

## Avaya IP Office Basic/Partner Mode SIP Trunk Setup with a SIP Gateway Telquest Tech Support

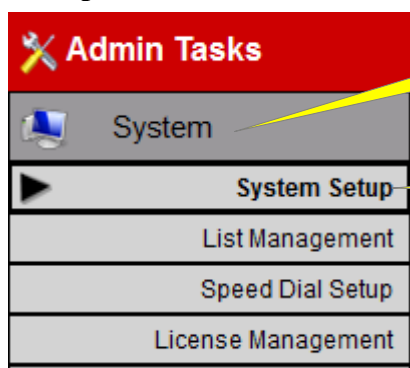
This Help Sheet will show you how to program the Avaya IP Office Basic / Partner Mode KSU to allow a SIP Gateway to be connected.

A SIP Gateway is a device that allows you to access SIP Service (Lines/Channels) locally on your LAN and not over the Internet. Make sure the SIP Gateway has valid LAN settings for your network.

A SIP Gateway **USUALLY** does not require SIP Registration from the KSU.

Optonline and other SIP Service Providers use such devices and may call them by a different name.

### Set up the KSU on the Network



System Parameters	
System Name	Partner Key
System Mode	Key System
Country	United States
Language	English (US)
Receive IP Address Via DHCP Server	<input type="checkbox"/>
IP Address (LAN1)	192 . 168 . 111 . 234
Sub-Net Mask (LAN1)	255 . 255 . 255 . 0
Default Gateway	192 . 168 . 111 . 1

1. Click here...

2. Click here...

3. This should not be checked...

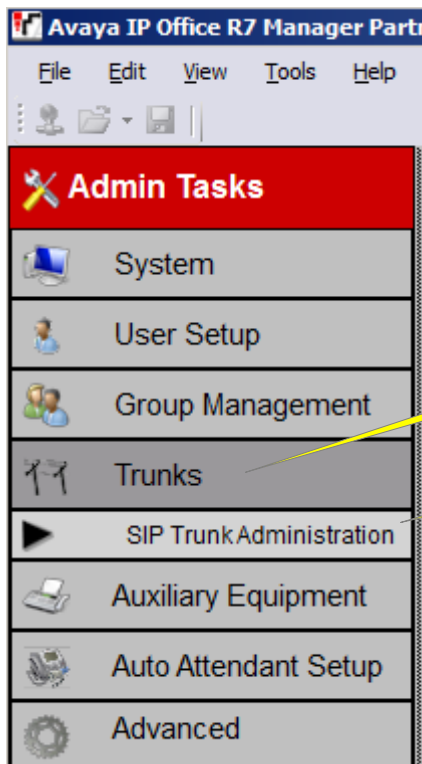
4. A valid IP Address on the LAN...  
Or one that is in the range of the  
SIP Services Gateway Box

5. A valid Subnet Mask for the LAN...  
Or one that is in the range of the  
SIP Services Gateway Box

Note: If you see 192.168.42.1 here  
Then check the "Receive IP" box  
Save Configuration  
And select "Immediate"  
Instead of "Merge"

6. The Default gateway on the LAN...  
Or the IP Address of the  
SIP Services Gateway Box

## Configure the SIP Trunk Channels



1. Click here...

2. Click here...

3. Name the SIP Gateway

4. The IP Address of the SIP Gateway or Remote Service

The KSU will accept SIP Messages from anywhere, it will only process SIP Messages from this IP Address

SIP Trunk Setup

	Descriptive Name	Domain Name	Authentication Name	Password	No Of Channels	Transport Protocol	Send Port	Listen Port
▶	To Edgemark	192.168.111.116			8	UDP	5060	5060

[Advanced Setup](#)

5. Leave blank or KSU will try to Register...

6. Set this to the number of Channels provided by The SIP Gateway...

Note: Be sure you have enough SIP Channel Licenses And VCM Channels...

7. Note – Yours may be different  
This is the button for this Channel.  
Ringing is assigned automatically...

8. Put the Customers Name here...

9. Put the Main Telephone  
Number here on  
EACH Channel...

10. Leave it at None...

**SIP Trunk Channel Setup**

	Channel	Appearance ID	Direction	Display Name	Local URI	Anonymous	Coverage Destination	Unique Line Ringing
▶	1	5	Bothway ▼	Jones & Jones	12125551234	<input type="checkbox"/>	None ▼	Pattern 1 ▼
	2	6	Bothway ▼	Jones & Jones	12125551234	<input type="checkbox"/>	None ▼	Pattern 1 ▼
	3	7	Bothway ▼	Jones & Jones	12125551234	<input type="checkbox"/>	None ▼	Pattern 1 ▼

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

11. Scroll over to see  
This view...

12. UNCHECK for  
EACH Channel...

13. Not needed...

**SIP Trunk Channel Setup**

	Local URI	Anonymous	Coverage Destination	Unique Line Ringing	Registration Required	Authentication Name	Password	P-Assert-ID
	12125551234	<input type="checkbox"/>	None ▼	Pattern 1 ▼	<input type="checkbox"/>			
	12125551234	<input type="checkbox"/>	None ▼	Pattern 1 ▼	<input type="checkbox"/>			
▶	12125551234	<input type="checkbox"/>	None ▼	Pattern 1 ▼	<input type="checkbox"/>			

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

Note – This is not used...

**Call By Call Table**

	Local URI	Display Name
▶	<New>	<New>

This is where your Outgoing Invites are sent to.  
Leave it blank to send to the same IP Address as the Domain Name.  
Fill it in if you need to send the Invites to a different IP Address.

## Trunks - SIP Trunk Administration

Trunk Name : Voip Innov

Trunk Parameters		VOIP Parameters	
Proxy Server Address	65.121.25.244	Compression mode	Automatic Select
DNS Server Address	0 . 0 . 0 . 0	Call Initiation Timeout	4*
		VOIP Silence Suppression	<input type="checkbox"/>
		Re-invoke Supported	<input type="checkbox"/>

In this area...

Click here...

## SIP Trunk Setup

	Descriptive Name	Domain Name	Authentication Name	Password	No Of Channels	Transp. Protocol	Send Port	Listen Port
▶	To Edgemark	192.168.111.116			8	UDP	5060	5060

[Advanced Setup](#)

If you want an Auto Attendant to answer the call after a number of rings:

## Channel Setup

	Channel	Appearance ID	VMS Delay - Day	VMS Delay - Night	VMS Schedule	VMS Auto Attendant
▶	1	5	2	2	Always	Auto Attendant 1
	3	6	2	2	Never	Auto Attendant 1
	4	7	2	2	Never	Auto Attendant 1

In this area...  
Set the Day/Night Delay --- The Schedule --- The AA to answer with...

## General SIP Information

This Help Sheet is for a simple, single telephone number (DID) set up.

A SIP Trunk can have multiple channels and can also have multiple telephone numbers. (DID's)

You may need to adjust the settings in the KSU to meet your customers needs.

Keep in mind that there is no absolute standard used in SIP Trunks/Channels as there is with regular metallic CO Lines.

Every SIP Service provider likes to use their own terminology.

Since the IP Office Basic/Partner gives each SIP Channel its own “Appearance ID” (CO Line) it is possible to use the SIP Channels the same way we use CO Line Appearances.

You can assign SIP Channels to phones like you would CO Lines.

You can control the ringing on SIP Channels like you would CO Lines.  
Ring immediate, delayed or no ring.

You can control outgoing access on SIP Channels like you would CO Lines.

Just think of each SIP Channel as a CO Line.

If a SIP Channel is assigned to a phone that does not enough buttons and that SIP Channel is also assigned to ring, then the call will come in on an idle Intercom Button.

You can also use the “Dial 9” method to make outgoing calls.  
Just be sure to set the “ALS” (automatic Line Selection) to Intercom 1, Intercom 2 and then each SIP Channel.

Don't forget that the SIP Gateway provided by the SIP Service Provider must be “connected to” and “configured to” match the LAN that your KSU is connected to.

You cannot use the WAN Port on the KSU.

You will need to speak to the SIP Service Provider if you are unsure of how to do this.

Telquest does not provide support on the SIP Providers equipment.

Note: Sometime the SIP Gateway may be called a “Router” by the SIP Service Provider.

## Typical Connection Method

