

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 **MUST** 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

Avaya IP Office Manager 00E00708644A [9.0.600.979] [Administrator(Unknown) C:\Users\billh\Desktop\AAA IPO Configs\IPO Basic R9-0-6 SIP Link via VPN.cfg]

File Edit View Tools Help

Admin Tasks

- System
- System Setup**
- List Management
- Speed Dial Setup
- License Management
- User Setup
- Group Management
- Trunks
- Auxiliary Equipment
- Auto Attendant Setup
- Advanced Parameters

System - System Setup

Installed Hardware

Number	Type	Name
1	Control Unit	IP 500 V2
2	Internal	COMBO6210/ATM4 V2
3	Internal	PHONE2/PRIS U

System Parameters

System Name: Main Location

System Mode: PBX System

Voicemail Mode: IP Office Mode

File Writer IP Address: 0 . 0 . 0 . 0

Country: United States

Language: English (US)

Receive IP Address Via DHCP Server: ☐

IP Address (LAN1): 192 . 168 . 111 . 47

Sub-Net Mask (LAN1): 255 . 255 . 255 . 0

Default Gateway: 192 . 168 . 111 . 1

DNS Server IP Address: 8 . 8 . 8 . 8

Backup DNS Server IP Address: 8 . 8 . 4 . 4

Automatic Daylight Saving Time: ☒

Outside Line: 9 (Operator Is 0)

System Password:

Log All Caller ID Calls for Users: None None None

☐ Unsupervised Analog Trunk Disconnect Handling

What can I do from here?

Change selected system parameters which apply to all users and calls.

System Details

Name: 00E00708644A

IP Address: 192.168.111.47

Version: 9.0.600.979

Edition: BASIC

Status: Offline

Feature Key: Local 1365033086

Ready

Apply Cancel

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

1. The IP Address of the REMOTE KSU...

Admin Tasks

- System
- User Setup
- Group Management
- Trunks
- SIP Trunk Administration**
- Outbound Call Handling
- Auxiliary Equipment
- Auto Attendant Setup
- Advanced Parameters

System Details

Name: 00E00708644A
 IP Address: 192.168.111.47
 Version: 9.0.600.979
 Edition: BASIC

Trunks - SIP Trunk Administration

SIP Trunk Setup

Descriptive Name	Domain Name	Authentication Name	Password	No Of Channels	Transport Protocol	Send Port	Listen Port
Main	192.168.111.48			3	UDP	6363	6363

SIP Trunk Channel Setup

Channel	Appearance ID	Direction	Display Name	Local URI	Anonymous	Coverage Destination	Registration Required
1		Call By Call		Link 1	<input type="checkbox"/>	None	<input type="checkbox"/>
2		Call By Call		Link 2	<input type="checkbox"/>	None	<input type="checkbox"/>
3		Call By Call		Link 3	<input type="checkbox"/>	None	<input type="checkbox"/>

Note: Local URI will need to be configured to make and receive calls on a channel.

Call By Call Table

ARS	Local URI	Display Name	Destination	Registration Required	Authentication Name	Password	P-Assert-ID
68	999	< VOIP Link >	Operator Group	<input type="checkbox"/>	<New>		
None	10		10::	<input type="checkbox"/>	<New>		
None	11		11::	<input type="checkbox"/>	<New>		

2. Set like this...

3. Click Drop Down and select 68...

4. Set like this...

5. All Unchecked

Info: Local URI=
This is the received DID Number

Info: Destination=
This is where the call will go to

10::
11::
#42::
#43::
#44::
#45::
#46::
#47::
#48::
#49::
#50::
#51::
#52::
#53::
#54::
#55::
#56::
#57::
#26:: Calling Group 1
#27:: Calling Group 2
#28:: Calling Group 3
#29:: Calling Group 4
#30:: Hunt Group 1
#31:: Hunt Group 2
#32:: Hunt Group 3
#33:: Hunt Group 4
#34:: Hunt Group 5
#35:: Hunt Group 6
#36:: Operator Group
#37:: Voicemail
#38:: 76::Modem
#39:: Auto Attendant 1

Info: Destination
These are the ONLY Destinations
That can be called from the Remote Location

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

1. Click here...

2. Click here...

3. Set like this...

The screenshot shows the 'SIP Trunk Administration' window with the 'Outbound Call Handling' tab selected. A yellow callout points to the 'ARS Selectors' radio button in the 'Modify ARS' section. Another yellow callout points to the 'ARS Selector Table' window, which displays a table of ARS selectors. A third yellow callout points to the '68' selector in the table, which is highlighted with a blue background.

Selector	Type	Details
65	Group Of Lines	Line Appearances = 01 , 02 , 03 , ...
66	ISDN Standard Call	Local Number = Default
67	ISDN Number Withheld	Local Number = Number Withheld
68	SIP Call By Call	Local URI = 999

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.

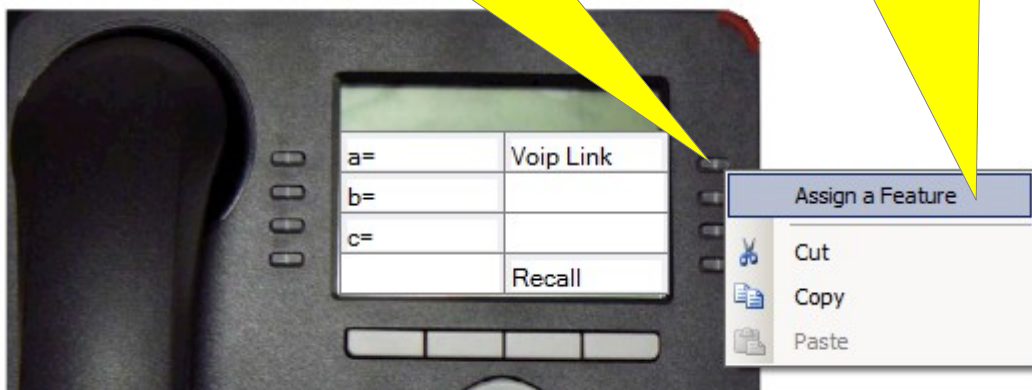
The method is the same, the pictures are different.



1. Click in any White area and type in Voip Link

2. Right click on the button...

3. Left click here...



Put Selector number 68 on a button continued

Set Button Programming Information

Programming Features | System Programming Feature | **Line Assignment**

Lines

- ☐ 01
- ☐ 02
- ☐ 03
- ☐ 04
- ☐ 05
- ☐ 06
- ☐ 07
- ☐ 08
- ☐ 09
- ☐ 10
- ☐ 11
- ☐ 12
- ☐ 65
- ☐ 66
- ☐ 67
- ☒ 68

Ringing Options

- ☐ Immediate
- ☐ Delayed Ring
- ☒ No Ring

Blank

- ☐ Blank

Set like this...

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.

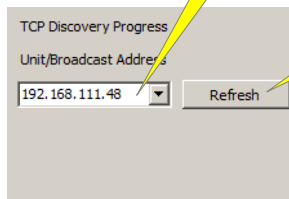
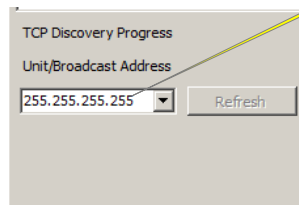
All SIP and H323 Transformations/Fixups turned off

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.



Then click Refresh...

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.