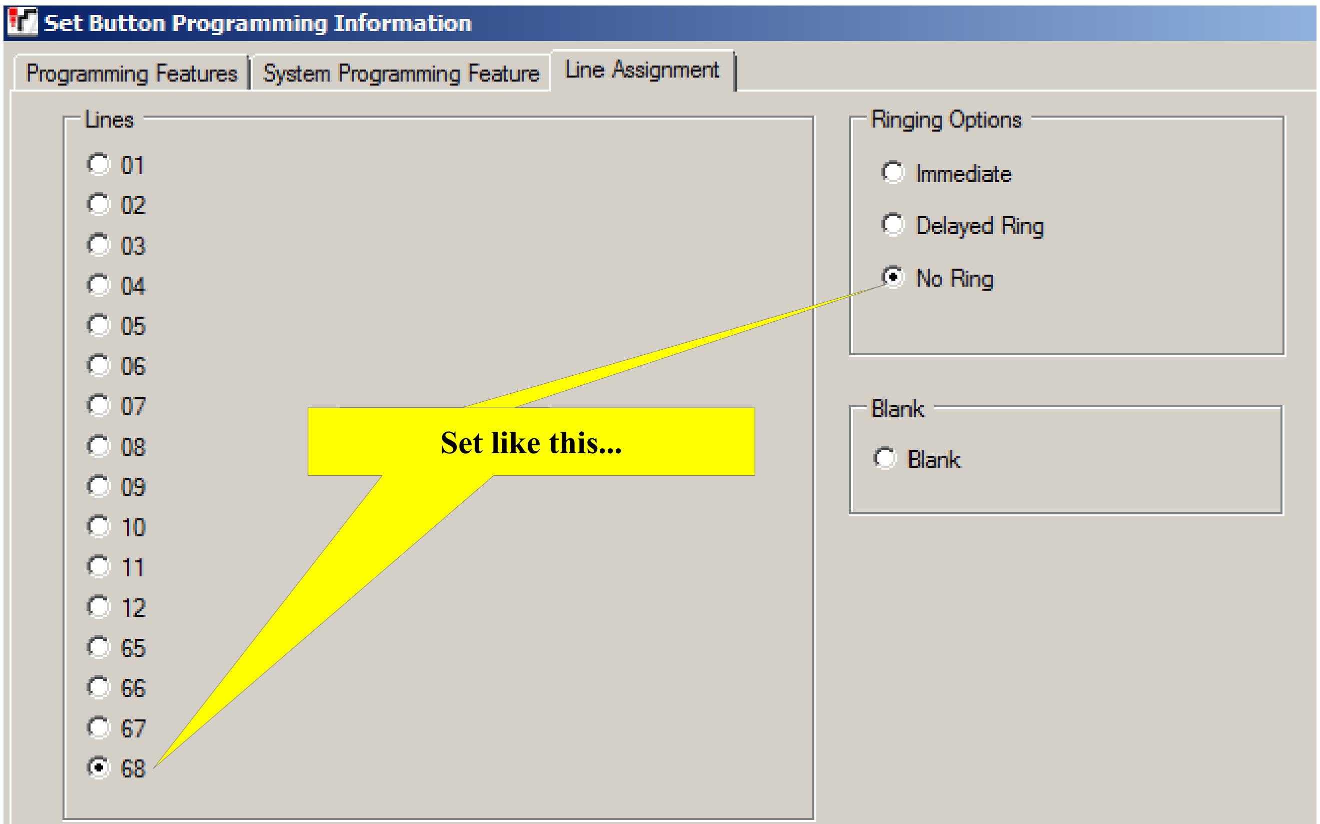
Put Selector number 68 on a button continued

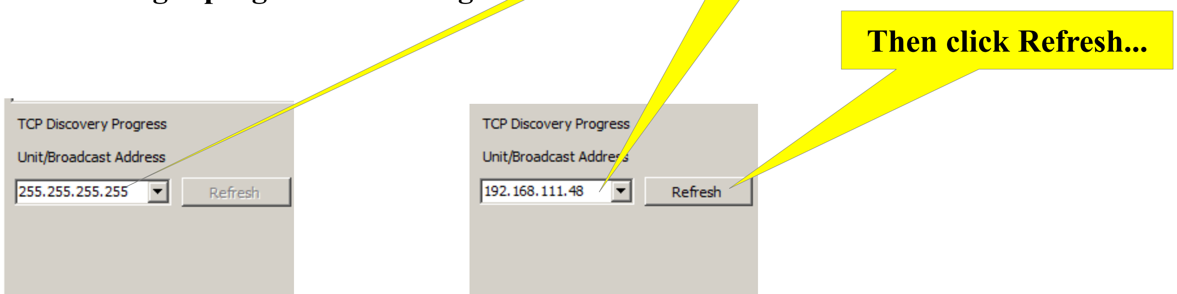


Operation

Be sure that the VPN that you set up has:
ALL Firewalls and Anti Virus Protections turned off
Stateful Packet Inspection (SPI) turned off.
SIP Application Layer Gateway (ALG) turned off.
All SIP and H323 Transformations/Fixups turned off
All anything to do with SIP or H323 turned off
All TCP and UDP ports open

For test purposes, you should be able to connect to the KSU at the far end via the VPN.

Run the Manager program and change this to be the IP Address of the remote KSU.



Calling from a Digital Telephone:

To call an Extension at the far end, press the "Voip Link" button that you created on Page 5.
You should hear a Dial Tone.
Dial the Extension that you want plus a # (POUND)
To reach Extension 10, you dial 10#
The # (POUND) makes the call go through faster.
It is like pressing the SEND button on a cell phone.
There is No Caller ID on the Voip Link.
All calls from one system to the other will show < Voip Link > and 999 as the Caller ID

Calling from a Single Line Analog Telephone:

To call an Extension at the far end, dial 868.
You should hear a Dial Tone.
Dial the Extension that you want plus a # (POUND)
To reach Extension 10, you dial 10#
The # (POUND) makes the call go through faster.
It is like pressing the SEND button on a cell phone.
There is No Caller ID on the Voip Link.
All calls from one system to the other will show < Voip Link > and 999 as the Caller ID

Transferring a caller from one system to the other.

Answer the Incoming Call

Press the TRANSFER button

Press the Voip Link button

Dial the extension number of the remote phone that the caller needs to be transferred to.

Now you can either wait for the extension to answer and announce the call or just hang up for the transfer to complete.

If the extension does not answer, the caller will be sent to that extensions mailbox. (if enabled)

Special Note:
Because of certain SIP limitations in the IP Office Basic Mode, it is not possible to transfer that caller back to the original system.

You can call or transfer calls to anywhere listed on Page 3.

Just remember, you must create a "Local URI" number in the Call by Call Table to reach that Destination.

If you don't, you will get an "Incompatible" message when calling.

As mentioned on Page 1:
Due to the Basic Mode system, there are some limitations on what can and cannot be done.
This is a free configuration that is intended to be used to call and transfer callers between two systems.

If it falls short of your needs, then you need to consider the Standard Mode.

This is not a complete fool proof configuration.
The official statement is that "you cannot connect Basic Mode Systems together".

This is a configuration that Telquest has developed to help our customers and their End Users make use of the Free SIP Trunk Channels that come with the system in a different way then intended.