



Configuring

G-Tek

**AQ-101 (aka QuickPhones QA-241) and
AQ-102 (aka QuickPhones QA-242)
version 1110X.27.1.02**

for use with

Nortel

Software Communication System

(SCS) Release 3.0

Task Based Guide

Version: 1.0
Date: 11/01/09

G-Tek
16F, 106, Sec 1, Hsin-Tai 5th Road,
Hsinchih, Taipei County, Taiwan
support@gtek.com.tw
www.gtek.com.tw

Table of Contents

G-Tek AQ-101 (aka QuickPhones QA-241) and AQ-102 (aka QuickPhones QA-242)	5
Overview	5
Requirements	5
Flow Chart	7
Installing/Activating the G-Tek AQ-101/102.....	8
Activation	8
Configuring Account Settings.....	9
Locating the SIP Password	10
Setting User Credentials.....	12
Voicemail Settings	13
Music on Hold Settings.....	14
Configuring and Using Multiple Accounts	15
Enabling Accounts.....	15
Making and Receiving Calls.....	15
Dialing using the Keypad.....	15
Making a Call using a Different Registered SIP Account.....	15
Receiving a Call.....	16
Other Features and Functions	16
Conducting a Three-Way Conference Call	16
Transferring a Call	16
Blocking All Calls Using the Do Not Disturb Feature	16
Checking Voice Messages	17
Referenced Documents	18

G-Tek AQ-101 (aka QuickPhones QA-241) and AQ-102 (aka QuickPhones QA-242)

Overview

The G-Tek AQ-101 (aka QuickPhones QA-241) and AQ-102 (aka QuickPhones QA-242) (hereinafter referred to inclusively as AQ-101/102) are SIP user agent applications that provide VoIP capabilities through an Internet telephony server.

The purpose of this brief guide is to familiarize the reader with the G-Tek AQ-101/102, and to guide the reader through the process of configuring the AQ-101/102 for use with Nortel Software Communication System (SCS) release 3.0.

For comprehensive information on features and functionalities beyond initial configuration and pairing with SCS, please refer to SCS 'End User' guide and AQ-101/102 User Manual.



Fig. 1: AQ-101/102

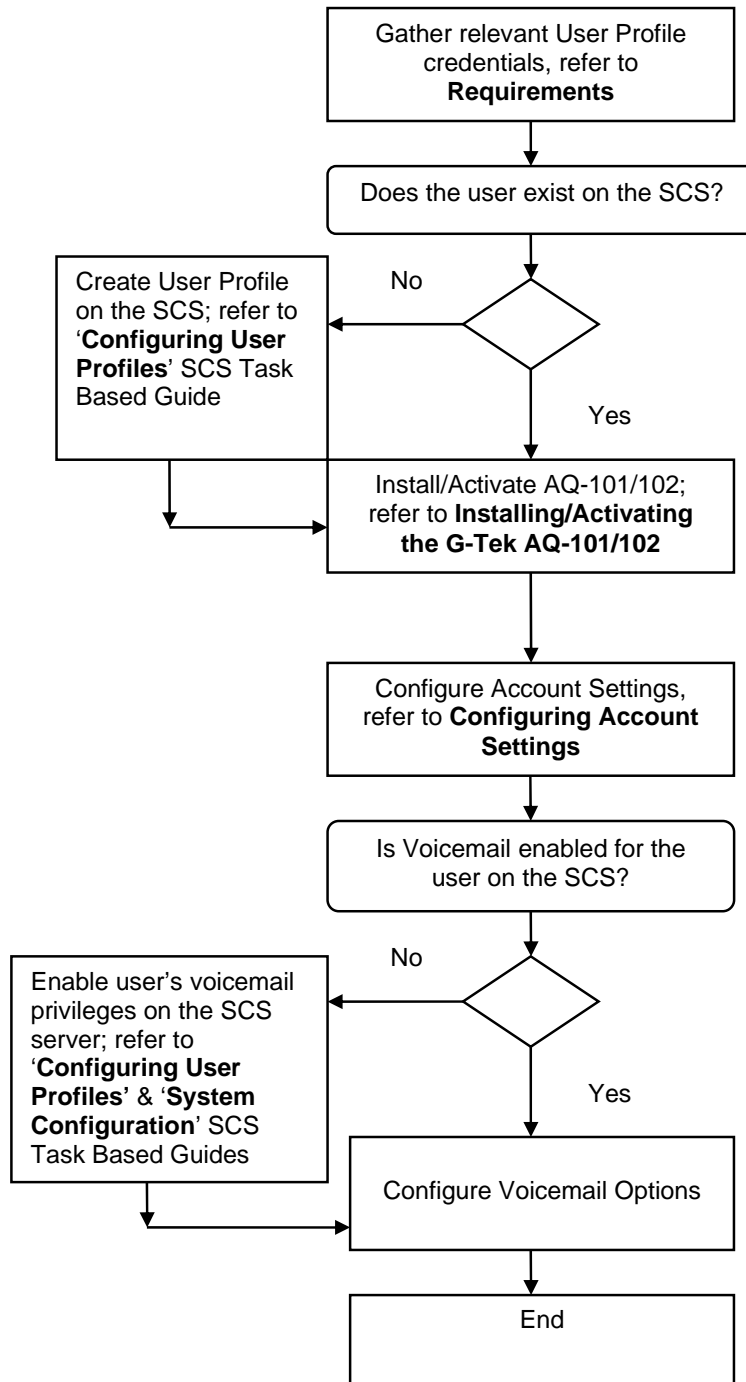
Requirements

The following are the minimum requirements to successfully install and run the G-Tek AQ-101/102.

Before attempting to configure an account on AQ-101/102, ensure that a valid user account has been set up on the SCS platform. The following credentials will be required during AQ-101/102 account configuration:

- The fully qualified domain name of the SCS platform
- A valid SCS User ID
- The SIP Password associated with the User
- Voice Mail Access Number to SCS
- All MAC addresses of the AQ-101/102
- Display name, User Name, Register Name and Password
- Standard Configuration File
- Tool to create multiple configuration files
- A TFTP Server which is pointed at by DHCP option 66

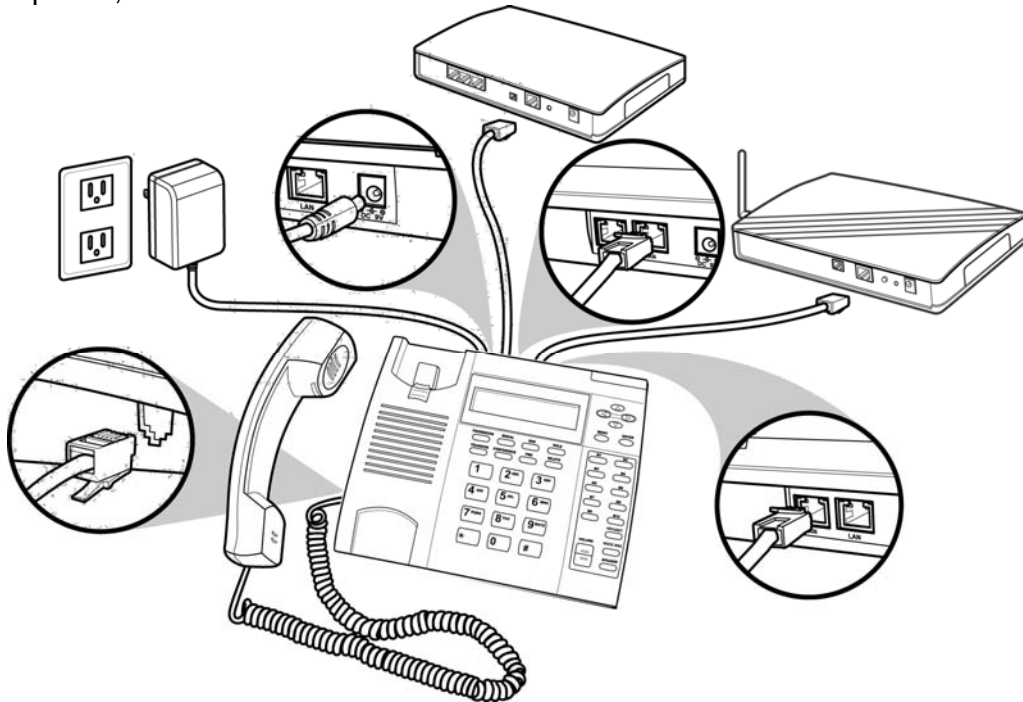
Flow Chart



Installing/Activating the G-Tek AQ-101/102

To install the G-Tek AQ-101/102:

Refer to the following illustration which shows how to connect the VoIP phone to power, LAN and WAN.



1. Power on the phone: You can either use an AC power adapter (AQ-101/102) or PoE (AQ-102 only) to power up the phone.
2. Connect an Ethernet cable to the WAN of the phone.

Activation

After the phone is powered up and has acquired an IP address from the DHCP server, the AQ-101/102 is ready for configuration.

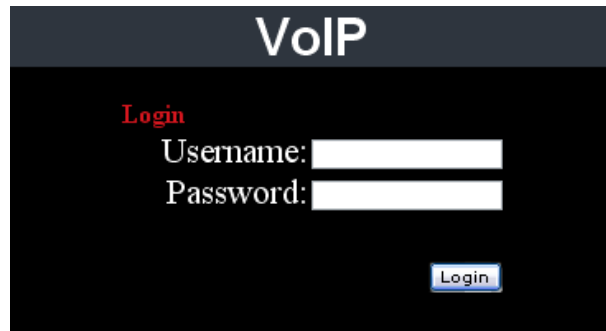
Configuring Account Settings

Note: Before attempting to configure SCS account settings on the AQ-101/102, ensure that a valid user profile exists on the SCS.

If there is a pre-configured file on the TFTP server pointed at by DHCP option 66, the phone will auto configure using this file after boot up. Alternatively the AQ-101/102 can be configured using the Web User Interface or telephone user interface.

To perform configuration using the Web User Interface of the phone:

1. Check the IP address of the AQ-101/102 by pressing **MENU, 4, 5** on the phone.
2. Use a web browser to access the configuration page at <http://IP.Address.of.AQ-101/102:8000>. The login page (shown below) is displayed.
3. Enter the administrator's user name and password and click on **Login**. The administrator's user name and password are "admin" and "1234" respectively.



The following section details the procedure for configuring an SCS user profile on the AQ-101/102. You will need to know the following user and domain information:

- **User SIP Address** – The user name of the SCS account holder in its full SIP address format , e.g., user@scshost.domain.com
- **SIP Password** – The password used by the user profile to initiate communication. This password is usually generated automatically by the SCS during profile creation, for instructions on retrieving the SIP password from the SCS read the next section, 'Locating the SIP Password'.

- **Domain Name** - The SCS server's domain name. If SRV records are not available on the DNS, the fully qualified domain name (host name and domain name, for example **scshost.scsdomain.com**) must be supplied.

If the user does not already have a profile configured on the SCS server, a new profile must be created before continuing. See the 'Configuring User Profiles' SCS Task Based Guide for full instructions on how to create a new profile.

Locating the SIP Password

Note: You will need 'superadmin' user privileges to access the SCS system administrator portal.

In order to successfully configure a link between the AQ-101/102 and the SCS server, the AQ-101/102 must be given a SIP user name and password. Typically, the SIP user name is made up of the SCS **User ID** and the host and domain name, similar to an email address:

abcde@scshost.domain.com

For the purpose of the AQ-101/102 configuration, User ID refers to the value before the '@'.

The SIP password is slightly harder to find. The SIP password is used to register the user's phone with the SIP proxy, and it is therefore important that it is a secure password. Because of this security requirement, the SCS automatically generates an alpha-numeric SIP password when an account is created. The password is then 'hidden' from view when the account profile is opened for editing. To find out a user profile's SIP password:

1. Open a web browser and log in to the SCS administrator portal. See 'Configuring User Profiles'.
2. Place the mouse pointer over the **Users** menu and then select **Users**.



3. Select the required user profile from the list displayed on the screen.

NORTEL Jul 27, 2009 9:52 AM

Home Help Logout Search

USERS Add New User

Filter by

User ID	First Name	Last Name	Aliases
200	chau	nguyen	chau
201	Be	teo	
202	Duong	Thi	
203	Tran	Quang	
205	test1	polycom	
208	Nguyen	Tuyen	
210	giang	nguyen	giang
superadmin			

Software Communication System (4.0.1-015823 2009-05-19T00:09:44)

4. The User Identification screen is displayed. By default, only basic settings are displayed. Click **Show Advanced Settings**.

NORTEL Jul 27, 2009 9:54 AM

Home Help Logout Search

USERS **User: 200** Show Advanced Settings

Existing Groups: administrators

New Groups: You can create new groups simply by adding the new group name to the Groups form value.

Select Phones to add this user to one or more phones.

User ID: 200

Last name: nguyen

First name: chau

Active greeting: default system greeting

E-mail address:

Attach voicemail:

Additional E-mail address:

Attach voicemail:

PIN: [REDACTED]

Confirm PIN: [REDACTED]

Groups:

Aliases: chau

OK Apply Cancel

5. The SIP Password is now displayed among the other settings. Highlight the password, right-click and select **Copy**.

NORTEL Jul 27, 2009 10:03 AM

Home Help Logout Search

USERS DEVICES FEATURES SYSTEM DIAGNOSTICS

Identification
 Phones
 Call Forwarding
 Schedules
 Speed Dial
 ACD Agent Supervisor
 Personal Auto-Attendant
 Conferences
 Registrations
 Permissions
 Caller ID

User: 200 Hide Advanced Settings Existing Groups: administrators

User ID: 200
The User ID can be a numeric extension like "123" or a name like "jsmith". The User ID is displayed by the phone and it is therefore recommended to use the internal extension or the name of the user. If using Direct Inward Dialing (DID), then it is recommended to define the DID number (or its DIDIS portion) as an alias.

Last name: nguyem
First name: chau

Active greeting: default system greeting
Voicemail prompt callers will hear before leaving a message.

E-mail address:
Used for sending notification about new voicemail left for this user. Leave empty to disable e-mail notification.

Attach voicemail:
If checked, the voicemail message will be attached to the notification e-mail. Otherwise, the e-mail will contain a link to retrieve voicemail message.

Additional E-mail address:
Used for sending voicemail message notification to the additional e-mail address.

Attach voicemail:
If checked, the voicemail message will be attached to the notification email sent to the additional e-mail address.

PIN:
Confirm PIN:
The PIN is a password used to log in to voicemail or to the user portal. Numeric PINs are recommended, since only numbers can be dialed.

SIP password: 1234
This password is used by the user's phone to register with the SIP proxy. For phones managed by this system, the SIP password will be configured automatically on the phone. For unmanaged phones, the SIP password is needed for registering lines on the phone. The security of this password is very important and that is why a secure password is generated.

Groups:
Assign this user to one or more groups. If a group does not exist, it will be created. When entering multiple groups, separate them with a comma.

Aliases: chau
Aliases are additional names for the user. Like the user ID, an alias can be either a numeric extension or a name. When entering multiple aliases, separate them with a comma.

6. Login to AQ-101/102 and configure user credentials.

Setting User Credentials

To set the user credentials, select **SIP Settings**→**Service Domain** after logging in to the Web User Interface.

Service Domain Settings

You could set information of service domains in this page.

Realm No.: Realm 1

1.	Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
2.	Display Name:	Hostel 533
3.	User Name:	5331
4.	Register Name:	5331
5.	Register Password:	●●●●●●●●●●●●●●●●
	Domain Server:	nortel.com
	Proxy Server:	47.124.109.21
	Outbound Proxy:	47.124.109.21
	Subscribe for MWI:	<input checked="" type="radio"/> On <input type="radio"/> Off
	Status:	Registered

Submit Reset

1. For Realm No.1, activate the Realm by selecting **On**.
2. Enter the Display Name.
3. Enter the User Name.
4. Enter the Register Name.
5. Enter the Register Password.

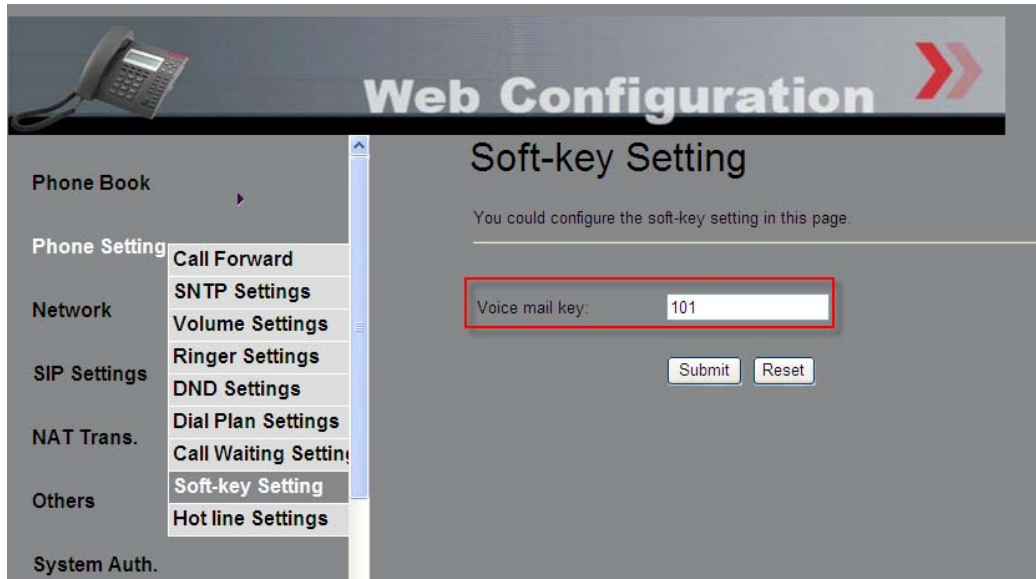
Voicemail Settings

Note: Call forwarding to voicemail is controlled by the SCS. Ensure that voicemail has been activated on the SCS for the user in question.

To set call forwarding to voicemail, see the SCS 'Voicemail Setup and Operation' SCS Task Based Guide. Some settings can be configured within the AQ-101/102 interface.

After logging into the Web User Interface of the phone, select **Phone Settings** → **Soft-key Setting**.

Enter the voice mail extension into the box labelled **Voice mail key:** and click **Submit**. Select **Save Change** → **Save** to save this setting and cause it to take effect.



Music on Hold Settings

To activate Music on Hold, select **SIP Settings** → **Other Settings**.

Enable Nortel Music on Hold by selecting **On** and entering the MoH server using the format sip:~mh~@scs.domain.com. Please refer to the figure below.

The screenshot displays the 'Web Configuration' interface for a phone system. The left sidebar contains a navigation menu with categories: Phone Book, Phone Setting, Network, SIP Settings, NAT Trans., Others, and System Auth. Under 'SIP Settings', sub-menus include Service Domain, Port Settings, Codec Settings, Codec ID Setting, DTMF Settings, RPort Settings, and Other Settings. The main content area is titled 'Other Settings' and contains a list of configuration options. The 'Nortel Music on Hold' option is highlighted with a red rectangular box. It is currently set to 'On' (indicated by a checked radio button) and the server address is entered as 'sip:~mh~@scs.domain.com'. Below this, the 'Exp.' field is set to 'sip:~mh~@domain name'. Other settings visible include Voice QoS (Diff-Serv) at 40, SIP QoS (Diff-Serv) at 40, SIP Expire Time at 600, Use DNS SRV checked, Send Keep Alives Packet unchecked, Keep Alives Period at 60, Prack checked, and Send Update Message unchecked.

Setting	Value	Range/Options
Nortel Music on Hold:	<input checked="" type="radio"/> On <input type="radio"/> Off sip:~mh~@scs.domain.com	Exp. sip:~mh~@domain name
Voice QoS (Diff-Serv):	40	(0~63)
SIP QoS (Diff-Serv):	40	(0~63)
SIP Expire Time:	600	(15~86400 sec, 0=define by Server)
Use DNS SRV:	<input checked="" type="radio"/> On <input type="radio"/> Off	
Send Keep Alives Packet:	<input type="radio"/> On <input checked="" type="radio"/> Off	
Keep Alives Period:	60	(15~250 sec.)
Prack:	<input type="radio"/> On <input checked="" type="radio"/> Off	
Send Update Message:	<input type="radio"/> On <input checked="" type="radio"/> Off	

Configuring and Using Multiple Accounts

Enabling Accounts

There are three supported SIP accounts for each Realm Number associated with an AQ-101/102. To configure the accounts, go to **SIP Settings**→**Service Domain** and select a Realm Number from the pulldown menu. Activate the account by selecting **On**.

Making and Receiving Calls

Dialing using the Keypad

1. Enter the telephone number using the alphanumeric keypad. If you enter an incorrect digit, press the **Delete** key to erase the incorrect digit.
2. After entering the number you want to call, press the **#** key to force dialing to begin immediately, otherwise the number is dialed automatically after 5 seconds.

Making a Call using a Different Registered SIP Account

To select an account:

Press “**1 + * + Speaker**”,
 or “**2 + * +Speaker**”,
 or “**3 + * +Speaker**”.

If the account is registered, the registered number will be displayed.

Receiving a Call

An incoming call is announced by a ringing tone and a red light on the incoming call indicator. Answer the call by doing one of the following:

1. Pick up the **Handset** to answer the call.
2. Press the **Headset** key on the phone to answer the call.
3. Press the **Speaker** key on the phone to answer the call.

Other Features and Functions

Conducting a Three-Way Conference Call

To initiate a three-way conference call, follow the steps below:

1. Call the first participant of the conference call as described in *Making and Receiving Calls*.
2. Put the first participant on hold by pressing the **HOLD** button.
3. Call the extension number of the second participant of the call.
4. Press the **CONFERENCE** key on the phone to include all participants in the call.

To end the three-way conference call, hang up the call.

Transferring a Call

To transfer a call to another phone number, follow the steps below:

Blind Transfer

1. While a call is in progress, press the **HOLD** button followed by the **TRANSFER** button on the phone.
2. Dial the number to which you want to transfer the call followed by the **#** key on the phone. When the call is ringing, press the **TRANSFER** button on the phone to complete the transfer.

Consultative Transfer

1. While a call is in progress, press the **TRANSFER** button on the phone.
2. Dial the number to which you want to transfer the call. After the call is answered, press **TRANSFER** to complete the transfer.

Blocking All Calls Using the Do Not Disturb Feature

The **Do not Disturb (DND)** feature allows you to reject and block all incoming

calls.

To activate the DND function, follow the steps below:

1. Press the DND key on the phone. A **# DND #** message is displayed on the screen.
2. Press the DND key again to disable the DND function.

Checking Voice Messages

If the message indicator on the phone is flashing, a new message is present in your voicemail box.

You can listen to your personal voice message by pressing the **VOICE MSG** key.

Referenced Documents

- SCS Configuring User Profiles Task Based Guide
- SCS System Configuration Task Based Guide
- SCS Voicemail Setup and Operation Task Based Guide
- SCS End User Task Based Guide
- SCS Troubleshooting Task Based Guide