

## Avaya IP Office Partner Edition

### Transferring a Caller from a Mailbox to a Cell Phone

#### Telquest Tech Support

**You must have at least 1 Combo Card and 1 ETR Card.**

**You must have either a PRI or SIP Trunks to do this.**

**You MUST use a spare Extension Port on an ETR Card.  
This will not work with Analog Extension Ports on a Combo Card**

Receive IP Address Via DHCP Server ☐ **Uncheck this...**

IP Address (LAN1) 192 . 168 . 111 . 234 **The KSU must have a Static IP Address**

Sub-Net Mask (LAN1) 255 . 255 . 255 . 0 **Fill these in too...**

Default Gateway 192 . 168 . 111 . 1

**This only appears on Release 7.0(36)**

**Turn on Mobile Twinning here...**

Configure User List

	Extension	Name	Language	Ex Directory	Outgoing Call Bar	Twinning	Twinning Number	List M
▶	14		English (US) ▼	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>		None
	15		English (US) ▼	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>		None
	16		English (US) ▼	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>		None
	17		English (US) ▼	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>		None
	18		English (US) ▼	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	12122351234	None
	19		English (US) ▼	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>		None

**Use a Spare Port on an ETR Card ONLY.**

No Dial 9 required

**Then enter your cell phone number.**

Set the Extension that will have its call sent to the cell phone.

**This is the Extension that will have its calls sent to the cell phone**

**User Selection**

Select User **12**

**Advanced Parameters**

Ring Pattern	1*	VMS Cover Ring	3
Abbreviated Ringing	Active*	Intercom Dial Tone	Regular*
Call Coverage Ring	2	Distinctive Ring	Active*
Call Waiting Extension	Not Assigned*	Hotline Alert Number	
Automatic VMS Cover	Assigned*	Privacy Enabled	<input type="checkbox"/>
Transfer Return Extension	None	Override Line Ringing	<input type="checkbox"/>

**Voicemail Settings**

Voicemail Code		<b>DTMF Breakout</b>	
Confirm Voicemail Code		Reception / Breakout (DTMF *0)	
Voicemail Email		Breakout (DTMF *2)	
		Breakout (DTMF *3)	18

**Set the \*3 Breakout DTMF Code to the Analog Extensions Number that you assigned on Page 1. (18 in this example)**

**User Selection**

Select User **18::**

**Advanced Parameters**

Ring Pattern	1*
Abbreviated Ringing	Active*
Call Coverage Ring	2
Call Waiting Extension	Assigned
Automatic VMS Cover	Not Assigned
Transfer Return Extension	None

**On Ext/User 18, turn off the Automatic VMS Cover**

## **Avaya IP Office Partner Edition**

### **Transferring a caller from a mailbox to a cell phone**

#### **Telquest Tech Support**

**In your Mailbox Greeting add this: “you can press STAR 3 to be connected to my cell phone.”**

**When the caller presses \*3, they will be transferred to the ETR Analog Extension which then uses the first SIP Trunk (or PRI Channel) available to call the cell phone.**

**Anyone in you office can dial 18 (in this example) to reach you on your cell phone.**

**Incoming CO Line calls can be transferred to Ext 18 (in this example) by a Receptionist or by an Auto Attendant and they will be sent to your cell phone.**

**Call Flow: Using Extension 12 and 18...**

**A call comes in and is Transferred to Extension 12.**

**The call is not answered by Extension 12 so the caller goes into Extension 12's mailbox.**

**There are instructed to dial \*3 (STAR 3) if they want to be connected to a cell phone.**

**The caller dials \*3.**

**They are transferred to Extension 18 which has Mobile Twining activated to the cell phone number.**

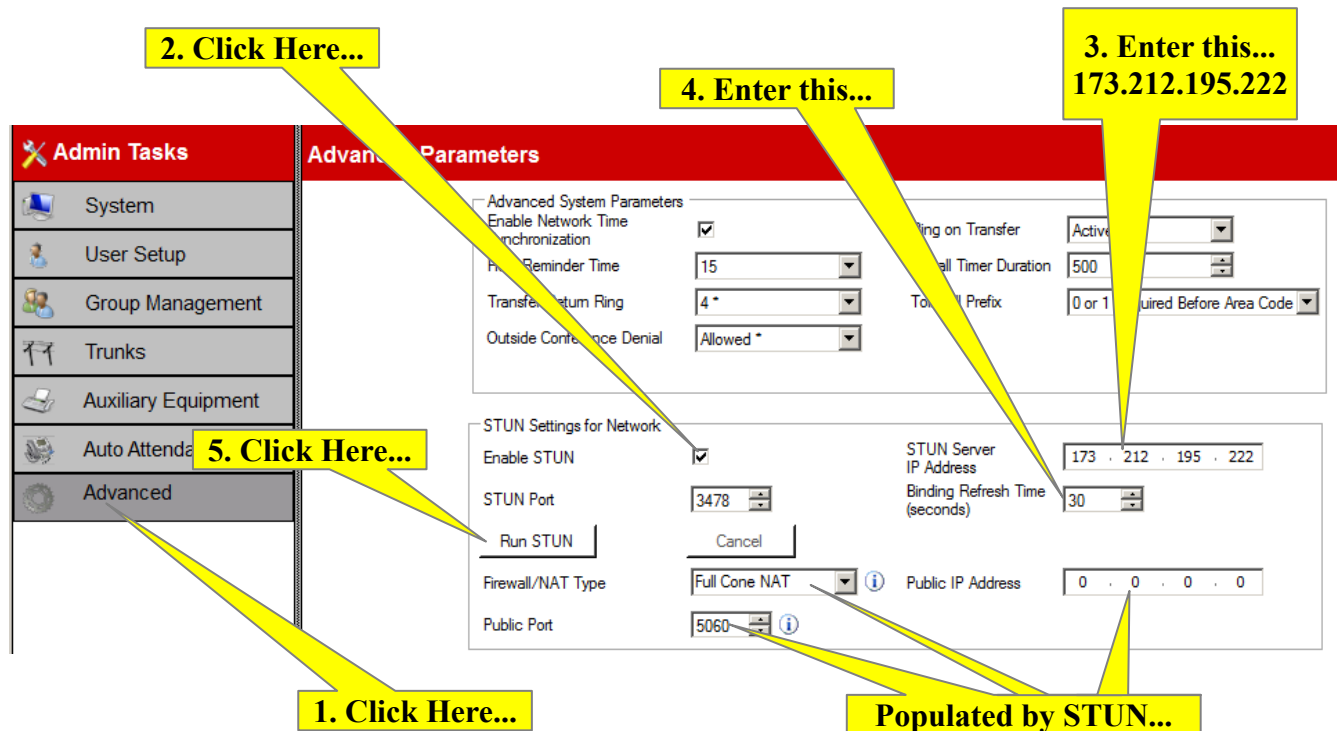
**The KSU uses the SIP Trunk and dials the cell phone number, the call is then connected to the cell phone.**

**Remember, I used Extension 12 and 18 as examples.  
Your Extensions will likely be different.**

## Troubleshooting

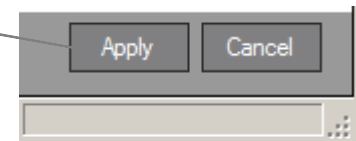
If you have “One Way Audio” then you may need to use a STUN Server to solve the trouble.

**Note:** If you use callcentric.com you are less likely to have “One Way Audio”.



After the KSU finishes communicating with the STUN Server, the above areas will be populated.

When is done, don't forget to click **Apply**, the Blue Floppy and then the final OK Button to send your changes to the KSU.



**Make a test call.**

If it works then it is fixed, if not then you can try the following:

**Open/Port Forward Ports 49000 UDP through 49500 UDP to the KSU in the network router.**

**Or:**

**Put the KSU in the DMZ of the network router.**

If the STUN Server fails to detect anything, then open a Command Prompt Window and ping `stun3.3cx.com` to get the current IP Address.

The IP Office does not allow you to use a URL like `stun3.3cx.com`, you can only use its IP Address. If the STUN Server changes its IP Address, your STUN will fail.