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

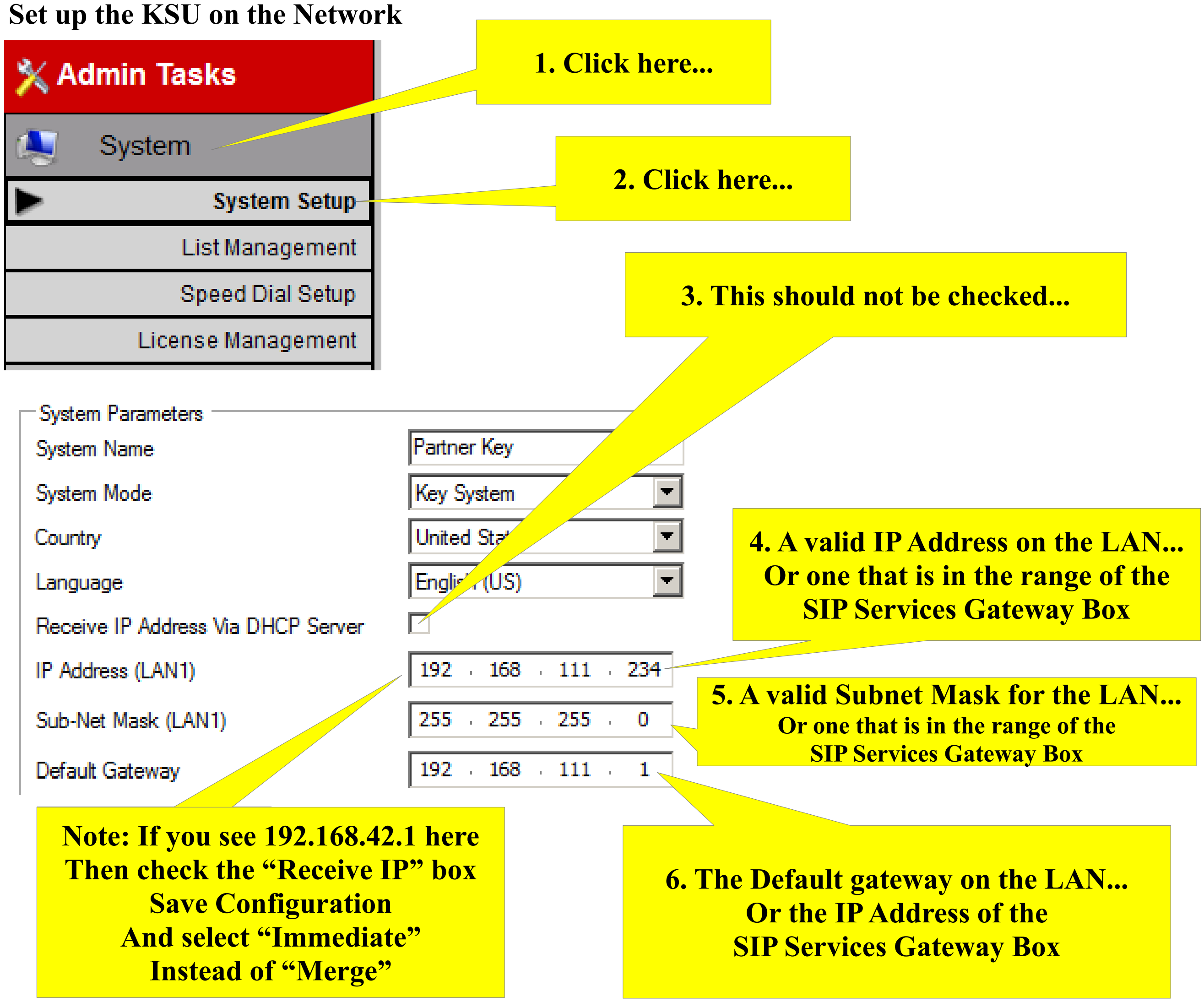
This Help Sheet will show you how to program the Avaya IP Office Basic / Partner Mode KSU to allow a SIP Gateway to be connected.

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.

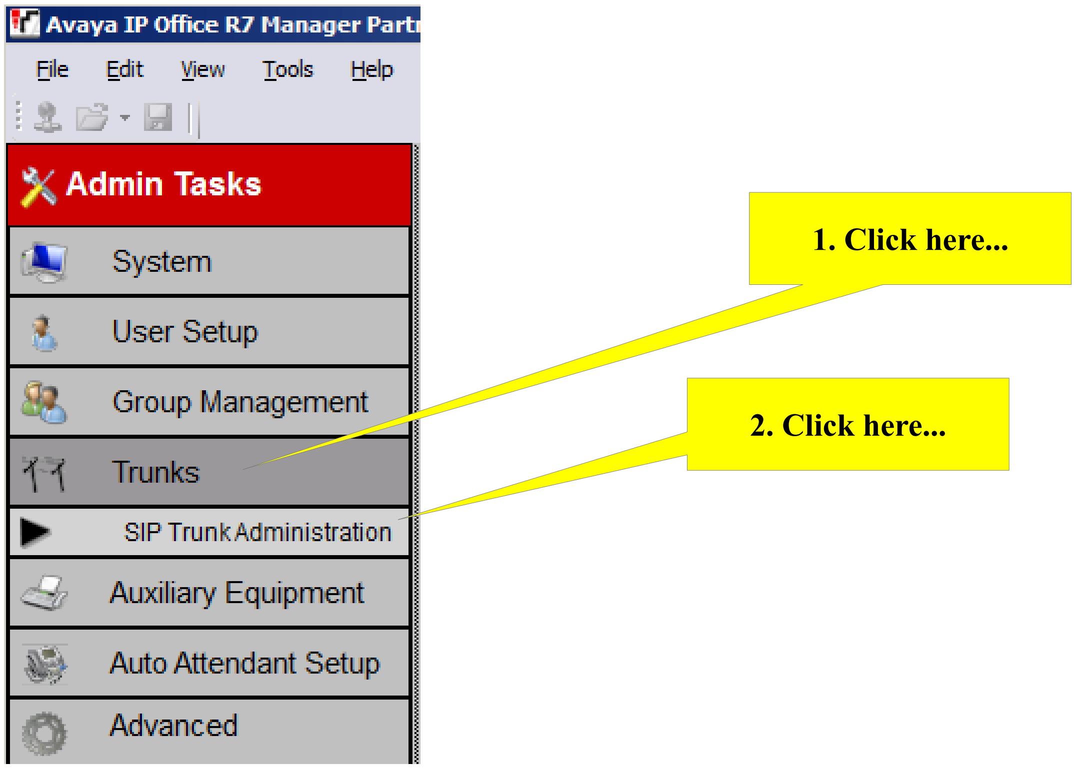
A SIP Gateway **USUALLY** does not require SIP Registration from the KSU.

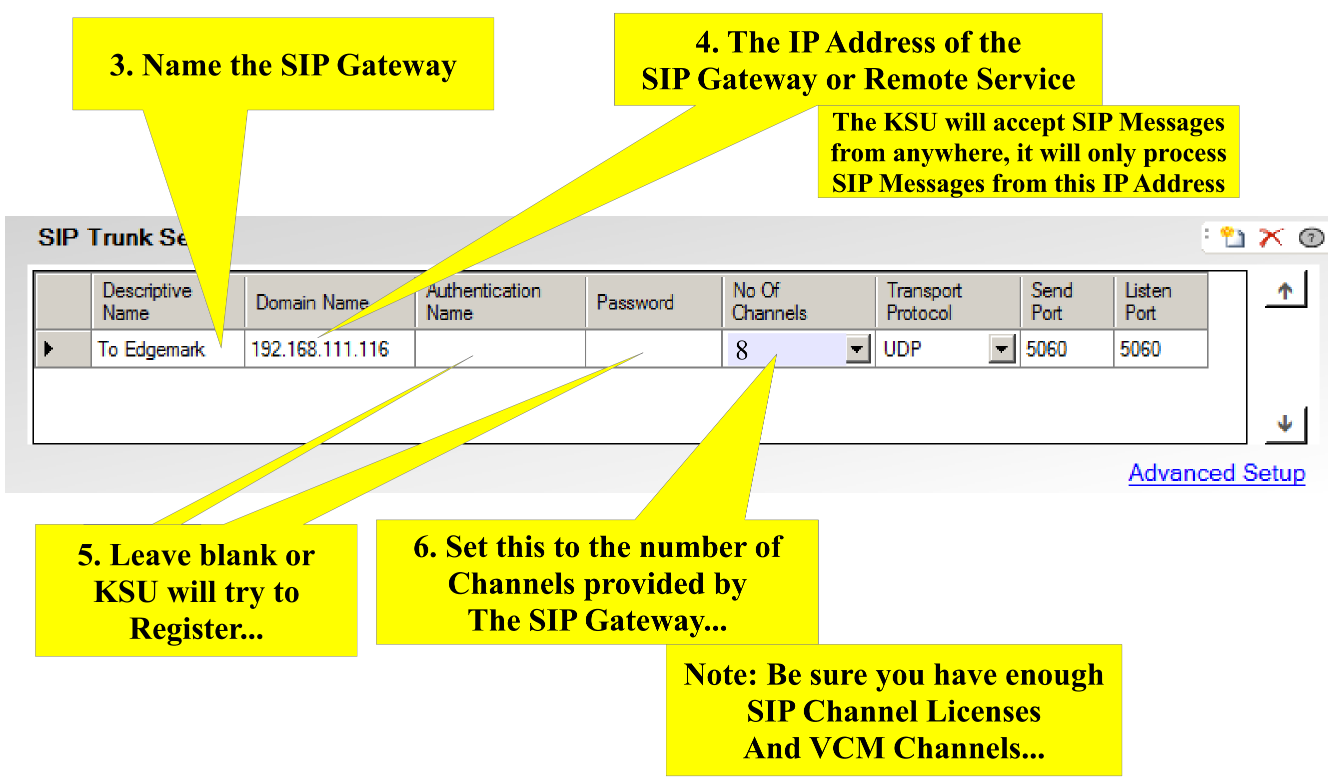
Optonline and other SIP Service Providers use such devices and may call them by a different name.

Set up the KSU on the Network



Configure the SIP Trunk Channels



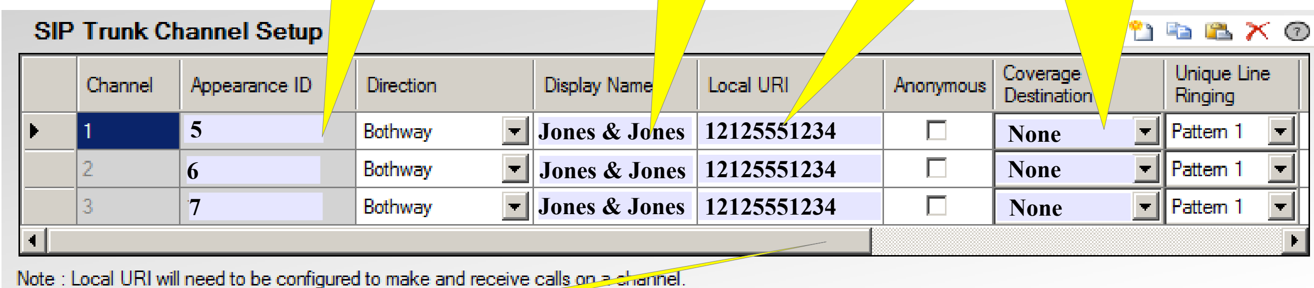


7. Note – Yours may be different This is the button for this Channel. Ringing is assigned automatically...

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

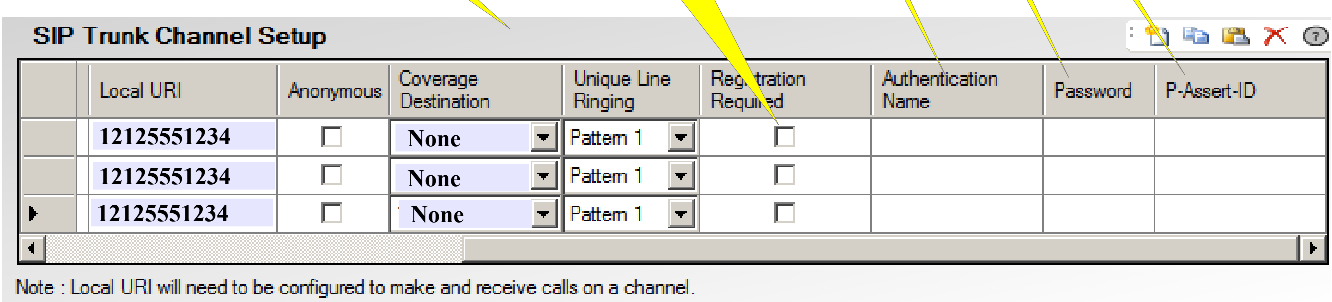
	Channel	Appearance ID	Direction	Display Name	Local URI	Anonymous	Coverage Destination	Unique Line Ringing
▶	1	5	Bothway	Jones & Jones	12125551234	☐	None	Pattern 1
	2	6	Bothway	Jones & Jones	12125551234	☐	None	Pattern 1
	3	7	Bothway	Jones & Jones	12125551234	☐	None	Pattern 1

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...

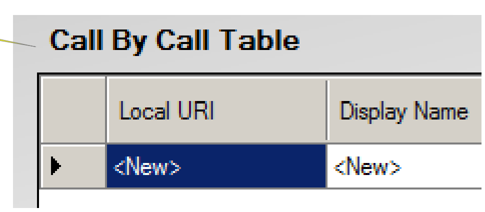


SIP Trunk Channel Setup

	Local URI	Anonymous	Coverage Destination	Unique Line Ringing	Registration Required	Authentication Name	Password	P-Assert-ID
	12125551234	☐	None	Pattern 1	☐			
	12125551234	☐	None	Pattern 1	☐			
▶	12125551234	☐	None	Pattern 1	☐			

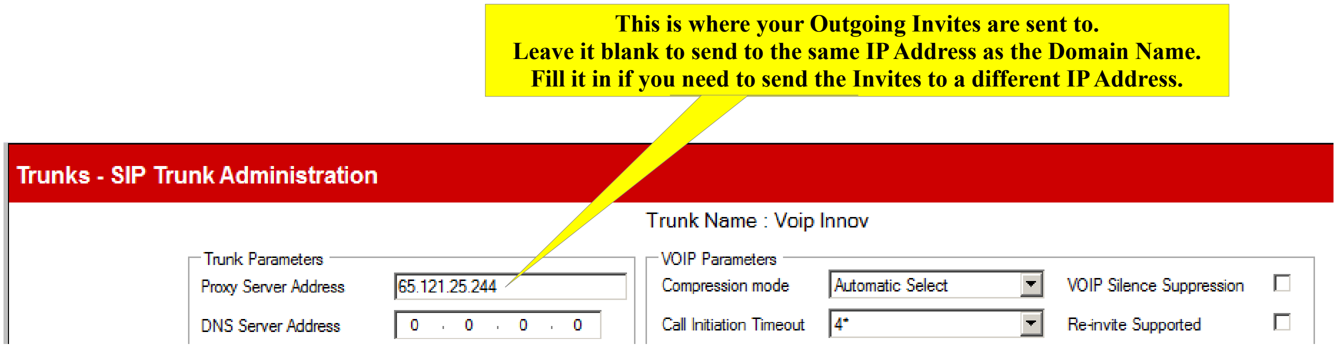
Note : Local URI will need to be configured to make and receive calls on a channel.

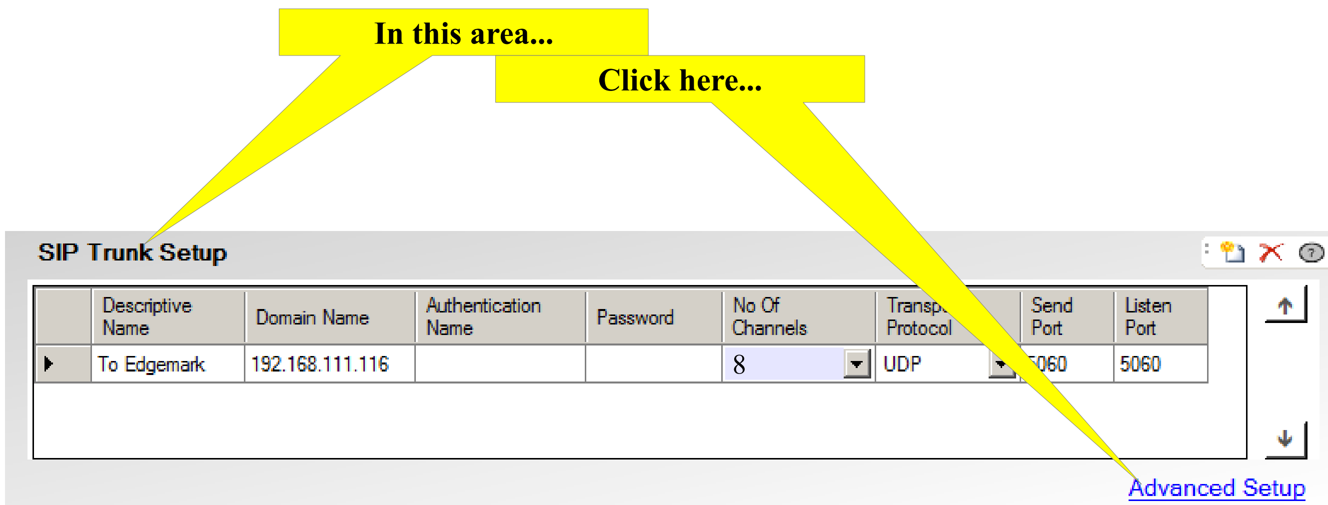
Note – This is not used...



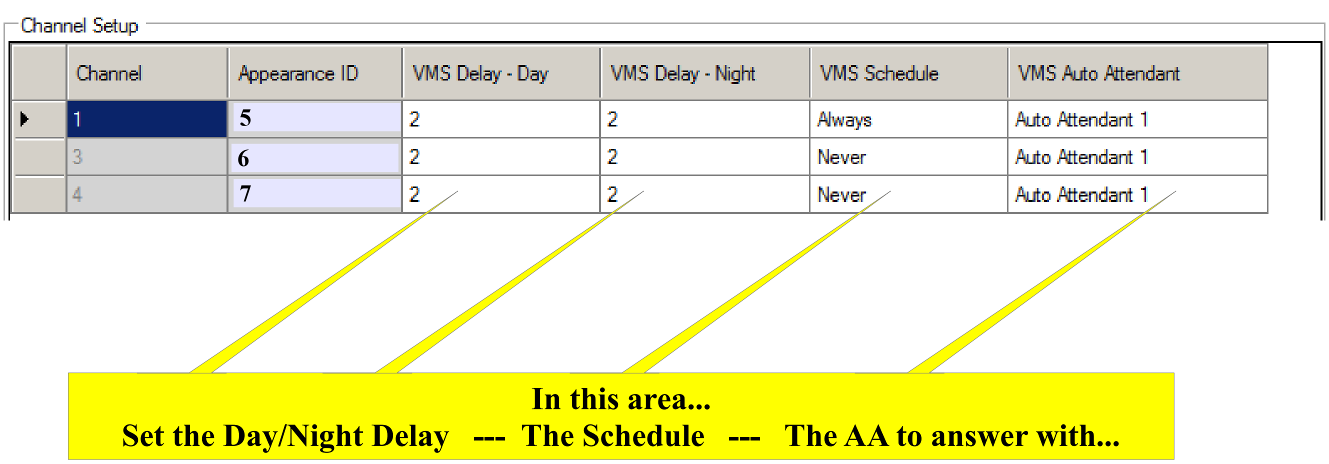
Call By Call Table

	Local URI	Display Name
▶	<New>	<New>





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



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