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DOCUMENT PURPOSE

This user guide describes the basic concept and tasks necessary to use and configure your GT210 phone. This document covers the topics of phone installation, making basic calls and using basic call features.

To learn the advanced features and configurations, please refer to “GT210 Administration Guide”.

THE TERMS AND CONDITIONS OF SOFTWARE

The software embedded in GT210 contains certain third party open source software components which are provided under the terms and conditions designated at

https://mind.bcom.nec.co.jp/customernet/soft-license/ITX-1615-1W_OSS.html.

FIRMWARE VERSION CHECK

GT210 has two FW versions (version 1.0.4.152). Before using this document, please check the GT210 FW version.

1. Enter MENU options. When the phone is in idle, press the round MENU button to enter the configuration menu.
2. Navigate in the menu options. Press the **UP/DOWN/LEFT/RIGHT arrow** keys to navigate in the menu options.
3. Go to **Status → System Status → Software Version → Prog**
4. Check the firmware version.
5. Exit. Press **LEFT arrow** key to exit to the previous menu.
6. The phone automatically exits MENU mode with an incoming call, when the phone is off hook or the MENU mode if left idle for more than 60 seconds.

 **Caution:**

Changes or modifications to this product not expressly approved by NEC, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

 **Caution:**

User ID and password are important information. Be careful when handling them. Default password is given for initial maintenance and operational settings. To enhance the safety, change the default password in the course of the initial settings and periodically thereafter.

 **Warning:**

Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission of NEC Corporation is not permitted.

 **FCC Caution:**

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

PRODUCT OVERVIEW

Model Line Up of GT210 Series

Model Name	Description
ITX-1615-1W(BK)TEL	IP Telephone with 2 line buttons (Black)

Feature Highlights

- 132*48-pixel graphical LCD display.
- 2 dual-color line keys, 3 XML programmable context-sensitive soft keys, 3-way conference, multi-language support.
- Superb full-duplex hands-free speakerphone with advanced acoustic echo cancellation and excellent double-talk performance.
- Large phonebook (up to 500 contacts) and call history (up to 200 records).
- Personalized music ring tone/ring back tone, flexible customizable screen content & format using XML, and advanced Web and enterprise applications.
- Dual switched 10/100 Mbps ports and integrated PoE.
- Automated provisioning using encrypted XML configuration file, SRTP and TLS for advanced security protection, 802.1x for media access control.
- Innovative UI design (easy and useable multi-language support).

Supporting Communication Servers

The following communication servers support GT210.

Communication Server Name	
1	UNIVERGE SV9500
2	UNIVERGE SV9300
3	UNIVERGE SV9100
4	SL2100
5	SL1000/1100
6	UNIVERGE 3C
7	SIP@Net

Technical Specifications (Firmware version 1.0.4.152)

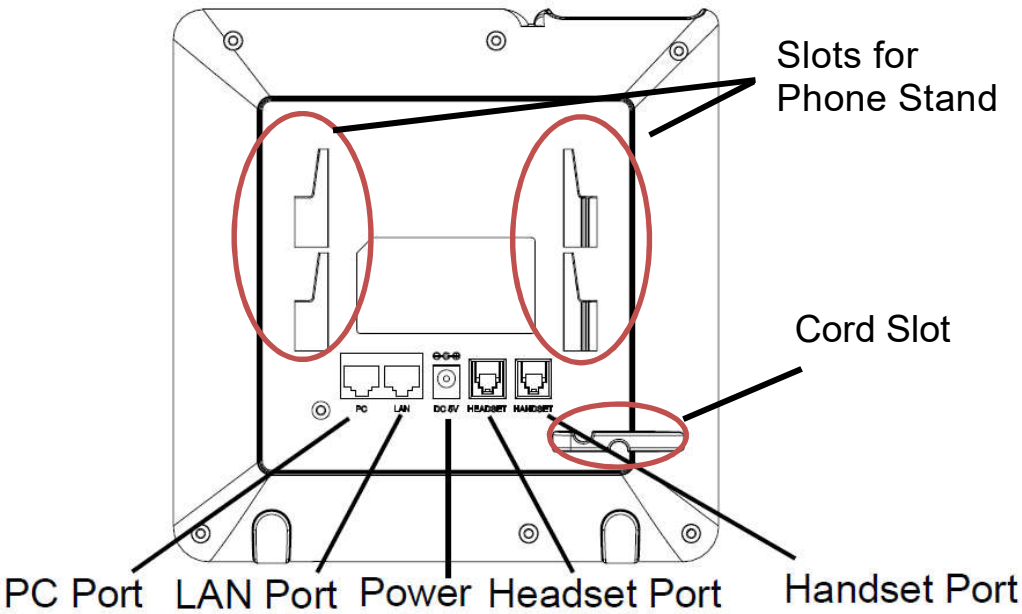
Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP/FTP/FTPS, NTP, STUN, SIMPLE, LLDP-MED, LDAP, 802.1x, SRTP, TLS.
Network Interfaces	Dual switched 10/100 Mbps ports, integrated PoE (IEEE802.3af class 1).
Graphic Display	132*48 LCD display.
Feature Keys	2 line keys with dual-color LED and 1 SIP account.3 XML programmable context sensitive soft keys. 5 (navigation, menu) keys. 13 dedicated function keys for PAGE/INTERCOM, PHONEBOOK, MESSAGE, HOME, HOLD, MUTE, HEADSET, TRANSFER, CONFERENCE, SEND and REDIAL, SPEAKERPHONE, VOLUME.
Voice Codexs	Support for G.711 μ /a, G.722 (wide-band), G.729 A, in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO).
Telephony Features	Hold, transfer,3-way conference, downloadable phone book (LDAP, up to 500 items), call waiting, call history (up to 200 records), off-hook auto dial, auto answer, click-to-dial, flexible dial plan, Hot Desking, personalized music ringtones, server redundancy & fail-over.
Headset Jack	RJ9 headset jack (allowing EHS with Plantronics headsets).
Base Stand	Yes, allow 2 angle positions available, Wall Mountable.
QoS	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS.
Security	User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control.
Multi-language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese, and more.
Upgrade/Provisioning	Firmware upgrade via TFTP / HTTP / HTTPS / FTP / FTPS, mass provisioning using AES encrypted XML configuration file.
Temperature and Humidity	Operation: 0°C to 40°C. Storage: -10°C to 60°C. Humidity: 10% to 90% Non-condensing.
Physical	Dimension: 209mm (L) x 184.5mm (W) x 76.2mm (H) (with handset). Unit Weight: 0.78kg. Package Weight: 0.99kg.
Package Content	Handset with cord, base stand and network cable.
Compliance	FCC: Part 15 (CFR 47) Class B CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1 RCM: AS/NZS CISPR22/24; AS/NZS 60950; AS/NZS 60950.1.

INSTALLATION

Equipment Packaging

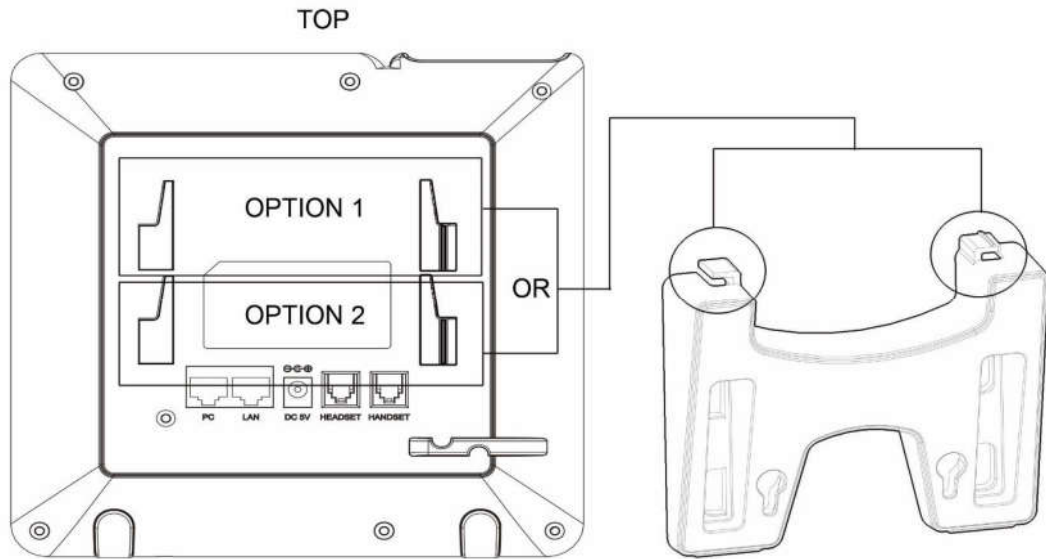
VoIP Phone	1
Handset	1
Phone Cord	1
Phone Stand	1
LAN Cable	1

Connecting Your Phone



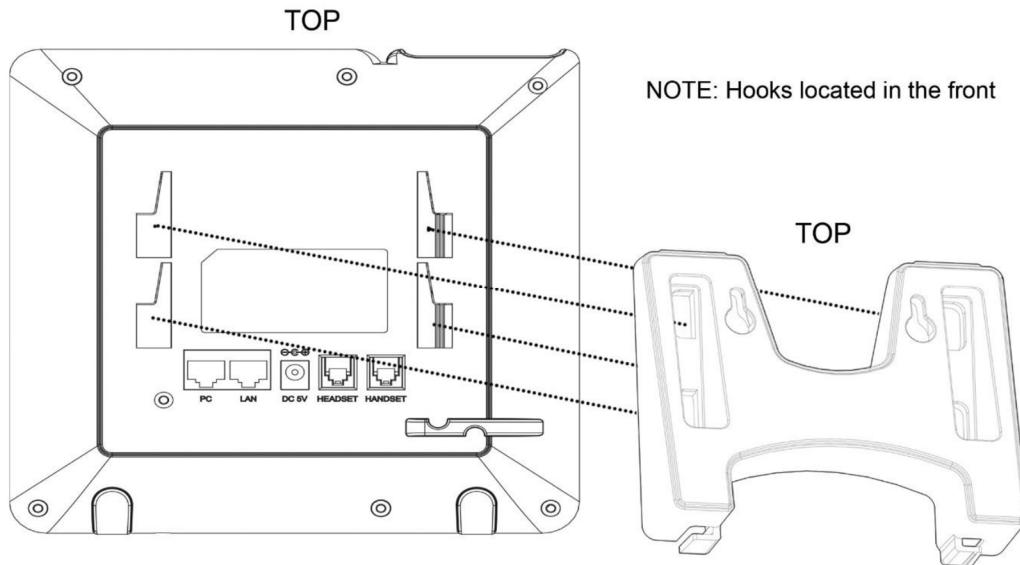
Slots for Phone Stand	Slots for the phone stand if placing the phone on a flat surface
PC Port	10/100M Ethernet to connect PC
LAN Port	10/100M Ethernet to connect LAN integrated PoE:IEEE802.3af class 2,-48V/75mA
DC 5V Power Port	5V/600mA power port to connect to power adaptor
HEADSET Port	To connect to RJ9 or EHS headphones
HANDSET Port	To connect handset RJ9
Cord Slot	To place and fix the phone cords

Stand Installation

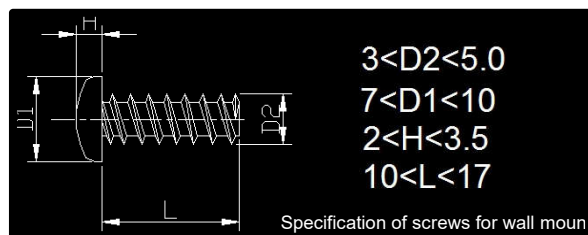


- 1 Insert the hooks on the top of the stand into the slots, you have option to use either upper slots OR lower slots.
- 2 After both hooks are in the slots, firmly slide the entire stand upward to lock them in place.

Wall Mount Installation



1. Insert all 4 hooks into the slots.
2. Firmly slide the wall mount upward to lock it in place.
3. Attach the phone to the wall via the wall mount hole. Refer the specification of screws.



To set up the GT210, follow the steps below:

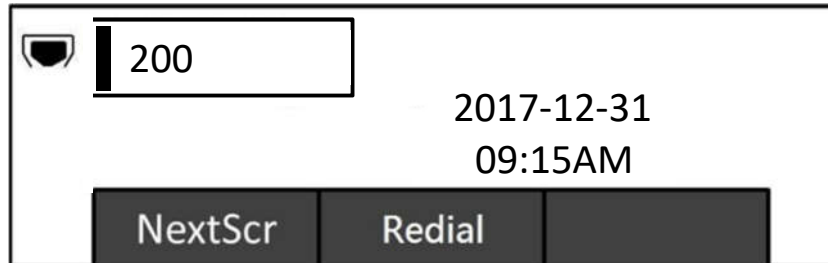
1. Attach the phone stand or wall mount to the back of the phone where there are slots.
2. Connect the handset and VoIP phone with the phone cord.
3. Connect the LAN port of the phone to the RJ45 socket of a PoE switch using the Ethernet cable.
4. The LCD will display provisioning or firmware upgrade information. Before continuing, please wait for the date/time display to show up.

Note: For details, contact the system administrator.

OPERATION (FW version 1.0.4.152)









Idle Screen

GT210 screen displays differently depending on whether the phone is idle or in use (active). User can show/hide softkeys and customize the main screen softkey by assigning a multi-purpose function

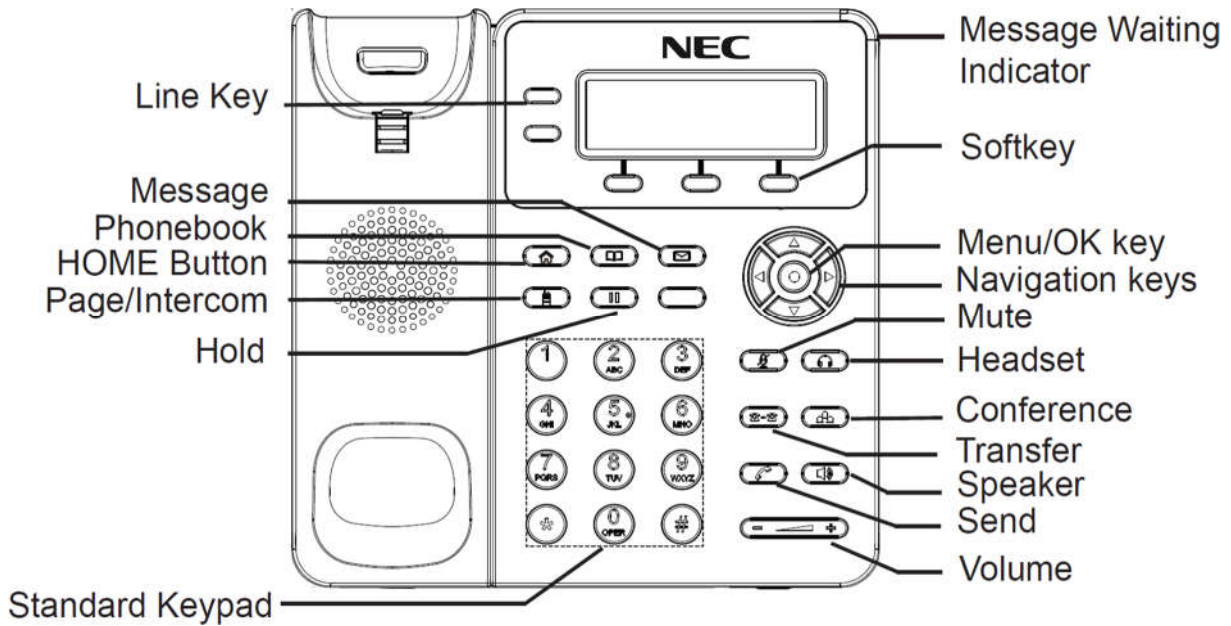







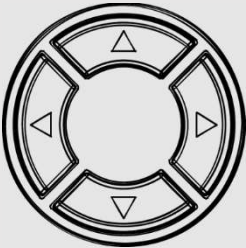
The following table describes the items displayed on the GT210 idle screen.




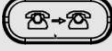



Date and Time	Displays the current date and time. It can be synchronized with Internet time servers.
Network Icon	Shows the status of network. It will indicate whether the network is down or starting.
Status Icon	Shows the status of the phone for registration status, call features and etc., using icons as shown in the next table.
Softkeys in Idle Screen	<p>The softkeys are context sensitive and will change depending on the status of the phone. Typical functions assigned to softkeys are:</p> <ul style="list-style-type: none"> • NextScr Toggles between default idle screen and IP address information. • Redial Redials the last dialed number when there is an existing number in the call log. • Missed Shows unanswered calls to this phone.
Softkeys in Call Screen	<p>The softkeys are context sensitive and will change depending on the call status of the phone. Here are the main softkeys in call screen.</p> <ul style="list-style-type: none"> • Redial Redials the last dialed number after off hook when there is an existing call in the log. • Dial Dials the call out after off hook and entering the number. • Answer Answers the incoming call when the phone is ringing. • Reject Rejects the incoming call when the phone is ringing. • EndCall Ends the active call. • Transfer Transfer softkey will show up after pressing TRANSFER key and entering transfer target number. Press Transfer softkey to do blind transfer. • Split In auto-attended transfer mode, after establishing the second call, press Split to quit transfer and go back to normal talking status. • ConfCall Conferences the active calls. • ReConf Re-establish the conference among the calls on hold.

	<p>Registration Status.</p> <ul style="list-style-type: none"> • Solid: Registered • Blank: Unregistered
	<p>Handset Status.</p> <ul style="list-style-type: none"> • OFF - handset on hook • ON - handset off hook
	<p>Speaker Status.</p> <ul style="list-style-type: none"> • OFF - speaker off • ON - speaker on
	<p>Headset Status.</p> <ul style="list-style-type: none"> • OFF - headset off • ON - headset on
	<p>Mute Status.</p> <ul style="list-style-type: none"> • OFF - The active call is not muted • ON - The active call is muted
	<p>Call Forward Status.</p> <ul style="list-style-type: none"> • OFF - Call Forward feature disabled • ON - Call Forward feature enabled <p>Note) This icon is not displayed even if the Call Forwarding is set by SIP server/PBX.</p>
	<p>DND Status.</p> <ul style="list-style-type: none"> • OFF - Do Not Disturb disabled • ON - Do Not Disturb enabled
	<p>SRTP Status.</p> <ul style="list-style-type: none"> • OFF - SRTP is not used • ON - SRTP is being used

Using the Keypad



<p>LINE Key</p>	<p>Switch between line 1 and line 2. Solid Green: Calling, Talking. Flash Green: Call on hold. Flash Red: Incoming call.</p>
<p>Softkeys</p>	<p>The softkeys are context sensitive and will change depending on the status of the phone.</p>
	<p>MESSAGE Press to view messages.</p>
	<p>PHONEBOOK Press to view phonebook.</p>
	<p>HOME Press to go to idle screen, when stay in submenu.</p>
	<p>PAGE/INTERCOM If the intercom number has been set, it will be called directly. If the intercom number has not been set, pressing this key will have no effect.</p>
	<p>HOLD Place an active call on hold or resume a current call on hold.</p>
<p>Standard Keypad</p>	<p>To dial phone numbers. Press numeric keys "0-9" to enter phone numbers. Press "*" to enter the voice menu, input special characters or long press it to lock the keypad. Press "#" to dial out numbers or input special characters.</p>
	<p>Navigation Keys. While in the menu: Press the UP/DOWN keys to scroll through menu options. Press the LEFT key to go back to the previous menu. The RIGHT key has no function. In Idle: Press the UP key to view missed calls. Press the DOWN key to view dialed calls. Press the RIGHT key to view answered calls.</p>

	<p>MENU/OK <u>In Idle:</u> Press to enter the main menu. Confirm a selection or the current information. <u>Off hook:</u> Press Menu to switch between an outgoing call or a paging call.</p>
	<p>MUTE To mute/unmute an active call. <u>When DND is on:</u> Enable/Disable DND when the phone is in idle.</p>
	<p>HEADSET Press to switch between headset and handset mode in an active call.</p>
	<p>TRANSFER Transfer an active call to another number.</p>
	<p>CONFERENCE Establish 3-way conference with other 2 parties.</p>
	<p>SEND <u>Send.</u> Enter the digits and then press send to dial out. <u>Redial.</u> Press redial when there is a previously dialed call.</p>
	<p>SPEAKER Press to switch between speaker and handset mode in an active call.</p>
<p>Message Waiting Indicator</p>	<p>Message Waiting Indicator. The LED Indicator will flash red when there is new voice mail.</p>

Call Operation List

Operation	SV9500	SV9300	SV9100	SL2100	SL1000 /1100	3C	SIP@NET
Making Phone Calls							
Handset, Speaker and Headset Mode	Yes	Yes	Yes	Yes	Yes	Yes	Yes
SIP Accounts and Lines	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Completing Calls							
On hook dialing.	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Off hook and dial	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Predictive dialing	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Redial	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Via Call History	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Via Phonebook	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Speed Dial from Line Key *1	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Call Return *1	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Via Paging/Intercom *2	No	No	No	No	No	No	No
Via Intercom Hard Button *2	No	No	No	No	No	No	No
Making Calls Using IP Addresses *3	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Quick IP Call Mode *3	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Answering Phone Calls							
Receiving Calls							
Single incoming call	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Multiple incoming calls	No*4	No*4	No*4	No*4	No*4	Yes	Yes
Do Not Disturb	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Reject	Yes*5	Yes	Yes	Yes	Yes	Yes	Yes
During A Phone Call *6							
Call Hold	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Mute	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Forward *7	No	No	No	No	No	No	No
Call Transfer							
Blind Transfer	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Attended Transfer	Yes*8	Yes	Yes	Yes	Yes	Yes	Yes
Auto-Attended Transfer	Yes*8	Yes	Yes	Yes	Yes	Yes	Yes
Conferencing							
Initiate a conference call	Yes*9	Yes	Yes	Yes	Yes	Yes	Yes*10
Cancel Conference	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Hold and Re-conference	Yes	No	Yes	Yes	Yes	Yes	Yes
End Conference	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Easy Conference Mode							
Initiate a conference call	Yes*9	Yes	No	No	No	Yes	Yes*10
Cancel Conference	Yes	No*11	No	No	No	Yes	Yes
Hold and Re-conference	Yes	No	No	No	No	Yes	Yes
End Conference	Yes	Yes	No	No	No	Yes	Yes

*1. When configuring the LINE keys for a function other than Line, Call transfer and Conferencing are not available because they require 2 line keys.

*2. If the called party cannot auto answer the phone will ring normally.

*3. This feature is available only the call between the Standard-SIP Phones.

*4. The Busy Tone is sent to the second calling party.

*5. When a called party presses **Reject**, ringing stops but the calling party continues to hear ring back.

- *6. The GT210 does not have internal hold tone. To provide the hold tone to the phone connected with the SV9100/SL2100/SL1000/SL1100 systems, it is required to change the setting of "Peer to Peer mode" to OFF. For details, contact the system administrator.
- *7. This feature is provided by phone itself however it cannot be used. Please use the similar feature supported by SIP server/PBX.
- *8. For the SV9500, the following system setting is required.
- AUACN : Hold 2nd Call =1 (Intermediate Party is Released)
For details, contact the system administrator.
- *9. When switching from two-party call to three party conference, if the third party is NEC's DT series SIP terminal, the RTP warning tone is heard from the DT series SIP terminal. The RTP Warning Tone can be canceled by the system data settings of SIP server/PBX.
- *10. Dialing the feature access code *30 (add-on) is not available.
- *11. Call Transfer will be performed instead of cancelling a conference call.
- *12. By default, # can be used as SEND key to dial the number out. If the # has must be used as a normal digit, users can disable it by setting "Use # as Dial Key" to "No" from Web GUI→Account 1→Call Settings. In this user guide, if it is written as "press SEND key or #", press the SEND key.

Making Phone Calls

Handset, Speaker and Headset Mode

The GT210 allows users to switch among handset, speaker or headset when making calls. Press the Hook Switch to switch to handset; press the **HEADSET** key to switch to headset; or press the **SPEAKER** key to switch to speaker.

SIP Accounts and Lines

GT210 can support 1 SIP account. Each of the **LINE** keys is "virtually" mapped to an individual SIP account. In off hook state, select an idle line and the dial tone will be heard.

To make a call, select the line you wish to use. The corresponding LINE LED will light up in green. The user can switch lines before dialing any number by pressing the **LINE** keys.

For example:

Incoming calls to a specific account will attempt to use its corresponding LINE if it is not in use. When the "virtually" mapped line is in use, the GT210 phone will flash the other available LINE in red. A line is ACTIVE when it is in use and the corresponding LED is red.

Completing Calls

There are several ways to complete a call.

- **On hook dialing.** Enter the number when the phone is on hook and then send out.
 1. When the phone is idle, enter the number to be dialed out.
 2. Take handset off hook; or
Press **SPEAKER** key; or
Press **HEADSET** key with headset plugged in; or
Select an available **LINE** key.
 3. The call will be dialed out.
- **Off hook and dial.** Off hook the phone, enter the number and send out.
 1. Take handset off hook; or
Press **SPEAKER** key; or
Press **HEADSET** key with headset plugged in; or
Press an available **LINE** key to activate speaker.
 2. You will hear dial tone after going off hook.
 3. Enter the number.
 4. Press **SEND** key or **#** to dial out.
- **Predictive dialing:** While dialing, the phone will predict and list candidates of the target number based on the entered digits. If the target **number** exists in the phonebook/call log, the phone will

display a list of matched numbers and the user could select the number using the **Up/Down** key and dial out.

- **Redial.** Redial the last dialed number.
 1. Take handset off hook; or
Press **SPEAKER** key; or
Press **HEADSET** key with headset plugged in; or
Press an available **LINE** key to activate speaker; or
When the phone is in idle.
 2. Press **SEND** key, or the **REDIAL soft** key.
- **Via Call History.** Dial the number logged in phone's call history.
 1. Press **MENU** key to bring up the main menu.
 2. Enter Call History.
 3. Select the entry you would like to call using the navigation arrow keys.
 4. Press **SEND** key to dial out.
- **Via Contacts.** Dial the number from the Contacts.
 1. Press **MENU** key to bring up the main menu.
 2. Enter "Contacts" using the navigation arrow keys..
 3. Under Contacts, enter Local Phonebook using the navigation arrow key.
 4. Select the contact you would like to call using the navigation arrow key.
 5. Press **SEND** key to dial the selected contact.

Note: Broadsoft Phonebook is not available.
- **Speed Dial from Line Key.** Dial the number configured as Speed Dial on **LINE** Key.
 1. Go to phone's **Web GUI→Settings→Programmable Keys→Programmable Keys**, configure the LINE Key's Key Mode as Speed Dial. Select the account to dial from, enter the Name and User ID (the number to be dialed out) for the LINE Key. Click on "Save and Apply" at the bottom of the Web GUI page.
 2. Go off hook on the phone, or directly press the Speed Dial key to dial out.
- **Call Return.** Dial the last answered call.
 1. Go to phone's **Web GUI→Settings→Programmable Keys→Programmable Keys**, configure the LINE Key's Key Mode as Call Return. Select the account to dial from, no Name or User ID has to be set on for Call Return.
 2. Go off hook on the phone, or directly press the Call Return key to dial out.

- **Via Paging/Intercom.**
 1. Take handset off hook; or
Press **SPEAKER** key; or
Press **HEADSET** key with headset plugged in; or
Press an available **LINE** key to activate speaker.
 2. You shall hear dial tone after off hook.
 3. Press **MENU** key to switch the call screen from "Dialing" to "Paging".
 4. Enter the number.
 5. Press **SEND** key or **#** to dial out.

- **Via Intercom Hard Button.** Dial pre-configured intercom number.
 1. Go to phone's **Web GUI→Settings→Call Features**, configure the Intercom User ID as the destination number of intercom call.
 2. When phone is idle, press the in Intercom Hard Button to dial pre-configured number.

Notes:

- After entering the number, the phone waits for the No Key Entry Timeout (Default timeout is 4 seconds, configurable via Web GUI) before dialing out. Press **SEND** or **#** key to override the No Key Entry Timeout.
- If digits have been entered after handset is off hook, the **SEND** key will work as SEND instead of REDIAL.
- By default, # can be used as **SEND** key to dial the number out. Users can disable it by setting "Use # as Dial Key" to "No" from **Web GUI→Accounts→Account 1→Call Settings**.
- For Paging/Intercom, if the SIP Server/PBX supports the feature and has Paging/Intercom feature code set up already, users do not necessarily need toggle to paging mode in the call screen. Simply dial the feature code with extension as a normal call.

Making Calls Using IP Addresses

Direct IP Call allows two phones to talk to each other in an ad-hoc fashion without a SIP proxy. VoIP calls can be made between two phones if:

- Both phones have public IP addresses; or
- Both phones are on the same LAN/VPN using private or public IP addresses; or
- Both phones can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a direct IP call, please follow the steps below:

1. Press **MENU** key to bring up main menu.
2. Select "Direct IP Call" using the navigation arrow keys.
3. Press **MENU** to enter the Direct IP Call mode.
4. Input the 12-digit target IP address (Please see example below).

5. Press the "More" softkey to make sure the softkey selection "IPv4" or "IPv6" is correctly selected depending on your network environment.
6. Press "OK" softkey to dial.

For example:

If the target IP address is 192.168.1.60 and the port is 5062 (i.e., 192.168.1.60:5062), input the following: 192*168*1*60#5062. The * key represents the dot (.), the # key represents colon (:). Wait for about 4 seconds and the phone will initiate the call.

Quick IP Call Mode

The GT210 also supports Quick IP Call mode. This enables the phone to make direct IP calls using only the last few digits (last octet) of the target phone's IP address. This is possible only if both phones are under the same LAN/VPN. This simulates a PBX function using the CSMA/CD without a SIP server. Controlled static IP usage is recommended.

To enable Quick IP Call Mode, go to phone's **Web GUI**→**Settings**→**Call Features**, set "Use Quick IP - call mode" to "Yes". Clicking on "Save and Apply" on the bottom of Web GUI page to take the change. To make Quick IP Call, take the phone off hook first. Then dial #xxx where x is 0-9 and xxx<255. Press # or **SEND** key and a direct IP call to aaa.bbb.ccc.XXX will be completed. "aaa.bbb.ccc" is from the local IP address regardless of subnet mask. The number #xx or #x are also valid. The leading 0 is not required (but it's OK).

For example:

- 192.168.0.2 calling 192.168.0.3 -- dial #3 followed by # or "**SEND**".
- 192.168.0.2 calling 192.168.0.23 -- dial #23 followed by # or "**SEND**".
- 192.168.0.2 calling 192.168.0.123 -- dial #123 followed by # or "**SEND**".
- 192.168.0.2: dial #3 and #03 and #003 results in the same call -- call 192.168.0.3.

Notes:

- The # will represent colon ":" in direct IP call rather than **SEND** key as in normal phone call.
- If you have a SIP server configured, direct IP call still works. If you are using STUN, direct IP call will also use STUN.
- Configure the "Use Random Port" to "No" when completing direct IP calls. The option "Use Random Port" is under phone's **Web GUI**→**Settings** →**General Settings** page.

Answering Phone Calls

Receiving Calls

- **Single incoming call.** Phone rings with selected ring tone. The corresponding **LINE** key will flash in red. Answer call by taking handset off hook, or using Speaker/Headset, or pressing the flashing **LINE** key.
- **Multiple incoming calls.** When another call comes in while having an active call, the phone will produce a Call Waiting tone (stutter tone). The other **LINE** key will flash in red. Answer the incoming call by pressing the flashing **LINE** key. The current active call will be put on hold automatically.

Do Not Disturb

Do Not Disturb can be enabled/disabled from phone's Keypad Menu by following the steps below:

1. Press the **MENU** key and select "Preference" using navigation keys.
2. Press **MENU** key again to get into Preference options.
3. Select "Do Not Disturb" and press **MENU** key.
4. Use arrow keys to select and press **MENU** key to enable or disable "Do Not Disturb" feature.

When Do Not Disturb feature is turned on, the DND icon will appear on the right side of the LCD. The incoming call will not be accepted.

Reject

Reject the incoming call by pressing the "Reject" softkey when the phone is ringing.

During A Phone Call

Call Hold

- **Hold.** Place a call on hold by pressing the **HOLD** key. The active **LINE** key will blink in green.
- **Resume.** Resume call by pressing the blinking **LINE** key.

Mute

During an active call, press the **MUTE** key to mute/unmute the microphone. The LCD will show the Mute icon on the screen when the call is muted.

Group Listening with Speaker

If the option `g with Speaker` is enabled on the Web GUI (refer to the administration guide), the phone will display a softkey during the call to enable speaker listening along with handset or headset.

Call Transfer

GT210 supports Blind Transfer, Attended Transfer and Auto-Attended Transfer.

- **Blind Transfer.**
 1. During the first active call, press **TRANSFER** key and dial the number to transfer to.
 2. Press **SEND** key or **#** to complete transfer of active call.

- **Attended Transfer.**

[SV9500]

1. During the first active call, press idle **LINE** key. The first call will be put on hold.
2. Enter the number for the second call in the new line and establish the call.
3. Press **TRANSFER** key.

[SV9300]

1. During the first active call, press idle **LINE** key. The first call will be put on hold.
2. Enter the number for the second call on the new line and establish the call.
3. Hang up the call. The call will be transferred.

[SV9100/SL2100/SL1000/SL1100/3C/SIP@NET]

1. During the first active call, press idle **LINE** key. The first call will be put on hold.
2. Enter the number for the second call on the new line and establish the call.
3. Press **TRANSFER** key.
4. Press the **LINE** key which is on hold to transfer the call.

-

- **Auto-Attended Transfer.**

1. Set "Auto-Attended Transfer" to "Yes" under **Web GUI→Settings→Call Features**. And then click "Save and Apply" on the bottom of the page.
2. Establish one call first.
3. During the call, press **TRANSFER** key. A new line will be brought up and the first call will be automatically placed on hold.
4. Dial the number and press **SEND** key or # to make a second call. (Once the number is entered, a "Transfer" soft key will show. If "Transfer" soft key is pressed instead of **SEND** key or #, a blind transfer will be performed).
5. Press **TRANSFER** key again. The call will be transferred;

- **Cancel Auto-Attended Transfer.**

1. For Auto-Attended Transfer, after connecting to the second party, a "Split" soft key will show. To cancel transfer, press "Split" and "EndCall" softkey.
2. Press **LINE** key on hold, will resume first call talking.

Notes:

- To transfer calls across SIP domains, SIP service providers must support transfer across SIP domains.
- The SV9300 cannot **Cancel Auto-Attended Transfer** because "Split" softkey is not available.

Conferencing

GT210 can host up to a 3-way conference call.

- **Initiate a conference call.**

1. Establish calls with 2 parties respectively.
2. While 1 call is active, the other call/calls will be put on hold with its **LINE** key blinking in green.
3. Press **CONFERENCE** key.
4. Press the desired **LINE** key on hold, the conference will be established.

- **Cancel Conference.**


1. If after pressing the **CONFERENCE** key, the user decides not to conference, press Cancel soft key or the current active **LINE** key (LED in solid green);
2. This will resume the 2-way conversation with the current line.

- **Hold and Re-conference.**

[SV9500/SV9100/SL2100/SL1000/SL1100]

1. During the conference, press **HOLD** key.

2. If users would like to resume conference call, press **HOLD** key or blinking **LINE** key.

 **Notes:** This function is not available with SV9300.

[3C/SIP@NET]

1. During the conference, press **HOLD** key. The conference call will be split and the calls will be put on hold separately on the 2 **LINE** keys blinking in green;
2. Select one **LINE** key and press to resume the 2-way conversation;
3. If user is a conference organizer, and would like to re-establish conference call, press a **“ReConf” softkey**. Also, if users are participant of conference, and would like to re-establish conference call, press **HOLD** key or blinking **LINE** key.

- **End Conference.**

Users could press the **“EndCall”** softkey or simply hang up the call to terminate the conference call.

Easy Conference Mode

GT210 supports **Easy Conference Mode**, which can be used combined with the traditional way to establish the conference.

- **Initiate a conference call.**

1. Establish 1 call.
2. Press **CONFERENCE** key and a new line will be brought up using the same account.
3. Dial the number and press **SEND** key to establish the second call.
4. Press **CONFERENCE** key or press the **“ConfCall”** softkey to establish the conference.

- **Cancel Conference.**

1. If users decide not to conference after establishing the second call, press **EndCall** softkey instead of the **“ConfCall” softkey/CONFERENCE** key.
2. This will end the second call and the screen will show the first call on hold.


Notes:

- For SV9300, when **“EndCall”** softkey is pressed, transfer will be performed.

- **Hold and Re-conference.**

[SV9500/SV9100/SL2100/SL1000/SL1100]

1. During the conference, press **HOLD** key.
2. If users would like to resume conference call, press **HOLD** key or blinking **LINE** key.

 **Notes:** This function is not available with SV9300.

[3C/SIP@NET]

1. During the conference, press **HOLD** key. The conference call will be split and the calls will be put on hold separately on the 2 **LINE** keys blinking in green;
2. Select one **LINE** key and press to resume the 2-way conversation;
3. If user is a conference organizer, and would like to re-establish conference call, press a "**ReConf**" softkey. If users are participant of conference, and would like to re-establish conference call, press **HOLD** key or blinking **LINE** key.

- **End Conference.**

Users could press the "**EndCall**" softkey or simply hang up the call to terminate the conference call. If the remote party hangs up the call itself, it will be disconnected from the conference but other parties on GT210 will stay in the existed conference.



Notes [3C/SIP@NET]:

- The party that starts the conference call has to remain in the conference for its entire duration, you can put the party on mute but it must remain in the conversation.
- The option "Disable Conference" has to be set to "No" to establish conference.
- For 3-party conference, if the host hangs up the phone the conference will end. If users want to keep the other 2 parties in conference after the host hangs up, go to **Web GUI→Accounts→Account 1→Call Settings→Transfer on conference Hangup**, check "Yes" and save the change.

Voicemail

A blinking red LED indicator on the top right corner of the GT210 indicates a message is waiting. Go to **Web GUI→Accounts→Account 1→General Settings** to configure **Voice Mail Access Number**.


To retrieve the voicemail:

1. Press **MESSAGE** key to dial into the voicemail box.
2. Select the account and confirm to access the dial-up interface, make operations according to the voice prompt.
3. Enter the account and password which is configured on SIP PBX.

CONNECTING TO DEVICES

EHS Headset

The GT210 supports normal RJ9 headset and Plantronics EHS headset. To use Plantronics EHS headset, go to **Web GUI→Settings→Preferences→Audio Control→Headset Type**. Select "Plantronics EHS" and reboot the phone to take effect.

1. Connect EHS Headset (Plantronics) to GT210. Insert headset connector into the RJ9 headset port on the back of GT210.
2. To use headset mode, press **HEADSET** key. A headset icon  will show on the GT210 status bar. The EHS headset will ring when there is an incoming call.

Notes:

- The GT210 keeps the headset mode on regardless of power off or reboot.
- The following Plantronics headset and cables are supported.

Headset	Cable
CS540	APD-80 + 85638-01 cable
	APV-63

RESTORE FACTORY DEFAULT SETTINGS

Warning:

Restoring the Factory Default Settings will delete all configuration information on the phone. Please backup or print all the settings before you restore to the factory default settings. NEC Corporation is not responsible for restoring lost parameters and cannot connect your device to your VoIP service provider.

Please follow the instructions below to reset the phone:

Restore to Factory Default via Keypad Menu

1. Press **MENU** key to bring up the keypad configuration menu.
2. Select "Config" and enter.
3. Select "Factory Reset".
4. A warning window will pop out to make sure a reset is requested and confirmed.
5. Press the "OK" softkey to confirm and the phone will reboot. To cancel the Reset, press "Cancel" softkey instead.

Restore to Factory Default via Web GUI

1. Login Web GUI.
2. At the top right corner of the web page, click "Factory Reset" key to reset the device.

Standard SIP Terminal
IP Phone GT210
User Guide

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