

SCS 4.0

Troubleshooting

Task Based Guide

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Troubleshooting

Introduction

There may be occasions when users and administrators encounter operational issues with the SCS. Often such issues are caused by minor configuration errors or incorrect usage of a particular function. Normally such issues can be solved by following set procedures to identify the source of the problem and then taking steps to rectify it in the quickest time possible – usually these procedures start with the simplest and most innocuous causes and then work towards more complex issues. This guide aims to provide you with some of the more common issues you may come across when using the SCS and their solutions.

Note: For outage and system recovery related issues please see the SCS 'Disaster Recovery Planning Guide'.

As well as the steps suggested in this guide, you might also consult some of the following diagnostic tools when trying to resolve a problem:

- **Registrations** The Registrations facility provides information on the status of users registered to the SCS.
- DNS Advisor Determines and validates DNS records required for the SCS
- Call Detail Records Call Detail Records show historic call data and can be filtered to provide information based on particular names or dates.
- **Statistics** The Statistics diagnostic can be used to find information on performance-related issues, for example: HDD usage, memory availability, and CPU load.
- Alarms The alarm server collects alarms in the system and sends out notification to system administrators. The advanced section displays all alarm types defined by the system. You can enable/disable e-mail notification on each alarm type.
- **Syslog** The location of a Syslog Server can be defined that is used to obtain Syslog messages from devices connected to the SCS, such as Phones or Gateways that have Syslog enabled. Logs from these devices will be sent to the Syslog server. The logs can then be collected with the SCS snapshot facility.
- ACD Reports Use ACD Reports to generate reports on the ACD queue, including: agent availability, queue activity, and abandoned calls.

- Job Status The job status page shows the status of various processes on the SCS, for example phone profile creation, file replication across a high availability configuration.
- Configuration Tests These are a collection of tests that can be run to test the availability of certain services that support the SCS infrastructure. Available tests include: DHCP server verification, TFTP service verification, local host configuration. You may be asked to run some of these tests in a support call scenario.
- Services The Services utility enables you to stop or restart services, as well as refresh service status. It is recommended that you only access this area if instructed to by a support engineer or documentation.
- **Snapshot** The Snapshot tool is a very useful facility that allows you to create an instant archive of SCS and Apache web server logs and other diagnostic information. Output can be filtered to particular time ranges or user names. Again, you may be requested to create snapshot logs by a support representative.

Note: When a Snapshot is taken in a HA environment, 2 Snapshots are produced one for Master and other for the distributed system

• Login History – Use the Login History tool to create a report showing successful and unsuccessful login attempts.

Scenarios

Problem: Unable to access the SCS GUI

Solutions:

There could potentially be any number of reasons why you are unable to access the SCS GUI, chief among which would be:

 Connectivity between your workstation and the LAN Start simple. Ensure that your workstation or laptop is connected to the LAN. Is the Ethernet cable securely connected to the network card on your workstation

 if using Wi-Fi, make sure that you have a connection to your usual wireless access point.

Once you have ensured that your physical or wireless connection to the LAN is intact try **pinging** the IP address of another device on the network – for example the default gateway or a neighbour's workstation. If the ping is successful then you know your workstation is connected to the LAN, if you still cannot access the SCS GUI, explore the other issues in this guide.

- **IP Address** Ensure that you have the correct IP address and domain name for the SCS your SCS administrator should be able to inform you of the correct details.
- Connectivity between the SCS and the LAN Once you know that your workstation is connected to the network and that it can see other devices, and you are certain of the IP address and SCS domain name, it's time to check whether or not you can contact the SCS. Use Ping again and ping the SCS IP address. If the ping packets are not returned then there is the possibility that the SCS is not connected to the network, in which case you may wish to escalate this issue to the network administrator.

If the ping packets are returned then you know that the SCS is connected to the LAN. Attempt to browse to the SCS GUI using the IP address instead of the Fully Qualified Domain Name (FQDN) in your web browser of choice. If you are able to browse to the SCS using the IP address but not the FQDN then there is a chance that there is a **DNS issue**.

• **DNS issues** In simple terms DNS is used to resolve address names such as *scs.scsdomain.com* to IP addresses, which is why if there is a DNS issue you will probably be able to browse to the SCS GUI using an IP address and not the fully qualified domain name. Not being able to browse to the GUI using the FQDN is a symptom of a larger problem because DNS plays an important roll in the SCS environment. If you suspect a DNS issue escalate the problem to your local network administrator.

- Internet browser restriction Some internet browsers can be configured to connect to the Internet through a proxy. This can cause some issues when attempting to connect to the SCS GUI. Proxy settings can be disabled within IE and Firefox in the following way (check with your network administrator that it is OK to disable proxy settings first):
 - Mozilla Firefox 3.5: Open the Tools menu and select Options. Click on the Advanced tab and select the Network sub-tab. Under the Connection header, click on Settings. Select No Proxy. Click OK to confirm changes and exit the options screen.
 - Microsoft Internet Explorer 7: Open the Tools menu and select Internet Options. Click on the Connections tab and then the LAN settings button. Under the Proxy server heading, untick the Use a proxy server for your LAN check-box. Click the OK button to confirm changes and exit the options screen.



Problem: Cannot create user accounts

Solution:

In order to create user accounts on the SCS, the system must be licensed to support the required number of users. If you are unable to create new accounts then there is a strong chance that either the system has not been licensed or the maximum number of allowed user accounts has been reached. See your SCS administrator or contact Avaya for further licensing options. Also, please see the 'System Configuration' guide for more on licensing the system.



Problem: Incoming calls do not route to SCS user DIDs

Solutions:

Calls can be routed to individual **DIDs** (Direct Inward Dial) in one of two ways, you can either:

1. Configure a dial plan for every user which defines where calls to a certain number route. For example, all calls to 670238 are routed to User ID 238. This is a perfectly valid way of routing calls to individual worker DIDs but a **Custom Dial Plan** will need to be created for **every** worker.

If you find that this solution is not working, ensure that:

- All plans have the correct **Dialled Number Prefix** (that is the number that callers will dial to reach the user) entered – you can check these with your service provider.
- The **Resulting Call** field is populated with the correct user ID/extension.
- All plans have been **enabled**.

For more on Dial Plans see the SCS 'System Configuration' guide.

2. Define an **Alias** for every user that is made up of their DID. This approach eliminates the need to create a separate dial plan for every worker in the system.

All user accounts must have a User ID. This is a unique identifier that marks the user out from everybody else on the system, it can be a number (a three digit extension for example) or a name (johns). As well as the User ID, you can define **Aliases**, which are effectively another name by which the user is known on the system. Aliases can be anything you like but must be unique. To facilitate direct inward dialling you could assign every user their DID as an alias; that way, when a call comes in to those numbers the SCS will know exactly where to send the call. For more on Aliases, please see the 'Configuring User Profiles' guide.



Problem: Managed IP phones (phones automatically provisioned and managed by the SCS) do not register

Solutions:

If you are unable to register a managed phone to the SCS then the problem lies either with the SCS or the phone itself.

- 1. On the SCS check the following:
- Has a **Phone Profile** for the phone in question been created?
- Has a user account been added as a Line to the phone profile?
- Has the profile been **sent**?

Before a managed phone can register to the SCS a profile must first be created on the SCS. This profile contains information about the phone and the user who will be using the phone (described in the profile as a 'line'. The profile must be 'sent' to the SCS TFTP server so that it can be collected by the phone device when it boots on the network.

For more on creating phone profiles, please see the 'Configuring User Profiles' task based guide.

- 2. In order for the phone to successfully gather its profile information, certain phone and/or network settings must be configured:
 - DHCP Option 66 can be configured to 'point' phones at the SCS for profile collection. This is not mandatory but can make configuring a phone quick and easy. If Option 66 is not utilized you will have to configure the IP address of the SCS as the TFTP value on the phone manually when you configure other LAN settings.
 - Phone LAN Settings. Either enable DHCP to allow automatic configuration LAN settings, or configure LAN settings manually. Once LAN settings have been configured, set the TFTP address if Option 66 is not in use, this is the IP address of the SCS.



For more on configuring settings on a managed phone see the Configuration Task Based Guide appropriate to your phone model.

Problem: Users cannot access voicemail

Solutions:

There are four possible reasons why a user would not be able to access their voicemail inbox:

- They are dialling the wrong voicemail extension number. Check the number that has been configured in the **Voicemail Dial Plan**. See the 'Voicemail Setup and Operation' guide for more on this.
- They are using the wrong PIN. Check the PIN.
- They do not have permission to access voicemail. Ensure that the **Voicemail** permission is checked on their user profile permissions page see the 'Configuring User Profiles' guide for more.
- The user account has been configured with the wrong voicemail service, e.g., Internal voicemail service instead of Microsoft Exchange UM Voicemail service. See the 'Voicemail Setup and Operation' guide for help on selecting a voicemail service.



Problem: ACD does not function

Solutions:

There are several configuration details that you should check:

- Check that the ACD server role has not been disabled on the SCS.
- Define ACD queues. The ACD queue determines how incoming calls are handled and potentially queued.
- Assign agents to queues to ensure that agents can log in and receive calls. If there is no queue then agents cannot field calls.
- Check that agents are signed in to the ACD queue.
- Ensure that a dial plans exist to 'point' to the correct 'ACD Line'. An ACD line is a virtual line that defines how calls reach the ACD queue. ACD lines, like user extensions, have an identifier (for example, 500) or a DID number (for example, 200786). You therefore assign an ACD line to a queue in the same way that you would assign an extension to a user in order to direct calls to them.
- Check that the sign in and out codes used by agents is correct. The default sign in code is ***88**, the default sign out code is ***86**.
- Ensure that the ACD is 'active'. Just because the ACD server is present it does not necessarily follow that it has been activated. If the server is not active the ACD will not function.

For more help on the ACD please see the 'ACD Setup and Operation' guide.



Checking the ACD Server Role is Enabled

The ACD server role required for the ACD to function, is enabled by default on the SCS server. However, before commencing with ACD configuration, it is good practice to check the status of the server role to ensure the role has not been disabled.

To view the status of the ACD Server Role, navigate to the SCS server:

1. Click the **System** heading, followed by **Servers**.

VAV	4			Tue, 02 Mar 2010 10:20 AM 🛛 🎆 Home 👘 7 Help 👘 Logout 🔍 Search
Users	Devices	Features	System	Diagnostics
lome			Licensing	
101110			Servers 4	
			Branches	
		There are 5 users define	Domain	There are 4 phones defined:
		200, 201, 202, 203, super	Dial Plans	0004f21eea0d, 0004f21f4bc7, 002162ff7032, ed,
		Add New User	Internet Calling	more
			Permissions	Add new phone
			Import / Export	Discover Devices
			LDAP / AD	
			Backup	
			Restore	
			Localization	Communication System (4.1.6-0181.04.2010-02-251.02/57/38)
			Certificates	
			Software Updates	
			Date and Time	
			Logging Levels	

2. You will be presented with the Servers screen. Click the server's link.

VAV	Ά				Tue, 02 Mar 2010 10:23	3 AM 🍈 Home े Help	🔒 Logout 🔍 Search
Users	Devices	Features	System	Diagnostics			
ervers							
	Serv	/ers		/		Add Server	Clicking the <i>Send Profiles</i> button will cause configuration files for all
		Name		IP Address	Description	Status	services to be sent to the selected servers, and all affected services to be
		scs1.scsnetv	work.local	200.30.30.5	Primary server	Registered	restarted automatically. This
	Sei	nd Profiles Del	ete				is rate in iterated as configuration files are sent by default when their associated configuration has been changed. However, in the case where a distributed server was not available at the time of a configuration change, this button can be used to ne-send the configuration.

3. The Server screen will be displayed. Click the Server Roles link.

AVAYA			Tue, 27 Apr 2010	3:56 PM 💮 Home	e 7 Help Cogout	Search
	Users	Devices	Features	System	Diagnostics	
Server						
Configure Server Roles Services NAT Monitor	Servers > scsitel2.it ✓ ACD ✓ SIP Trunking ✓ Conferencing ✓ Instant Mess ✓ Managemen ✓ Primary SIP I ✓ Voicemail OK As	eluk.com aging t Router pply Cancel				One or more roles can be enabled on each server. All roles can run on one single server or the different roles can be distributed to several servers forming a cluster. A high availability configuration can be configured by enabling a redundant SiP router role. Roles can be moved to dedicated servers to improve performance.

4. Within the grey panel, there are a number of checkboxes defining the server roles. The **ACD** check box should be selected by default.

AVAVA			Tue, 27 Apr 2010	3:57 PM Hom	ne 7 Help 🕆 Lo	gout Search
	Users	Devices	Features	System	Diagnostics	
Server						
Configure Services NAT Monitor	Servers	teluk.com g saging t Router pply Cancel				One or more roles can be enabled on each server. All roles can run on one single server or the different roles can be distributed to several servers forming a cluster. A high availability configuration can be configured by enabling a redundant SIP router role. Roles can be moved to dedicated servers to improve performance.

5. If it is not selected, click the **ACD** check box followed by the **Apply** button and the **OK** button.

AVAVA			Tue, 27 Apr 2010	3:59 PM 💮 Hom	e 7 Help 🔒 Log	gout Search
	Users	Devices	Features	System	Diagnostics	
Server						
Configure Server Roles Services NAT Monitor	Servers > scsitel2.it ✓ ACD ✓ SIP Trunking ✓ Conferencing ✓ Instant Mess ✓ Managemen ✓ Primary SIP ✓ Voicemail ✓ OK Ag	eluk.com g aging t Router				One or more roles can be enabled on each server. All roles can run on one single server or the different roles can be distributed to several servers forming a cluster. A high availability configured by enabling a redundant SIP router role. Roles can be moved to dedicated servers to improve performance.

6. If changes have been made, you will be prompted to restart services. Click the **here** link.

sers	Devices	Features	System	Diagnostics			
rs							
							1
	One o	r more services need Vers	I to be restarted. Fo	r details click: <u>here</u>			
						Add Server	Clicking the Send Profiles button will cause
							services to be sent to the
		Name		IP Address	Description	Status	affected services to be
	E	scs1.scsnetv	vork.local	200.30.30.5	Primary server	Registered	restarted automatically. This
	Se	nd Profiles Del	ete				configuration files are sent by
							default when their associated configuration has
							been changed. However, in
							server was not available at
							the time of a configuration change this button can be
							used to re-send the
							configuration.

7. You will be presented with the **Affected Services** screen. Select the services and click **Restart**.

AVAVA	7				Tue, 02 Mar 2010 10:43 AM	ight Home ? He	elp 📄 🔒 Logout 🔍 Search
Users	Devices	Features	System	Diagnostics			
Affected Service	es						
	Affe	cted Servic	es				Quick Links
			Server		Service		Servers
	Re	start Ignore					Restarting services may cause service interruption.
			Software	Communication System	(4:1.6-018104 2010-02-25T02:57:38)		

8. The affected services will be restarted.

AVAVA	1				Tue, 02 Mar 2010 10:43 AM	- Home	? Help	Logout O Search
Users	Devices	Features	System	Diagnostics				
Affected Service								
	Affe	ected Service	es					Quick Links
			Server		Service			Servers
	Re	start						Restarting services may cause service interruption.
			Software	Communication System	n (4.1.6-018104 2010-02-25T02:57:38)			

9. To check that the ACD services are running, navigate to the **Servers** page and click the link for the SCS server.

VAV	Δ				Tue, 02 Mar 2010 10:23	AM 🧄 Home े Help	🔒 Logout 🔍 Search
Users	Devices	Features	System	Diagnostics			
	Ser	vers		/		Add Server	Clicking the Send Profiles button will cause configuration files for all
		Name		IP Address	Description	Status	services to be sent to the selected servers, and all
] 🕘 <u>scs1.scsnetw</u>	ork.local	200.30.30.5	Primary server	Registered	affected services to be restarted automatically. This
	Se	nd Profiles Dele	ste				is rately needed as configuration files are sent by default when their associated configuration has been changed. However, in the case where a distributed server was not available at the time of a configuration change, this button can be used to re-send the configuration.
			Softwar	e Communication System (4.1 &	018104 2010-02-25702-57-38	0	

	lisers	Devices	Features	System	Diagnostics			
	03613	Devices	reatures	System	Diagnostics			
Servers + scsite	2.iteluk.com				-			
_				Refresh every 30 seconds Warning: Restarting				
Name	Name Name	Status			Role	interruption. Do it only if a		
Park		Running			Primary SIP Router	service requires restart or not working properly. In		
🔲 🌼 Instant I	Messaging	Running			Instant Messaging	such case you might want to take a snapshot and		
Call Co	ntrol	Running			Management, Primary SIP Router	report an issue.		
MyBudo	<u>yt</u>	Running			Instant Messaging	This page will refresh		
🔲 🧼 Statistic	s	🕝 Running			Management	switch automatic refreshin		
Configu	iration	Running			Management	checkbox.		
ACD		📀 Running			ACD	refresh interval by clicking		
Dependence Phone P	Provisioning	📀 Running			Management	 on the current interval and then enter a new value. 		
ACD Ag	ent Status	Running			ACD			
🔲 🎲 ACD St	atistics	Running			ACD			
🔲 🎲 Shared	Appearance Agent	🕑 Running			Primary SIP Router			
Media S	Services	Running			Conferencing, Voicemail			
📃 🌼 Licensir	ng	Running			Management			
🔄 🎡 Voicem	ail MWI	Running			Voicemail			
SIP Tru	nking	Running			SIP Trunking			
Paging		🕝 Running			Primary SIP Router			
Media F	Relay	Running			Primary SIP Router			
SIP Reg	gistrar	Running			Primary SIP Router			
		Running			Management			
SIP Pro	xy	Running			Primary SIP Router			
Present	ce	Running			Primary SIP Router			
Confere	ence Recording	Running			Conferencing			
Voicem	ail and Auto Attendant	🔿 Running			Voicemail			

10. Select the **Services** link. The ACD services will be displayed.

Problem: Auto-attendant is not functioning

Solutions:

Auto-attendant issues can range from callers not being directed to the attendant (Operator) to users not being able to dial the attendant internally. If you experience any issues with the auto-attendant, check the following:

- Check that the right **extension number** is referenced in the autoattendant dial plan (the auto-attendant dial plan is called Operator). By default the extension number is **100**. If users are dialling an extension number other than the one referenced in the dial plan they will not be able to access it.
- Ensure that the dial pad selection prompts are mapped to the right keys so that callers and users can access the right service from the right button.
- Callers will only be able to access the auto-attendant if the correct received digits are referenced in the dial plan under **Attendant aliases**.

For full instructions on configuring these options see the 'Auto-Attendant Setup and Configuration' guide.



Problem: Contact names are not recognised by Dial-by-Name

Solution:

In order for contacts to be searchable using Dial-by-Name they must be added to the **Directory**. To be part of the directory a user must have the Directory **Permission** enabled in their user profile. Browse to a user's profile, select Permissions and ensure that the **Directory** option is ticked. SCS Users contained within the Directory must also record their name Refer to SCS 4.0 Voicemail Setup & Operation Task Based Guide.

Problem: Cannot 'Park' calls

Solution:

Check that call park **extensions** have been allocated and that the Call Park Service has been restarted.

Problem: Unable to retrieve parked calls

Solution:

Ensure that a **call park retrieval code** (*4 by default) been configured and also check that users are entering the right park extension number after the park retrieval code. This can be made more simple by configuring a speed dial of *4xxx on a user's phone – where xxx is the park extension - , the user could then retrieve calls simply by pressing the speed dial button.

For more on Call Park see the 'System Configuration' guide.

Problem: Unable to Page

Solutions:

Check the following:

- Are users dialling the correct Paging Prefix?
- Have Page Groups been defined and have users been added to page groups?
- Has the page group been Activated? An 'enabled' page group is not an 'activated' page group. In the page groups list, tick the check-box of the page group in question and then click the Activate button, then follow the pink link at the top of the screen to restart the paging service.

See the 'System Configuration' guide for more on paging.

Problem: Directed Call Pickup not functioning

Solution:

Check call **pickup code** on the **SIP Registrar** screen, accessed from the **Servers** page. In order to pick up a call, users must dial this code followed immediately by the extension number of the phone that is ringing.

See the 'System Configuration' guide for more on call pickup.

Problem: Calls to and from the PSTN are failing

Solutions:

PSTN Gateways are devices that enable calls from the Public Switched Telephone Network (PSTN) to interface with telephone devices on the SCS. If you are unable to make or receive calls from or to the PSTN check the following:

- Has the gateway been programmed with the correct IP address?
- Has the correct IP address been referenced for the gateway on the SCS?
- Have you produced an INI file for the gateway on the SCS and uploaded it to the gateway?
- Have you assigned **PSTN Lines** to the gateway in order to route calls from the PSTN to the SCS and vice-versa.
- Ensure the gateway is physically connected to both the PSTN and the LAN.



See the 'Device Configuration – Gateways' guide for detailed instructions on how to configure a PSTN gateway for use with the SCS.

Problem: Cannot create a conference

Solution:

Check that the **Conference Server Role** has not been disabled on the SCS (Open the **System** menu, select **Servers**, and then click on the server to view enabled roles).



Checking the Conference Server Role

The Conference Server role is enabled by default on the SCS server. To check the enabled status of the server role:

1. From the SCS Home Page, select the **System** link followed by **Servers**.

AVAVA			Wed, 21 Apr 2010 1	0:51 AM Hom	e 7 Help C Logout Search
	Users	Devices	Features	System	Diagnostics
Home				Licensing Servers	
	There are 13 use	are defined:	There are 10	Branches	
	200, 201, 202, 20	3. 204. 205. 206. 207.	0004f21e794s	Dial Plans	
	209, 210, 211, steve, superadmin, more		0004f21f4bc7 001b387bd1d	Internet Calling	
	Add New User		002162ff7032 xxxxxxxxxxx m	Import / Export	<u>ent,</u>
			Add new phr	LDAP / AD	×
			Discover Devi	Restore	
				Localization	
				Certificates	
l				Software Updates	
				Date and Time	
				Logging Levels	

2. The Primary SCS Server will be displayed

4\V <i>F</i> \YF\			1000, 21 Apr 2010		Tielp Logod	Obdicit
	Users	Devices	Features	System	Diagnostics	
Servers						
					Add Server	Clicking the Sand
	Name		IP Address	Description	Status	Profiles button will
	Scsitel2	.iteluk.com	10.1.1.195	Primary server	Registered	files for all services to
						an anecuse services at be restarted automatically. This is rarely needed as configuration files are sent by default when their associated configuration has been changed. However, in the case where a distributed server was not available at the time of a configuration change, this button can be used to re- send the configuration.

3. To view the Server Roles, click on the Name link of the Primary Server.

AVAVA			Wed, 21 Apr 2010	10:52 AM Home	7 Help Cogout	Search
	Users	Devices	Features	System	Diagnostics	
Servers						
			-		Add Server	
	Name		IP Address	Description	Status	Profiles button will
	scsitel2.i	teluk.com	10.1.1.195	Primary server	Registered	cause configuration files for all services to
	Send Profiles	Delete				selected servers, and all affected services to be restarted automatically. This is rarely needed as configuration files are sent by default when their associated configuration has been changed. However, in the case where a distributed server was not available at the time of a configuration can be used to re- send the configuration.

4. The server screen will be displayed. Click the Server Roles link.

АУА				Wed, 21 Apr 2010	10:54 AM Home Help	Logout Search
	Users	Devices	Features	System	Diagnostics	
ure	Servers > scsitel2.iteluk.com					
Roles					Refresh every 30 seconds	Warning: Restarting
es	Name	Status			Role	services causes service interruption. Do it only if a
	Park	📀 Running			Primary SIP Router	service requires restart or is not working property. In
r	Instant Messaging	📀 Running			Instant Messaging	such case you might want
	Call Control	🕑 Running			Management, Primary SIP Router	report an issue.
	MyBuddy	📀 Running			Instant Messaging	This page will refresh
	Statistics	📀 Running			Management	switch automatic refreshing
	Configuration	🕑 Running			Management	checkbox.
		📀 Running			ACD	You can also modify the refresh interval by clicking
	Phone Provisioning	🕝 Running			Management	on the current interval and then enter a new value.
	ACD Agent Status	Running			ACD	
	ACD Statistics	📀 Running			ACD	
	Shared Appearance Agent	💿 Running			Primary SIP Router	
	Media Services	Running			Conferencing, Voicemail	
	🔲 🌼 Licensing	Running			Management	
	Voicemail MWI	📀 Running			Voicemail	
	SIP Trunking	🕝 Running			SIP Trunking	
	Paging	📀 Running			Primary SIP Router	
	Media Relay	📀 Running			Primary SIP Router	
	SIP Registrar	🕑 Running			Primary SIP Router	
	CDR	📀 Running			Management	
	SIP Proxy	📀 Running			Primary SIP Router	
	Presence	📀 Running			Primary SIP Router	
	Conference Recording	🕝 Running			Conferencing	
	Voicemail and Auto Attendar	nt 💿 Running			Voicemail	
	Postart Pofrosh					

5. The Conferencing Server will be displayed in an enabled state. If it is not, select the roles check box and click the **Apply** button followed by the **OK** button.

Δνανα					Wed, 21 Apr 201	0 10:56 AM 🦷 Home 🦷 He	Ip Cogout Search
		Users	Devices	Features	System	Diagnostics	
Server							
Configure Server Roles Services NAT Monitor	Servers + scsitel2.iteluk.o V ACD V SIP Trunking V Conferencing V Instant Messaging V Management V Primary SIP Route V Voicemail OK Apply	om ar Cancel					One or more roles can be enabled on each server. All roles can run on one single server or the different roles can be distributed to several servers forming a cluster. A high availability configuration can be configured by enabling a redundant SIP router role. Roles can be moved to dedicated servers to improve performance.

6. Click the Features link, followed by Conferencing.

			Users	Devices	Features	System	Diagnostics	
erver					ACD			
Configure Services Services NAT Monitor	Serve V V V V V V	ACD SIP Trunking Conferencing Instant Messaging Management Primary SIP Rout Voicemail OK Apply	oom 9 er Cancel		Agent Status Conferencing Auto Attendants Intercom Paging Groups Hunt Groups Call Park Music on Hold Phonebooks Instant Messaging			One or more roles can be enabled on each server. All roles can run on one single server or the different roles can be distributed to several servers forming a cluster. A high availability configured by enabling a redundant SIP nouter role. Roles can be moved to declated servers to improve performance.

7. The Conference Servers screen will be displayed.

		Users	Devices	Features	System	Diagnostics	
nference Servers							
						Refresh every 30 seconds	Quick Links
		Name	De	scription	Confer	ences	Servers
		scsitel2.iteluk.com	Prir	nary server	⇒ <u>0(0</u>	active)	User Groups
	Refresh						Conterence Servers are created and administered under System / Servers. A single conferences server can host a large number of conferences. For every user it is possible to user Groups / Conference Assignments to configure this feature before creating the users. The conference server can run on dedicated hardware or be colocated with other services. Several conference servers can be created per system. This page will refresh automaticating the <i>Refresh</i> checkbox. You can also modify the refresh linkerval by clicking on the current linkerval and

Problem: Cannot change conference settings

Solution:

Check that the conference has been assigned an owner.

Problem: Cannot access the conference bridge

Solutions:

Check the following:

- Has the conference server role been disabled?
- Is the conference extension correct?
- Is the conference participant PIN correct?
- Have the default conference dial pad options changed?



Problem: Conferences are not automatically assigned to SCS users

Solutions:

Check the following:

- Has a conferencing **User Group** been defined?
- Has the **Automatic Conference Assignment** facility been enabled for the conferencing User Group?
- Have users been created but not added to the conferencing User Group?

See the 'Conferencing' guide for in-depth guidance on configuring and using the conference server.



Problem: Users can't dial Outlook contacts direct from soft phones

Solution:

Due to the length and make-up of contact numbers within the Outlook contact list, some issues may arise when you attempt to dial a number using the softphones' 'Click-to-Dial' facility. You therefore need to find a way to format numbers before they leave the softphone and are passed to the SCS. For example:

- A customer has dial plans for local and long distance calls configured on the SCS.
- The local dial plan is configured to deal with numbers of seven digits with a prefix of 9. For example: 9 123-4567.
- The long distance plan is configured to deal with ten digit numbers with a prefix of 91. For example: 91-123-4567-899.
- The customer wishes to use Click to Dial to dial contacts directly from the Outlook contact list.
- Problem: contacts within the Outlook contact list have been stored as ten digit numbers, whether they are local or long distance.

The solution to this problem is to configure a dial plan directly within the softphone to enable it to format all numbers dialled either directly from the contact list or via the dial pad so that the SCS can interpret them and then utilize its local and long distance dial plans accordingly. See the SCS 4.0 Configuring the Avaya 3456 UC Client Task Based Guide for instructions on how to configure such a plan.

Problem: SIP Trunk incoming calls routing to wrong destination

Solutions:

Check the following:

- Is the internal Session Border Controller available?
- Has the SIP Trunk server Role been added?
- Has the SBCs SIP Incoming Call destination been left as the default 'Operator'? If the operator is not required as the incoming SIP call destination, clear the field and add aliases referencing incoming SIP trunk numbers.
- Have user aliases been defined referencing incoming DID numbers provided by the ITSP?
- For conferences, have Dial Plan rules been created to direct calls to the conference extension. Alternatively, do conference extensions represent incoming DID numbers?



Problem: SIP Trunks General Call Failure

Solutions:

Check the following:

- Is the internal Session Border Controller available?
- Has the SIP Trunk server Role been disabled?
- Has the Public Port been defined?
- The Public Port field allows you to specify a port as requested by your ITSP. The public port is the port exposed to the public network through your firewall settings. If your firewall restricts inbound traffic, you must open this port on the firewall to allow inbound signaling from the ITSP. The SCS 'external port' is the port that 'faces' the firewall and is associated with the firewall's public port. Therefore, the firewall must be configured to send packets from the firewalls public port to the SCS external port. If you leave the public port field blank, the external port is assumed to be the same as the public port and the mapping assumed to be symmetric. Please confer with your ITSP for details regarding NAT and any specific firewall rules the ITSP may require.
- Has NAT been configured on the SCS?
 - Is STUN utilised and if so, is the correct STUN Server referenced?
 - Will a Public IP address be referenced for NAT. When the server is deployed behind a NAT, the "Public IP address" field must be set to the Internet-facing IP address of the NAT / firewall device fronting the SCS server.
- Has NAT traversal been configured on the SCS, this is found under Internet Calling where the Server Behind NAT option should be enabled.
- With regards to SIP Trunk Gateways, are you using the default templates in relation to your ITSP?
- Has the ITSP defined specific configuration parameters for you to use? For example, specific parameters are required for AT&T ITSP configuration.
- Has the SBCs SIP Incoming Call destination been left as the default 'Operator'

- Have user aliases been defined referencing incoming DID numbers provided by the ITSP?
- For conferences, have Dial Plan rules been created to direct calls to the conference extension. Alternatively, do conference extensions represent incoming DID numbers?
- Have Dial Plan rules been configured to provide external call access utilizing the SIP Trunk gateway for your ITSP?

See the SCS 4.0 Device Configuration Gateways Guide for full configuration details include required firewall / NAT parameters.



Problem: Mediant FXO Gateway – Inbound Calls do not terminate on SCS

After the Gateway configuration is completed on the SCS, the ini file is uploaded to the gateway and the gateway restarted. However if you find that the data from the ini file has not propagated all of the gateway fields:

Solutions:

Check the following:

- Has the Primary DNS server been defined on the gateway?
- Does each port on the gateway have a corresponding value for the terminating extension number?



The Primary & Secondary DNS server settings configured on the SCS under Devices > Network Parameters are propagated to all gateway devices and ini files for those devices. Consequently, these DNS settings will be those used by the gateway device. For Static IP Address configuration of an gateway device, DNS settings must be defined, otherwise calls to and from the device will fail. Please refer to the *Propagating DNS Settings to Gateway Devices* section of this guide.

AVAVA				/	Wed, 19 May 201	0 10:59 AM 🕜 Home 🦳 Help
		Users	Devices	Features	System	Diagnostics
Global Network Paran	neters		Phones			
			Phone Groups			
	DNO		Gateways			
	DNS		SBC Routes			
	Enabled		Device Files			
		B 100 (Network Parameters			
	Name	DNSServer1	Device Time Zone	-		
	Description				4	
	Address	10 1 1 3				
	Address	10.1.1.0				
	The IP		or host name of the service. F	or DNS services an IP	address must be specif	fied.
	ОК Ар	ply Cancel				

Logging Levels

The SCS logging level parameters determines the amount of information stored in a system snapshot. Most of the time the system should run at *NOTICE* level. During troubleshooting increase the logging level temporarily before taking system snapshots. Reset logging level back to *NOTICE* once the problem is resolved. The most detailed logging level is *DEBUG*, the least detailed -*EMERG*.

AVAVA					Wed, 28 Apr 2010	10:57 AM Home Help
	u	lsers	Devices	Features	System	Diagnostics
Logging						
						Hide Advanced Settings
	General Logging Level	select level	~			
	Services Logging Level	Changing general lo	g level will reset th	e logging levels for all the	e services.	
	ACD	NOTICE	~			
	Supervisor	INFO	*			
	SIP Proxy	NOTICE	*			
	SIP Registrar	NOTICE	*			
	Park	NOTICE	*			
	ACD Agent Status	NOTICE	*			
	CDR	NOTICE	~			
	Voicemail and Auto Attendar	NOTICE	*			
	Conference Recording	NOTICE	~			
	MyBuddy	NOTICE	*			
	Voicemail MWI	NOTICE	*			
	Presence	NOTICE	*			
	Paging	NOTICE	~			
	Media Services	NON-DEBUG	*			
	Media Relav	NOTICE	~			
	Shared Appearance Agent	NOTICE				
	Call Control	NOTICE				
	Phone Provisioning	NOTICE				
	Instant Messaging	NOTICE				
	OID Truckies	NUTICE				
	SIP Trunking	INFO	*			

As a general guideline:

You should not have to use the DEBUG logging level unless advised to do so by technical support. It is generally never required in a production installation. The INFO logging level is more than adequate and will produce dramatically less log data.

Whenever possible, set the proxy logging to INFO as the majority of messages pass through the proxy. Therefore, it is invaluable as a data collection point. As long as a system has a reasonable size disk, leaving the proxy at INFO level will only have a negligible impact on system performance.

Collect snapshot data while the problem is occurring, not after you've found a solution to the problem.

Limit the log data to a reasonable period of time when the problem is occurring; if possible, it's best to capture a window from before the problem started to after it is known to have occurred, but support rarely need more than a few minutes of data either side of the event if it can be narrowed down that much.

Avaya 3456 UC Client

The Avaya 3456 will not upload contacts from SCS phonebook if Internet Explorers proxy is in use. Internet Explorer's Proxy settings must be disabled or exceptions for the SCS address must be put in place in order for the 3456 to gain access to the Phonebook. This is because non-Microsoft products take their connection settings from IE.

SAS Stand Alone Survivability

Problem: When configuring SAS with AudioCodes gateway firmware prior to version 5.6, if the SCS fails external calls may also fail. The FXO gateway attempts to route to the SCS instead of making an external call.

1. Login to the SCS. From SCS Home Page select **Devices** followed by **Gateways**.



2. Select the Gateway to be configured with the SAS feature.

		Users Dev	vices Featu	res System	Diagnostics		
ateways							
				A	dd new gateway	~	Quick Links
	Name	Address	Location	Model	Description		Dial Plans
	AVAYABCM50	10.1.1.156	All	Unmanaged gateway			Job Status
	MP118FXO	10.1.1.187	Boston	AudioCodes MP118 FXO	Local PSTN Gatewa	ay	gateways", "SIP trunks" and
	To_Skype	sip.skype.com	All	SIP trunk	SIP Trunk Config de relating to Skype	etails	gateway models listed here are plug & play configurable
	To_Bandwidth.co	om ot.bandwidth.com	All	SIP trunk			very similar to how phones are configured. A
	ToCbeyond	sipconnect-fca.atl0.cb	eyond.net All	SIP trunk	Sip Trunk Gateway Cbeyond ITSP	for	configuration file for the respective gateway is automatically generated. It
	ToVoxitas	netlogic.net	All	SIP trunk	SIP Trunk Gateway Voxitas ITSP	for	can be automatically downloaded by the gateway
	To BT ITSP	sip.ser-001.nat.bt.com	n All	SIP trunk	SIP Trunk Gateway ITSP	for BT	when the gateway is plugged in and powered up, or the configuration file can
	ToCS1000	200.30.30.5	All	SIP trunk			be manually downloaded
	Med_1000	10.1.1.190	All	AudioCodes Mediant 1000	PRI Mediant 1000 PSTN Gateway	4	gateway. An "unmanaged gateway" needs to be
	Med1000BRI	10.1.1.210	All	AudioCodes Mediant 1000	BRI Mediant BRI Gatewa	ау	created for all manually configured gateways so that
	Send Profiles Sen	d All Profiles Restar	t Delete				they can be inserted into the dialplan.

3. The gateway's configuration details will be displayed. Click the **Advanced Parameters** link.

AVAYA					Thu, 20 May 201	10 9:39 AM 💮 Home 🛛 7 Help
		Users	Devices	Features	System	Diagnostics
Gateway Details						
Configuration Gateway : MP111 PSTN Lines Enabled Caller ID Enabled Dial Plan Name SIP Address Voice Codecs Proxy and Registration DTMF & Dialing Advanced Parameters	Gateway : MP118FXQ / Audi Enabled Name Address Serial Number	MP118 WP118 200.30 For a PST External An STER enter the 004021	FXO FXO .30.26 .30.26 P address or fully quilling P address or fully quilling IP address o	of the gateway (axample The gateway can be or ed hostname of the Inte d in the field below. For fied name of the other sy	: 10,1,1,1) or the fully q any routed subnet with net Telephany Service F a Direct SIP Trunk: To ir stem.	Show Advanced Settings alfed hostname of the galeway out NAT. For an ITSP SID Trunk rovider (e.g. itsp.oxample.com). Note: terconnect two VoIP systems using SIP
PSTN to IP Call Routing Supplementary Services FXO Network Media RTP/RTPC Management	Firmware Version Location Shared	Usually th 6.0 ♥ - all - Pasticit h and you n based on office wou bandwidth gateway in ♥ If checkeed checkeed b used by o	e serial number is set to v e gateway by selecting red to create a branch- in which location or by id like to have a gateway id like to have a gateway id like to have a gateway id like to have a gateway the second to the second or to use Cateford to the s not available call routin d this gateway can be u y default so that users ther users in other locat	the device's MAC addre a specific location for wh for every location that no which user the call origin ty preference so that call red by an analog gatewar- ng will fall back to otherg sed by any user in any lo in an identified location a ions.	ss, for example: 004021 ich it can be used. A loc reds to be distinguished the (source nouting). Thi a are nouting through the s are nouting through the PSTN n ateways specified for the cation, even if a specific cation, even if a specific	1131fa. Ition is represented by a group of users This setting allows routing of calls is useful if users located in a branch ir local gateway, i.e. to preserve WAN umber assigned to it. Only if that orresponding dialing rule. location is selected. This setting is leway, but the gateway can also be
	Description OK Apply (Cancel				2

4. If the Advance Parameters are not displayed, click the **Show** Advanced Parameters link.

AVAVA					Thu, 20 May 2010 9:41 AM Home 7 Help Logout Search			
		Users	Devices	Features	System	Diagnostics		
Gateway Details								
Configuration PSTN Lines Caller ID Dial Plan SIP Voice Codecs Proxy and Registration DTMF & Dialing Advanced Parameters IP to PSTN Call Routing PSTN to IP Call Routing Supplementary Services	Gateway : <u>MP118FX0</u> /Au	dioCodes MP118	I FXO			Show Advanced Settings	To download the device configuration file citck on the link(s) below: <u>Q040214131FA.ini</u> To setup a new gateway fill in the parameters on this page, then setup PSTN Lines. No other settings need to be considered as all gateway parameters are auto-configured for a hytical deployment. Consult the gateway vendor's manual or ask an expert for advice if other parameters need to be adjusted.	
FXO Network Media RTP/RTPC Management								

5. The SAS configuration parameters are located at the bottom of the **Advanced Parameters** page.

6. In the SAS Default Gateway field, enter the IP address of the device that has access to the PSTN. In this example, the device is an AudioCodes FXO Gateway.

Max Call Duration [min]	omente	(Default: 0)
Enable LAN Watchdog		(Default: unchecked)
Stand-Alone Survivability		
Enable SAS	V	(Default: unchecked)
	Enable/disable Stand-Alone Su	urvivability
SAS Binding Mode	✓	(Default: checked)
	Determines the SAS applicatio user=phone is defined, the bin entire URI, i.e., User@Host (d	in database binding mode. [0]: URI = If the incoming AoR in the INVITE requests is using a tel: URI or ding is performed according to the user part of the URI only. Otherwise, the binding is according to the elault). [1]: User Part only = The binding is always performed according to the User Part only.
SAS Survivability Mode	0	(Default: 0)
	The Survivability mode used by	y the SAS application. 0:standard; 1:Always Emergency; 2:Ignore Register
SAS Enable ENUM		(Default: unchecked)
	Determines whether the SAS a	application uses ENUM queries to route incoming INVITE requests when in Emergency mode.
SAS Registration Time	20	(Default: 20)
-	SAS Registration Time	
SAS Local SIP UDP port	5080	(Default: 5080)
	UDP Port for SAS SIP signalin	, 19
SAS Local SIP TCP port	5080	(Default: 5080)
	TCP Port for SAS SIP signaling	9
SAS Local SIP TLS port	5081	(Default: 5081)
	TLS Port for SAS SIP signaling	9
SAS Default Gateway	10.1.1.187	(Default: 10.1.1.187)
	SAS Default Gateway	
SAS Proxy Set	0	(Default: 0)
	Determines the Proxy Set (ind served by the SAS application	, ex number) used in SAS Normal mode to forward REGISTER and INVITE requests from the users that are
Redundant SAS Proxy Set	-1	(Default: -1)
	Determines the Proxy Set (ind database.	ex number) used in SAS Emergency mode for fallback when the user is not found in the Registered User
OK Apply Cancel		

Note: For firmware versions pervious to version 5.6, the SAS Short Number Length field will be displayed. The **SAS Short Number Length** determines the length of the suffix used for Stand-Alone Survivability (SAS) Short Numbering support. The dialed number is compared to the registered number. This will be the number of digits to be passed to the gateway. For example, if an IP phones registers with 9 01244 670200 the parameter would be set to 11 (digits). The valid range is 0 to 63 characters.

Enable/disable Stand-Alone Survivability 20 UDP Port for SAS SIP signaling	(Default: unchecked) (Default: 20)
Enable/disable Stand-Alone Survivability 20 UDP Port for SAS SIP signaling	(Default: unchecked) (Default: 20)
Enable/disable Stand-Alone Survivability 20 UDP Port for SAS SIP signaling	(Default: 20)
20 UDP Port for SAS SIP signaling	(Default: 20)
UDP Port for SAS SIP signaling	
5080	(Default: 5080)
UDP Port for SAS SIP signaling	
5080	(Default: 5080)
TCP Port for SAS SIP signaling	
5081	(Default: 5081)
TLS Port for SAS SIP signaling	
SAS Default Gateway	
	UDP Port for SAS SIP signaling 5080 TCP Port for SAS SIP signaling 5081 TLS Port for SAS SIP signaling SAS Default Gateway

Please refer to the SCS 4.0 Device Configuration gateways TBG for details relating to SCS SAS Configuration with gateway firmware version 6.0.

Instant Messaging

Problem: Users cannot log in to Instant Messaging clients.

Solution

First ensure that the Instant Messaging server is running:

1. From within the SCS administrator interface, open the **System** menu and select **Servers**.

AVAYA			Tue, 20 Apr 2010 2	2:33 PM 🔶 Home	7 Help 6 Logout Search
	Users	Devices	Features	System	Diagnostics
Home				Licensing	
				Servers 🚤	
				Branches	
	There are 13 use	ers defined:	There are 10	Domain	
	200 201 202 20	3 204 205 206 207	0004f21e794	Dial Plans	
	209, 210, 211, ste	eve, superadmin,	0004f21f4bc7	Internet Calling	
	more		001b387bd1c	Permissions	
	Add New User		<u>00216207032</u>	Import / Export	<u>ent,</u>
			<u></u>	LDAP / AD	
			Add new ph	Backup	
			Discover Dev	Restore	
				Localization	
				Certificates	
				Software Updates	
				Date and Time	
				Logging Levels	

2. Select the server on which the IM service is situated. On some networks, some services will be distributed across several different servers. In this example all SCS service run from the same server.

AVAVA		Search				
	Users	Devices	evices Features System		Diagnostics	
Servers						
		1			Add Server	
	Name		IP Address	Description	Status	Clicking the Send Profiles button will
	E scsitel	2.iteluk.com	10.1.1.195	Primary server	Registered	files for all services to
	Send Profiles	Delete				be sent to the selected servers, and all affected services to be restarted automatically. This is

3. Click on the **Server Roles** menu, located on the left-hand side of the screen.

	Users	Devices	Features	System	Diagnostics	
erver						
Configure	Servers scsitel2.	iteluk.com				
Server Roles	ACD					be enabled on each
Services	SIP Trunking	9				server. All roles can run on
NAT	Conferencin	g				one single server or
Monitor	Instant Mes	saging				be distributed to
	Managemer	nt				forming a cluster.
	Primary SIP	Router				configuration can be
	Voicemail					configured by enabling a redundant
	OK A	pply Cancel				Roles can be moved to dedicated servers to improve performance.

4. A list showing all available roles is displayed. Roles that are ticked are enabled. Ensure that the **Instant Messaging** role is ticked. By default all roles should be enabled by default.

	Users	Devices	Features	System	Diagnostics	
Server						
Configure Server Roles Services NAT Monitor	Servers ► scsitel2. ✓ ACD ✓ SIP Trunkin ✓ Conferencin ✓ Instant Mes ✓ Managemen ✓ Primary SIP ✓ Voicemail OK A	iteluk.com g g saging nt Router				One or more roles car be enabled on each server. All roles can run on one single server or the different roles can be distributed to several servers forming a cluster. A high availability configuration can be configuration can be configured by enabling a redundant SIP router role. Roles can be moved to dedicated servers to improve performance.

5. Click **Apply** followed by **OK** to confirm any changes made.

Note: If you do make a change to this screen you will be asked to restart services. Click on the purple link that appears at the top of the screen to got to the service restart page. Tick the check-boxes of any services listed and then click the **Restart** button.

6. You will be returned to the server selection page.

Once you have ensured that the Instant Message role is enabled, ensure that users are entering the correct IM password. If IM passwords have not been configured then they will be the same as the users' IM IDs.

Problem: MyBuddy not present in users' IM clients

Solution:

If Instant Messaging group are members of a group:

1. Open the **Users** menu and select **User Groups**.

AVAYA			١	Wed, 26 May 2010 3:5	58 PM 💮 Home	7 Help	C Logout	earch
	Users	Devices	Features	System	Diagnostics			
Home	Users							
	User Groups							
	Extension Pool							
	There are 15 u TLS Peers		There are 12 phones	defined:				
	200, 201, 203, 204, 205, 206, 207, 2	08, 209,	0004f21d8d5f, 0004f2	1e794a, 0004f21eea	<u>0d,</u>			
	210, 211, 212, 213, Lisa, superadmin	n, more	0004f21f4bc7, 000ee	1238b51,001b386bd	<u>1d0.</u>			
	Add New User		franksoft, paulsoftclier	it, xxxxxxxxxxx, more	32,			
https://	A .		Add new phone					
- mar and	and the second s	prosent and	D' over Devices	a Caracteria	$\sim\sim\sim\sim\sim$	and the second s	Auran-	- Are Million and Areadan

2. Click on the Instant Messaging group.

AVAVA		Wed, 26 May 2010 3:59 PM Home Help Logout Search								
		Users	Devices	Features	System	Diagnostics				
User Groups										
	Groups allow you to organize users into logical groups and share settings between users in the same group. Users can be in any number of groups. Groups can also be used to specify a location, such as a branch office. This can be useful if location based routing is used (see gateway configuration).									
			Group Name		Num	ber of Members	group has highest precedence.			
		1. 🝰 administrators		1						
		2. 🍰 🔟 🗲 🗕		10	1					
لحسبت		3. 🛃 Phonebook	\sim		المعسرية	-	- And and the second second			

3. Click on the Instant Messaging menu.

				-				
lear Group Sattinge	User	'S	Devices	Features	System	Diagnostics		
ser Group Gettings								
Configure						Show Advar	ced Settings	
Unified Messaging	Instant Massaging							
Schedules	instant wessaging							
Conference	Enable IM account		(De)	fault: unchecked)				
External User		If checker	the associated i					
Speed Dials	Add user group as IM	_						
Music On Hold	group		(Dei					
Permissions		If checked	d, the user group v					
Caller ID	Add MyBuddy to the IM	N	(De)	fault: unchecked)				
Personal Auto-Attendant	group							
Instant Messaging		If checked	d, MyBuddy will be	added as a member of the	IM group. MyBuddy ca	n not be removed by a u	ier.	
MyBuddy Call Farwardian	OK Apply	Cancel						
Call Forwarding								

4. Ensure that the **add MyBuddy to the IM group** option is enabled by ticking the check-box.

Speed Dials Music On Hold Permissions	Add user group as IM group	If checked, the assoc If If checked, the user	Salis d'account will be configured on the Instant Messaging service. (Default: unchecked) group will be added as an IM group on the Instant Messaging service.	and the second of the second o
Caller ID Personal Auto-Attendant Instant Messaging	Add MyBuddy to the IM group	If checked, MyBuddy	(Default: unchecked) will be added as a member of the IM group. MyBuddy can not be removed by a user.	
MyBuddy Call Forwarding	OK Apply Car	ncel		

5. Click **Apply** followed by **OK**.

If IM users are not grouped on your network each user will have to enable MyBuddy from within their own SCS user portal:

1. Log in to the user interface and click on **My Information**.

AVAVA					Alex Samson	Wed, 26 May	2010 4:10 PM	Home	Help	Logou
	Voicemail My I	nformation	Call Forwarding	Speed Dial	Call History	Phonebook	Phones			
My Information										
Contact Information Unified Messaging	MyBuddy Click butte MyBuddy through IM clien	on to enable M nt.	/lyBuddy, which will ac	id MyBuddy to your	IM buddy list. Alternativel	y you can manua	lly add			
Distribution List Conferences	IM Notification									
Instant Messaging Music On Hold	Conference Entry IM	Send me a	<i>(Default: check</i> n Instant Messages as pa	red) rticipants arrive						
Attendant MyBuddy	Conference Exit IM	l⊽ Send me a	<i>(Default: check</i> n Instant Messages as pa	re <i>d)</i> rticipants depart						
	VoiceMail Start IM	Send me a	<i>(Default: check</i> n Instant Message when a	ed) a caller begins a call an	swering session					
have	VoiceMail End IM		(Default: check	ed)		and the	Manager Barriel	Andreas		, manual

- 2. Click on MyBuddy.
- 3. Click the **MyBuddy** button at the top of the screen to enable MyBuddy.

AVAVA					Alex Samson	Wed, 26 May 2	010 4:10 PM	Home	? Help	Logou
	Voicemail	My Information	Call Forwarding	Speed Dial	Call History	Phonebook	Phones			
My Information										
Contact Information	MyBuddy C	lick button to enable N	lyBuddy, which will ad	id MyBuddy to your I	I buddy list. Alternative	ly you can manually	add			
Unified Messaging	MyBuddy through	IM client.								
Distribution List	IM Notificatio	on								
Conferences										
Instant Messaging	Conference E	ntry IM 🗵	(Default: check	ed)						
Music On Hold	- marker	and and and a second	thant hssage	s in ts an	~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~	with	and and the second		and the second	tel _{ke} rener

4. Click **Apply**.

Problem: MyBuddy and/or other SCS IM Buddies do not appear in Gtalk.

Solution:

There are two solutions that you might try for this issue, the first of which is server related, the second relates to user account settings.

First, check that IM federation is enabled:

To configure Federation:

1. Open the Features menu and select Instant Messaging.

-	Users	Devices	Features	System	Diagnostics	
lome			ACD			
			Agent Status			
			Conferencing			
	There are 6 user	s defined:	Auto Attendants	hones defined:		
	200 201 202 20	3 204 superadmin	Intercom	0004f21eea0d		
	more	0, 204, 30peradmin,	Paging Groups	,002162ff7032,		
	Add New Lleer		Hunt Groups	nore		
	Add New Oser		Call Park			
			Music on Hold	Jile		
			Phonebooks	<u>ues</u>		
			Instant Messaging			
				-		

2. Click on Server to Server..

AVAYA			Wed, 12 May 2010 3	51 PM Home	7 Help 🔒 Logo	ut Search
	Users	Devices	Features	System	Diagnostics	
Instant Messaging						
Server to Server Message Logging	Server to Se	rver Federation				
	Enabled			(Default:	unchecked)	
		If checked IM servic	e allows server to server f	ederation with external	M servers.	
	Port	5269		(Default:	5269)	
hand a second	a Assessed	Rort number used b	y IM danying for service to	server federation.	A	a share where the

3. Enable federation by ticking the **Enable** check-box.

AVAYA			Tue, 20 Apr 2010 4	40 PM 🧄 Mome	7 Help Logo	ut Search
	Users	Devices	Features	System	Diagnostics	
Instant Messaging						
	Servers Scsitel:	<u>2.iteluk.com</u> ▶ Instar	nt Messaging		Show Advanced Setting	<u>15</u>
	Server to Se	rver Federation				
	Enabled	V		(Defaul	t: unchecked)	
		If checked IM	service allows server to se	erver federation with ext	ernal IM servers.	
	Port	5269		(Defaul	t: 5269)	
	-	Port number u	used by IM service for sen	er to server federation,		, have a second s

4. Next, decide which IM servers will be allowed communication with the SCS. If all servers are acceptable simply tick the **Allow any server** check-box.

Ανάγα			Tue, 20 Apr 2010 4:	44 PM 💮 Home	e ? Help 🕆 Logo	ut Search
	Users	Devices	Features	System	Diagnostics	
Instant Messaging						
	Servers Scsitel2.itelu	. <u>com</u> ⊾ Instar	nt Messaging			
					Show Advanced Setting	<u>15</u>
	Server to Server	Federation				
	Enabled	V		(Defau	ilt: unchecked)	
		If checked IM	I service allows server to se	rver federation with ex	ternal IM servers.	
	Port	5269		(Defau	ılt: 5269)	
		Port number	used by IM service for serv	er to server federation.		
	Allow any server			(Defau	ilt: unchecked)	
		of Disallowed	iy external IM server is allo servers. d the Allowed servers need	wed to connect. Excep I to be listed.	ntions can be specified on the lis	51
	Allowed servers					
		List of the se	rvers that are allowed to co	nnect.		
		Separate mul	tiple servers with commas:	sabled. sever1.com:4563, ser	ver2.ca, server3.net.	
	Disallowed servers					
		List of the ser Only relevant	rvers that are prohibited fro when Allow any sever is e	m connecting. nabled.		
	and the second	Separate mul	tiple servers with commas:	sever1.com:4563, ser	ver2.ca, server3.net.	a second

Use the **Disallowed servers** field to define any servers that are exceptions to the 'Allow any server' clause. This is a useful feature if, broadly speaking, all servers are allowed but there are a few that you would like to block. For example, you may wish to allow all IM servers except Google Talk, in which case you would enter *gtalk.com* in the Disallowed servers field.

- 5. Click **Apply** followed by **Home** to continue.
- 6. Restart any necessary services by following the pink here link.

AVAVA				١	Wed, 12 May 2010 3:5	7 PM 🧄 Home	7 Help	Logout	Search
		Users	Devices	Features	System	Diagnostics			
Home									
	One or more service	es need to be restart s defined:	ed. For details cl	lick: <u>here</u>	lefined:				
	200, 201, 202, 203 Add New User	3, 204, superadmin, m	<u>iore</u>	0004121d8d5f,00042 002162ff7032,xxxxxx Add new phone Discover Devices	1eea0d, 002162ff36b xxxxx, more	<u>4.</u>			

The SCS IM service can now be used to message contacts on other messaging services outside of its own domain.



If the problem persists once you have checked IM Federation settings, check whether an alternate IM account has been defined within the user's SCS profile:

1. Navigate to the user account in question by opening the **Users** menu and selecting **Users**.

AVAYA			Thu, 22 Apr 2010	9:27 AM 🕜 Home	? Help 💧 Logout	Search
· · · · ·	Users	Devices	Features	System	Diagnostics	
Home	Users Users Groups Extension Pool TLS Peers 200, 201, 202, 20 210, 211, superac Add New User	Devices *s defined: 3, 204, 205, 206, 207, min, more	Features There are 10 0004f21e794 0004f216bf2 001587bd1 002162ff7032 xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx	System phones defined: a. 0004f21eea0d, .001536b41d0, .001536b44, .franksoft, paulsoftclien tore ices	L L	

2. Select a user.

AVAVA				Thu, 22 Apr 2010	9:30 AM 🍈 Home	e ? Help	Cogout	Search
		Jsers	Devices	Features	System	Diagnostic	5	
Users								
	Filter	y				Ad	d New User	Select the Add New
		User ID	First	Name Las	Name	IM ID	Allases	new user.
		<u>8 200</u>	Ed	Jone	3 200			After user is created you can associate it
		<u>8 201</u>	Frank	Corte	lli 201			with one or more managed phones
-		a 202	Tony	Smith	202			
		203	Eric	Cons	tantine 203			
		<u>8 204</u>	John	Carte	r 204			
/ manager and a second		<u>a 205</u>	Bob	Jenki	ns 205	A second		a summer

3. Click on **Contact Information**, located on the left-hand side of the screen.

AVAVA		Thu, 22 Apr 2010 9:32 AM 🚽 Home 📪 Help 👘 Logout Search				
	Users	Devices	Features	System	Diagnostics	
Identification						
Identification Unified Messaging Contact Information Phones Call Forwarding	User: 203 User ID	203 The User ID ci displayed by ti User ID, If user	an be a numeric extens he phone and it is there o DID configure the DI	on like 123 or a name like j fore recommended to use i D number (or its DNIS port	Show Advanced Settings smith. The User ID is the internal extension as the n) as an alias.	Existing Groups: administrators, Phonebook, InternalPhonebook, member, InternalGroup, IM, Sales

4. Toward the bottom of the contact information form you will find a field called **Alternative IM account**. Enter the user name and domain of the user's secondary IM account in this field – for example *ericcon*@googlemail.com.

Cathern	w r-ax ndh,	and the second	-
Instant Messaging MyBuddy	E-mail address	eric@business.com	
	IM account	203	
	Alternate E-mail address		
	Alternative IM account	[ericcon@googlemail.con	
	Location		
	Home address		
	Street		
	City		

5. Click **Apply** followed by **OK**.

The user can add MyBuddy or any other SCS-based IM contact to their external IM account. When adding an SCS contact to a non-SCS IM client, you must specify the full IM ID, including the domain name, for example: *mybuddy@scshost.scsdomain.com*.

Note: Clicking the **MyBuddy** button found in the user portal on the 'MyBuddy' page will not activate MyBuddy in an external IM account. MyBuddy must be added manually like any other contact using the client's 'add contact/buddy' process.

SCS Toolbar

Problem: Once installed, the toolbar does not register and function with the SCS.

Solution:

Ensure the SCS Host name has been entered correctly under the Toolbar's **Options** menu. The SCS user's login credentials should also be added.

] 📑 🤊 । -		The second se		Inbox - Mailbox	- Edward Jones - Microsoft Outlook			
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Mailbox - Edward Jones	<					- SCS account settings -		
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Problem: The Toolbar does not recognize UK numbers when numbers are dialed using the Toolbar application.

Solution:

With this release, the toolbar only recognises the North American telephone number format.

Below is a list of recognizable phone number formats of USA and Canada:

```
+1szzzsxxxsxxx
+1s(zzz)sxxxsxxxxx
(+1szzz)sxxxsxxxx
1szzzsxxxsxxxx
1s(zzz)sxxxsxxxxx
(1szzz)sxxxsxxxxx
zzzsxxxsxxxxx
(zzz)sxxxsxxxxxx
x - digit or character in upper case
zzz - area code
```

Avaya Support – Pre Contact Checklist

Before contacting SCS Avaya support, you may wish to consider and gather certain information which may assist the support team and lead to a quicker solution to the problem.

If possible, before contacting support:

- Provide a full description of the issue including appropriate screenshots of the SCS interface.
- A list of solutions/actions you have taken to attempt to resolve the issue.
- Note the release number of the SCS software installed.
- Confirm whether the SCS is in a High Availability or standalone environment.
- Confirm whether DNS A-Record/SRV records have been configured and utilised for the SCS installation.
- Confirm whether DHCP option 120 / option 66 in use in the SCS network.
- Create a diagnostics Snapshot relating to the configuration / problem issue.
- If appropriate, provide details relating to any gateways utilised, including firmware levels relating to PSTN gateways. For example, AudioCodes gateways.
- If appropriate, provide details relating to any SIP Trunk gateways utilised.
- If appropriate, provide details relating to the IP Phone models used in the SCS environment.

SCS Community Portal

The SCS Community Portal has a large amount of information including demonstrations and documentation relating to SCS 4.0. SCS administrators can register with a community account, and gain access to documentation, training and SCS related software.

Information on many SCS topics can be found within the SCS community portal. Information is also shared across the community and questions relating to the SCS can be raised with other community members.

The SCS Community Portal can be accessed via this web page.

http://scsavaya.com/

SCS administrators would first register with the Community portal.

SCS Community Portal	
MY ACCOUNT CONTENT	
CREATE NEW ACCOUNT LOG IN REQUEST NEW PASSWORD	
User account	
Username: *	
Spaces are allowed; punctuation is not allowed except for periods, hyphens, and underscores.	
E-mail address: *	
A valid e-mail address. All e-mails from the system will be sent to this address. The e-mail address is not made public and will only be used if you with to receive a new password or with to receive certain news or notifications by e-mail.	
CREATE HEW ACCOUNT	

Once registration is confirmed, access to the portal is available.

EDWARDJ	SCS TECHNOLOGY INFORMATION PAGE	
* Documentation * Demonstrations	This site is supported by Avaya R&D and is used to assist and support individ questions answered please post on the forums. To post a request for webs site better please click here.	tuals looking for information related to SCS. To get it eenhancement or things that would make this
Online Demo Vmware Demo Demo Software	Warning: This ste is NOT in anyway an emergency support portal. If you h attention please use the proper Nortel support channels.	ave a customer issue that needs immediate
 Mkt Readiness Modules 		
* Noces and Tricks	Posted Tue, 05/27/2008 - 13:01 by abdam	NEW FORUM TOPICS
· 3rd Party Development	This sta remotes information for Saftware Communication Sustem	* Is there a way to disable the IM
+ Polls		function on the Nortel 12XX phone?
 Create content 	Everything was just rebranded and moved to the scsavaya name. Please report any broken links or asses HERE.	 Getting out of an ACD Queue
* Forunis	All neuro lumon Brandad CPC proceedations and employing matantics	* 3456 LDAP
* BSS Feed	available HERE.	Information
Hy account		 Paging with the 1220
+ Log out		more
	SCS 4.0 BETA TECH TRANSFER	
	Posted PH, 05/14/2010 - 22:17 by albdam	2011
USEFUL LINKS	The SCS 4.0 Beta Tech Transfer was held May 14, 2010 and had about	PULL
 SinFoundry Wiki 	45 participants. The material along with narrated flash recordings a available on the site. Click Hare for the Material.	Which new feature in SCS 4.0 interests
+ SCS on Avava.com		you most?:
	alboam's blog - Aod new comment	O Instant Messaging Server
		O Avaya MyBuddy - Instant Message BOT
PECENT BLOG POSTS	Posted Wed, 05/12/2010 - 13:11 by abdam	with FMC ike capability
	Wall I hope eventone likes the new look. Red is definitely a hold miny. I	Conference Cal Passedine
 SCS 4.0 Beta Tech Transfer 	wanted to give everyone a quick update on where things are at with	True likes Phoneshook Cours in send
 SCS 4.0 Update 	the commercal release of SCS 4.0. SCS 4.0 passed the official commit	User editable
 SCS 4.0 Coming Soon New Rook based on 	mestone within the Avaya AGSP process. This committed to a 1st week of July release of the product. External Abha testing has alwardy	LANCE.
opensource side of SCS	started with beta coming in a few weeks. We are so confident with the	R.O.L.S
» SCS 3.0 Update	software quality that a 100+ user bank is live right now in external Alpha	
SCS 3.0 General Release	testing. Cack here to Read Hore.	
 SCS 3.0 Development Status 	albdam's blog Add new comment Read more	
SCS Development Blog SCS Madating Toom Pant-		
 SUS Planeting reall Rant' what we're working on & why 	SCS NEWS	
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