ADIRAN[®] NetVanta Unified Communications Technical Note

Configuring the SJphone

Introduction

The SJphone from SJ Labs provides an easy-to-use interface, modern style, and broad range of features. The SJphone is fully interoperable with the NetVanta Enterprise Communications Server, but cannot be automatically configured. This guide provides instructions for manually configuring the SJphone to operate with the UC server.



Known Integration Issues

ADTRAN has identified the following integration issues with the SJphone:

- The SJphone does not operate properly if it is installed on the same machine as the UC server.
- When Music-on-Hold is disabled and a call from an SJphone is parked and then picked up by a Cisco IP Phone 7912, only one-way audio is established. You must enable Music-on-Hold to avoid this issue.

Preparation

Checking the Software Version

Refer to *NetVanta UC Server Interoperable SIP Device Features and Comparisons* technical note available online at <u>http://kb.adtran.com</u> to determine the most recently supported version of software for the SJphone.

To determine what version of SJphone you have:

- 1. Start the SJphone application.
- 2. Select **Menu > About**.
- 3. Scroll down to **Application Load ID**.

If the software version on the phone is earlier than the version noted in *NetVanta UC Server Interoperable SIP Device Features and Comparisons* technical note available online at <u>http://kb.adtran.com</u>, upgrade the SJphone.

Obtaining the Current Software Version

To obtain the current software version:

- 1. Download the appropriate version from <u>http://www.sjlabs.com/.</u>
- 2. Install the file.

Telephone Configuration

Gathering Information

To configure the SJphone, you need the following information:

Name Equivalent to the name of the identity you want to associate with the SJphone.

- Account The Session Initiation Protocol (SIP) authentication identifier associated with the above identity. This is required by any SIP endpoint to register with the SIP private branch exchange (PBX).
- PasswordThe SIP authentication password associated with the above identity. This is typically (but
not always) the same value as your voicemail access PIN.

Determining the Authentication ID and Password as the User (that owns the identity)

- 1. Launch the UC client.
- 2. Log in as the user you want to associate to the phone.
- 3. In the left bottom pane, take note of the identity name.

🖏 User One [10	001	on UC Server]	
Number of rings:	4	~	

4. Select the icon on the right and select **SIP Authentication**.

🆏 User One [1001 on UC Server]		Edit Active Service
Number of rings: 4	\sim	Locate Active Service
		SIP Authentication

5. Record the **User/login name** and **Password** from the following menu. You will need them when you configure the phone.



Determining the Authentication ID and Password as the Administrator

- 1. Launch the UC client.
- 2. Log in as the admin user.
- 3. Select the **Identities** tab in the left pane.

4. From the menu bar, select **View > Display Identities for all Profiles**.



- 5. Find the identity in the list that you want to use and double-click the entry.
- 6. Select **SIP Authentication**.

Identity		
General inf	ormation	
1	Display name:	User One
	Address:	1001
	Number of rings:	4 SIP Authentication
Call answe	ring	
٩	Send caller to the	Personal Assistant
		OK Cancel Help

7. Record the **User/login name** and **Password** from the following menu. You will need them when you configure the phone.

SIP Authentication for User One [1001 on UC Server]			
SIP auther	SIP authentication information		
\$	SIP authentication information is used to allow your phone to connect to the communications services.		
	If you are manually programming your phone, you will need to enter this information into the phone in order for it to register with the communication system.		
	These fields may have different labels in each phone so be sure to consult your phone documentation.		
	User/login name: 1001		
	Password: 1001		
	OK Cancel Help		

Phone Configuration

The SJphone is configured through the options panel. From there, you can configure an identity.

To configure the phone:

- 1. Select **Menu > Options**, select the **Profiles** tab, and then select **New**.
- 2. In the **Profile name** field, enter the name of the user. Make sure that **Calls through SIP Proxy** is selected for **Profile type** and select **OK**.

😫 Create New	' Profile	×
Profile name:	Red Edison	ОК
File name:	Red Edison.ini	Cancel
Profile type:	Calls through SIP Proxy	Help
Important note Calls through S SIP proxy infor Caller informat	FP Proxy: Profile for a call through a SIP proxy, mation is permanently stored in the profile, ion can be easily changed by re-initializing,	

3. Select the **SIP Proxy** tab. Enter the IP address of the UC server followed by a colon (:) and **5060** into the **Domain/Realm** field.

😫 Profile Options		X
SIP Registration Profile Options	Advanced DTMF Initialization SI	STUN P Proxy
Domain/Realm:	10.10.8.255:5060	
Proxy (URI):		
Proxy usage mode:	Smart	~
📃 Use separate Outbo	ound Proxy for NAT	
NAT Proxy (URI):		
NAT Proxy mode:	Smart	~
	ОК	Cancel

4. Select the **Advanced** tab, enter ***864236245** in the **Voice mail number or address** field, and then select **OK**.

🚏 Profile Options		_	Þ
Profile Options SIP Begistration	Initializatio Advanced	n SIF	P Proxy STUN
Accept redirection	replies 🔽 Us	se "rport" exter	nsion
Use obsolete trans	version 📃 U: sfer mechanism (E	se short header 3YE/Also)	S
Use "standard" st be taken from SIP	atus messages (o 'packets)	therwise messa	ages will
Voice mail number or	address:		
■ 864236245	aracters from pho	ne numbers	
Enable service co	des 🔽 Re	emove service	codes
Fix incoming Cont	act header		
Use Address-Of-R	ecord as Contact	URI	
		ОК	Cancel

5. Enter the user/login name and password from *Gathering Information on page 2* in the **Account** and **Password** fields and select **OK**.

😫 Service: Re	d Edison	×
Please enter th	is information to initialize the service profile —	ОК
Account:	105	Cancel
Password:	••••	
Save service	information permanently	

- 6. Select the Audio tab and select Compression Settings.
- 7. Hold the **Ctrl** key and select both **Microsoft CCITT G.711 A-Law CODEC** and **Microsoft CCITT G.711 u-Law CODEC**. Select **Up** until they are at the top of the list and then select **OK**.

😫 Compression Settings		×
Codec preferences Codec name Microsoft CCITT G.711 A-Law CODEC Microsoft CCITT G.711 u-Law CODEC S1 Labs GSM 6.10 CODEC S1 Labs ILBC CODEC - 30ms S1 Labs ILBC CODEC - 20ms	Status soft soft soft soft soft	OK Cancel Hglp
Down Use Default SJ Labs extensions Lost data recovery Method: SJ Labs data recovery engine	Properties	

8. Select the User Information tab, enter the user's name into the Name field, and select OK.

🛱 Options 🛛 🔀
Skins Interface Neighborhood Support Jabber User Information Call Options Profiles Audio Hot Keys
Name: Red Edison
E-mail:
Location:
Comments:
Use image:
Image should be a 32x32 bmp, png or jpeg file less than 10 Kb.
OK Cancel

Troubleshooting

SJphone has problems placing, receiving, or transferring calls, or has one-way audio issues.

- Make sure that the SJphone is not running on the same computer as the UC server.
- Make sure that both Microsoft CCITT G.711 A-Law CODEC and Microsoft CCITT G.711 u-Law CODEC are at the top of the list as shown in Step 7 under *Phone Configuration on page* 7.
- Make sure that the version of software for the SJphone matches the supported version as indicated in *UC Server Interoperable SIP Device Features and Comparisons* technical note available online at <u>http://kb.adtran.com</u> for your corresponding version of the UC server.