



Configuring

G-Tek

HL-201 (aka QuickPhones QB-241) and HL-202 (aka QuickPhones QB-242) Version 1220X.27.1.02

for use with

Nortel

Software Communication System

(SCS) Release 3.0

Task Based Guide

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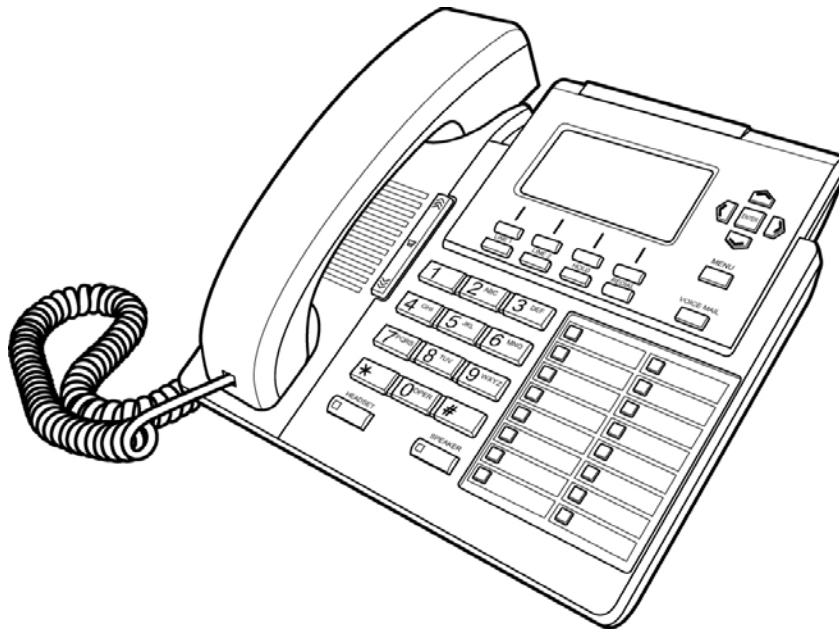
G-Tek HL-201 (aka QuickPhones QB-241) and HL-202 (aka QuickPhones QB-242)

Overview

The G-Tek HL-201 (aka QuickPhones QB-241) and HL-202 (aka QuickPhones QB-242) (hereinafter referred to inclusively as HL-201/202) 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 HL-201/202, and to guide the reader through the process of configuring the HL-201/202 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 HL-201/202 User Manual.



Requirements

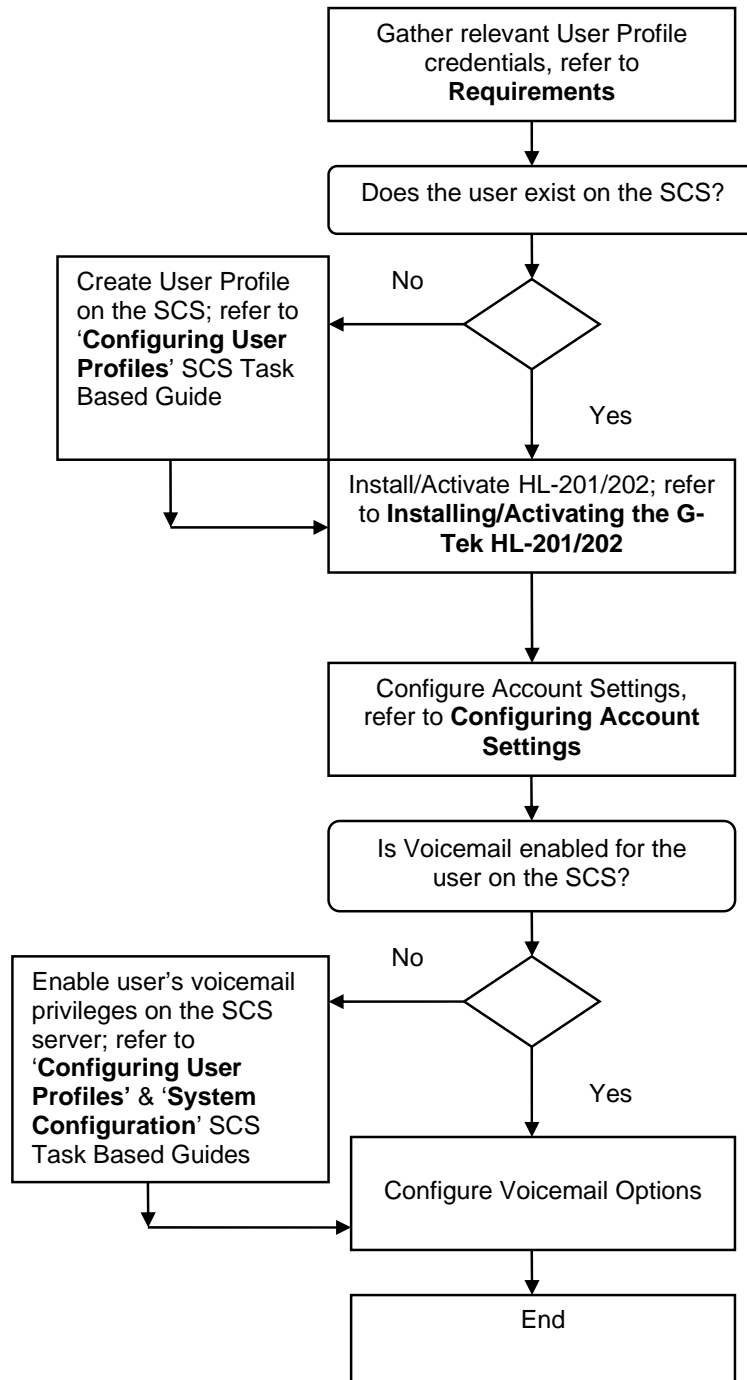
The following are the minimum requirements to successfully install and run the G-Tek HL-201/202.

Before attempting to configure an account on the HL-201/202, ensure that a valid user account has been set up on the SCS platform. The following credentials will be required during HL-201/202 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 HL-201/202
- Display Name, User Name, Register Name and Password
- Standard Configuration File
- Tool for creating multiple configuration files
- A TFTP Server which is pointed at by DHCP option 66

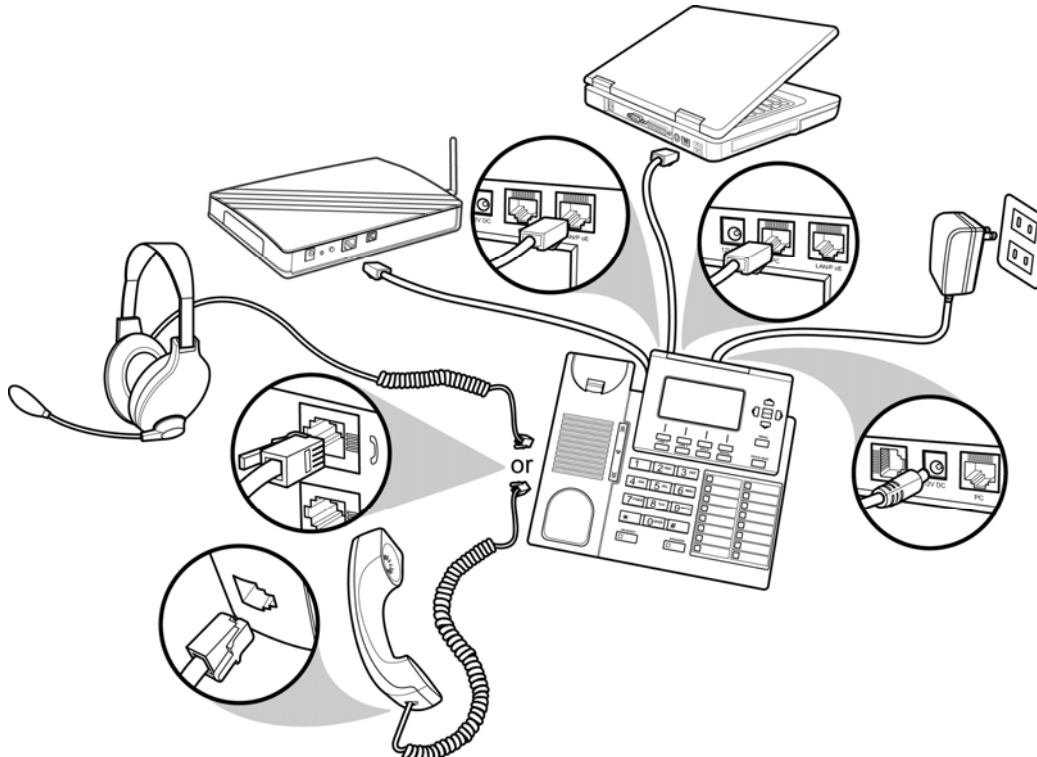
Flow Chart



Installing/Activating the G-Tek HL-201/202

To install the G-Tek HL-201/202:

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



1. Power on the HL-201/202: You can use a power adapter (HL-201/202) or PoE (HL-202 only) to power up the phone.
2. Connect an Ethernet cable to the LAN/PoE of the HL-201/202.

Activation

After the HL-201/202 is powered up and an IP address has been acquired from the DHCP server, the HL-201/202 is ready for configuration.

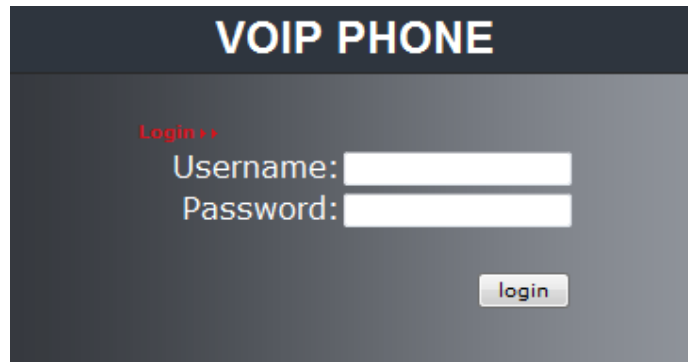
Configuring Account Settings

Note: Before attempting to configure SCS account settings on the HL-201/202, 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 HL-201/202 will auto configure using this file after boot up. Alternatively the HL-201/202 can be configured using the Web User Interface or telephone user interface.

To configure the HL-201/202 using the Web User Interface:

1. Check the IP address of the HL-201/202 by pressing **MENU->9.Network->1234(password)->1.LAN Port Settings->Info** on the phone.
2. Using a web browser, access the configuration page found at [http://IP.Address.of. HL-201/202:8000](http://IP.Address.of.HL-201/202:8000). The login page (shown below) is displayed.
3. Enter the administrator's user name and password and click **login** (the administrator user name and password are "admin" and "1234" respectively).



The following section details the procedure for configuring an SCS user profile on the G-Tek HL-201/202. 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 HL-201/202 and the SCS server, the HL-201/202 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 HL-201/202 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 2009-06-19T00:09:44)

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

Identification User: 200

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

→ 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: *****
 Confirm PIN: *****
 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

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

Select Phones to add user to one or more phones.

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 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 DID 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: [masked]

Confirm PIN: [masked]
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: [masked]
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: [empty]
List the groups to which this user belongs. If a group does not exist, it will be created. When entering multiple groups, separate them with a space.

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 space.

6. Login to HL-201/202 and configure user credentials.

Setting User Credentials

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

SIP Account 1	
1.	Registration: <input checked="" type="radio"/> Enable <input type="radio"/> Disable
2.	Registration ID: 4135
3.	Display Name: 4135
4.	Authentication Name: 4135
5.	Password: ●●●●
	Registration Server: nortel.com
	Proxy Server: 47.124.109.21
	Realm Address: 47.124.109.21
	Voice Mail: 101
	BLF LIST: ~*r~C~4135
	MOH Server: ~*mh~@scs.domain.com
	Expire Time: 300
	DTMF Type: RFC2833
	Ping before register: Disable
	Send KeepAlive: <input type="radio"/> On <input checked="" type="radio"/> Off
	MWI: Enable
	DNS SRV: UDP
	Status: registered

For SIP Account 1, refer to the figure above and:

1. Activate the account by selecting **Enable** for the **Registration** value.
2. Enter the Registration ID.
3. Enter the Display Name.
4. Enter the Authentication Name.
5. Enter the 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 HL-201/202 interface.

After logging in to the Web User Interface, select **SIP Accounts** → **ID 1** → **Voice Mail**.

Enter the voice mail extension into the box labelled **Voice Mail:** and click **Submit**.

SIP Account 1	
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	4135
Display Name:	4135
Authentication Name:	4135
Password:	••••
Registration Server:	nortel.com
Proxy Server:	47.124.109.21
Realm Address:	47.124.109.21
Voice Mail:	101
BLF LIST:	~*r*-C~4135
MOH Server:	~*mh~@scs.domain.com
Expire Time:	300
DTMF Type:	RFC2833
Ping before register:	Disable
Send KeepAlive:	<input type="radio"/> On <input checked="" type="radio"/> Off
MWI:	Enable
DNS SRV:	UDP
Status:	registered

Music on Hold Settings

To activate Music on Hold, select **SIP Accounts** → **ID 1** → **MOH Server**. Enter the MoH server using the format ~*mh~@scs.domain.com. Please refer to the figure below.

SIP Account 1	
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	4135
Display Name:	4135
Authentication Name:	4135
Password:	••••
Registration Server:	nortel.com
Proxy Server:	47.124.109.21
Realm Address:	47.124.109.21
Voice Mail:	101
BLF LIST:	~~rl~C~4135
MOH Server:	~~mh~@scs.domain.com
Expire Time:	300
DTMF Type:	RFC2833
Ping before register:	Disable
Send KeepAlive:	<input type="radio"/> On <input checked="" type="radio"/> Off
MWI:	Enable
DNS SRV:	UDP
Status:	registered

Busy Lamp Field Setting

To activate Busy Lamp Field, select **SIP Accounts** → **ID 1** → **BLF LIST**. Referring to the figure below, enter the BLF LIST using the format `~~rl~C~` "extension number".

SIP Account 1	
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	4135
Display Name:	4135
Authentication Name:	4135
Password:	••••
Registration Server:	nortel.com
Proxy Server:	47.124.109.21
Realm Address:	47.124.109.21
Voice Mail:	101
BLF LIST:	~~rl~C~4135
MOH Server:	~~mh~@scs.domain.com
Expire Time:	300
DTMF Type:	RFC2833
Ping before register:	Disable
Send KeepAlive:	<input type="radio"/> On <input checked="" type="radio"/> Off
MWI:	Enable
DNS SRV:	UDP
Status:	registered

Select **Phone Settings->Speed Dial** and enter the numbers of the extensions which you wish to monitor, and select the Account for each one entered.

Speed Dial

You could set the speed dial phones in thi page.

Mkey	Type	Number	Account	Delete
1	BLF	4200	4135	<input type="checkbox"/>
2	BLF	4300	4135	<input type="checkbox"/>
3	BLF	4500	4135	<input type="checkbox"/>

Configuring and Using Multiple Accounts

Enabling Accounts

There are a maximum of three supported SIP accounts, which can be associated with a single ID, and there are a maximum of 10 IDs per phone. Each SIP account can be configured by selecting **SIP Accounts** followed by the ID number, e.g **SIP Accounts -> ID 2**. After configuring each account, the account must be registered by selecting **Enable**.



Web Configuration

SIP Accounts

You could set information of service domians in this page.

ID	Display Name	Registration Server	Status	Registration	Select
1	1224	scs.gtek.com.tw	registered	Enable	<input type="checkbox"/>
2			OFF	Disable	<input type="checkbox"/>
3			OFF	Disable	<input type="checkbox"/>
4			OFF	Disable	<input type="checkbox"/>
5			OFF	Disable	<input type="checkbox"/>
6			OFF	Disable	<input type="checkbox"/>
7			OFF	Disable	<input type="checkbox"/>
8			OFF	Disable	<input type="checkbox"/>
9			OFF	Disable	<input type="checkbox"/>
10			OFF	Disable	<input type="checkbox"/>

Delete

SIP Account 2	
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	<input type="text"/>
Display Name:	<input type="text"/>
Authentication Name:	<input type="text"/>
Password:	<input type="text"/>
Registration Server:	<input type="text"/>
Proxy Server:	<input type="text"/>
Realm Address:	<input type="text"/>
Voice Mail:	<input type="text"/>
BLF LIST:	<input type="text"/>
MOH Server:	<input type="text"/>

Making and Receiving Calls

Dialing Using the Keypad

1. Enter the telephone number using the alphanumeric keypad of the HL-201/202. If you enter an incorrect digit, press the **Delete** soft key to erase the incorrect digit.
2. After entering the number you wish to call, press the **#** key to force dialing to begin immediately, otherwise the number will be dialed automatically after 5 seconds.

Making a Call Using a Different Registered SIP Account.

1. Press the “Down” key (labelled with a down arrow) or “Up” key (labelled with an up arrow) to choose an account to use. If the account is registered, the Registration ID is displayed.

Receiving a Call

Users are alerted to incoming calls by a ring tone and a red light on the incoming call indicator. Calls can be answered 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 above in ***Making a Call***.
2. Put the first participant on hold by pressing the HOLD button on the HL-201/202.
3. Call the extension number of the second participant of the call.
4. Press the **Conf.** soft 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 **Bxfr** key on the phone.
2. Dial the number to which you want to transfer the call and press **#** on the phone.
3. While the call is ringing, release the call (e.g. hang up the phone) to complete the transfer.

Consultative Transfer

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

Blocking All Calls Using the Do Not Disturb Feature

The **DND (Do not Disturb)** feature allows you to reject and block all incoming calls without ringing your phone.

To activate the DND feature, follow the steps below:

1. Press the **DND** soft key on the HL-201/202. A “DoNotDisturb” message is displayed on the screen and all incoming calls are blocked until you disable the DND function.
2. Press the **DND** soft key again to disable the DND function.

Checking Voice Messages

The “message indicator” on the HL-201/202 flashes to indicate that you have a new message in your voicemail box.

To listen to your voice message, press the **VOICE MAIL** key on the HL-201/202.

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