

Objectworld™

Objectworld Unified Communications Server®

Grandstream BudgeTone-200 SIP Phone User Guide

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1 Grandstream BudgeTone-200 SIP Phone Overview

This document is designed to outline the features and functions of the Grandstream BudgeTone-200 SIP in relation to the Objectworld Unified Communications Server™. This chapter covers the following topics:

- [Overview](#)
- [Display Features](#)
- [Key Descriptions](#)

1.1 Overview

This document is not intended to replace the Grandstream BudgeTone-200 SIP Phone User Manual. For more detailed information about the product, particularly its operation in the absence of UC Server, please refer to the Grandstream User Manual.



NOTE: For information on using your telephone mailbox see the *Quick Reference Card* available on the Objectworld web site at www.objectworld.com/support/documentation.

1.2 Display Features

The BT-200 phone has a numeric LCD of 64 mm x 24 mm size with backlight. This model has a small, red LED status reminder. Here is the display when all segments illuminate:



When the phone is in the normal idle state, the backlight is off. When a call occurs, the backlight turns on automatically to bring to the user's attention. In addition, if Voice Mail is configured and there is a voice mail message waiting, the blinks to remind users there is a voice mail in the Voice Mail server.

Icon LCD Icon Definitions



Connectivity Status / SIP Proxy/Server Icon

Off – IP address of SIP server is not found
On – IP address of SIP server are located
Blinking – Ethernet link failure or the phone is not registered properly



Phone Status Icon

Off – When the handset is on-hook
On – When the handset is off-hook



Speakerphone Status Icon

Flash – When phone rings
Off – When the speakerphone is off
On – When the speakerphone is on



Handset and Speakerphone / Headset Volume Icon

0-7 scales to adjust handset / speakerphone volume



Real-time Clock

Synchronized to Internet time server
Time zone configurable via web browser
AM/PM indicator



Call Logs

01-10 for CALLED history (dialed number)
01-10 for CALLERS history (Incoming caller ID)



Time Icon

AM for the morning
PM for the afternoon



IP Address Separator Icon

Icon LCD Icon Definitions



Numerical Numbers and Characters:

0 - 9

* = +

= +

A, b, C, c, d, E, F, G, g, H, h, I, L, n, O, o, P, q, r, S, t, U, u, Y

1.3 Key Descriptions



Key Button


Key Button Definitions

0 - 9, *, #

Digit, star and pound keys are usually used to make phone calls



1. Reduce handset, speakerphone/headset volume after off hook the phone via handset or speaker
2. Reduce ring tone volume when phone in IDLE and off hook to confirm the changed ring tone volume
3. Next menu item browsing when phone is in IDLE mode after MENU key pressed, off hook to interrupt and exit

Key Button	Key Button Definitions
	<ol style="list-style-type: none">1. Increase handset, speakerphone/headset volume after off hook the phone via handset or speaker2. Increase ring tone volume when phone in IDLE and off hook to confirm the changed ring tone volume3. Previous menu item browsing when phone is in IDLE mode after MENU key pressed, off hook to interrupt and exit
MENU	Enter keypad MENU mode when phone is in IDLE mode. It is also the ENTER key once entering MENU
CALLED	After off hook, press to display the dialed numbers. When number displayed, pressing the SEND key can make call using that displayed number.
CALLERS	After off hook, press to display the incoming Caller IDs. When number displayed, pressing the SEND key can make call using that displayed number.
MESSAGE	Enter to retrieve voice mails from Voice Mail Portal or Server
HOLD	Temporarily hold the active call
TRANSFER	Transfer the active call to another party
CONFERENCE	Establish 3-way conferencing call
FLASH	Flash event to switch between two lines
MUTE/DEL	Mute an active call; or Delete a key entry Also used to 'REJECT' incoming call.
SEND /(RE)DIAL	Dial a new number entered or Redial the number last dialed. After entering the phone number, pressing this key forces a call to go out immediately before timeout.
SPEAKER	Enter hands-free mode using the speakerphone

2 Using BudgeTone-200 Features

This chapter outlines the features that are available on the Grandstream BudgeTone-200 SIP telephone. UC Server supports all the telephone features that are outlined in this chapter, unless otherwise noted. For more information on the BT200 keys see:

UC Server supports the following main features of the Grandstream BudgeTone-200 SIP listed below, unless otherwise noted.

- [Using the BT200 Call Feature Access Codes](#)
- [Making Calls](#)
- [Answering Calls](#)
- [Holding and Resuming Calls](#)
- [Muting and Unmuting Calls](#)
- [Conferencing](#)
- [Transferring Calls](#)
- [Ending Calls](#)
- [Redialing](#)
- [Forwarding Calls](#)
- [Changing the Volume](#)
- [Selecting a Ring Tone](#)
- [Accessing Call Lists](#)
- [Accessing Voice Mail](#)

2.1 Using the BT200 Call Feature Access Codes

To access various call features, key press sequences are provided by the BudgeTone-200 phone.



NOTE: The Grandstream feature access codes listed below are disabled by default, due to a conflict with one of the UC Server feature access codes. Therefore, the codes below will only work if the system administrator has enabled the Grandstream feature access codes.

The BT200 phone supports a number of call features including:

- Block Caller ID (or Anonymous Call)
- Call Waiting (Disable/Enable)
- Call Forwarding all Calls, when Busy, or when Delayed

BT200 Call Features	Key Codes	Examples and Notes
Block Caller ID (for all subsequent calls)	*30	
Send Caller ID (for all subsequent calls)	*31	
Block Caller ID (per call)	*67	
Send Caller ID (per call)	*82	Note: This feature code conflicts with the <i>Call Queuing</i> feature code of UC Server. Therefore, your system administrator may have changed the <i>Send Caller ID</i> feature code number. See your system administrator for more information.
Disable Call Waiting (per Call)	*70	
Enable Call Waiting (per Call)	*71	
Forward all Calls	*72 +	To forward all calls to extension 300:
Forwards all calls unconditionally.	Forward Number + #	1. Dial: *72 2. At the dial tone, dial: 300 3. Press # 4. At the dial tone, hang up.
Cancel Forwarding all Calls	*73	1. Dial:
Cancels forwarding all calls unconditionally.		*73 2. At the dial tone, hang up.
Forward Calls when Busy	*90 +	To forward calls when busy to extension 300:
	Forward Number + #	1. Dial: *90 2. At the dial tone, dial: 300 3. Press # 4. At the dial tone, hang up.
Cancel Forwarding Calls when Busy	*91	1. Dial: *91 2. At the dial tone, hang up.

BT200 Call Features	Key Codes	Examples and Notes
Forward Calls when Delayed	*92 + Forward Number + #	To forward calls when delayed to extension 300: 1. Dial: *92 2. At the dial tone, dial: 300 3. Press # 4. At the dial tone, hang up.
Cancel Forwarding Calls when Delayed	*93	1. Dial: *93 2. At the dial tone, hang up.

2.2 Making Calls

For information on making calls, see the following sections:

- [Handset, speakerphone and headset mode](#)
- [Making a call using a phone number](#)
- [Making a call using an IP address](#)

Handset, speakerphone and headset mode



Speakerphone status icon

The regular Handset mode can be switched with either the Speaker mode (Hand free) or the Headset mode, however, whenever the Headset is plugged in, Speaker mode will be switched to the Headset mode automatically.

To switch between Handset and Speakerphone/Headset

- Press the Hook Flash in the Handset cradle or the SPEAKER button.

Making a call using a phone number

There are several ways to make phone calls:

To make a call

- Pick up the handset or press the **SPEAKER** button, and then enter the phone number.



TIP: If you make a mistake dialing a number, press **MUTE/DEL** to delete the last digit entered.

Other methods of placing calls are:

- Press the **SEND** button to redial the last number called.

Once pressed, the last dialed number is displayed on the LCD as the corresponding DTMF tones are played out and an outgoing call is sent.

- Browse the CALLED/CALLER history and press the SEND/REDIAL button.

Pick up the handset or press the **SPEAKER** button, then press the **CALLED/CALLERS** button to browse through the last 10 numbers dialed. Once the desired number is identified and displayed on the LCD screen, press the **SEND** button and a new call to that displayed number is sent out immediately.

Examples:

- To dial another extension on the same proxy, such as 1008, pick up the handset or press **SPEAKER**, dial 1008 and then press the **SEND** button.
- To dial a PSTN number such as 6266667890, you might need to enter in some prefix number followed by the phone number. Check with your VoIP service provider to get the information. If your phone is assigned with a PSTN-like number such as 6265556789, most likely you follow the rule to dial 16266667890 as if you were calling from a regular analog phone, followed by pressing the **SEND** button.

Making a call using an IP address

Direct IP calling allows two parties, that is, a BudgeTone phone and another VoIP device, to talk to each other in an ad hoc fashion without a SIP proxy. This kind of VoIP call can be made between two parties if:

- Both the BudgeTone phone and other VoIP Device (that is, another IP Phone or BudgeTone SIP phone or other VoIP unit) have public IP addresses, or
- Both the BudgeTone phone and other VoIP Device are on the same LAN using private or public IP addresses, or
- Both the BudgeTone phone and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

This model has the ability to dial an IP address under the same LAN segment by simply pressing the last octet in the IP address.

In the Advanced Settings page there is an option "Use Quick IP-call mode", which by default it is set to NO.

When this option is set to YES, and #XXX is dialed, where X is 0-9 and XXX <=255, the phone makes a direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of the subnet mask.

#XX or #X are also valid so the leading 0 is not required, but is okay.

For example:

192.168.0.2 calling 192.168.0.3 just dial #3 follow by SEND or #
 192.168.0.2 calling 192.168.0.23 just dial #23 follow by SEND or #
 192.168.0.2 calling 192.168.0.123 just dial #123 follow by SEND or #

192.168.0.2 dial #3 and #03 and #003 has same effect --> call 192.168.0.3



NOTE: If you have a SIP Server configured, Direct IP to IP calls will work. However, if you are using STUN, Direct IP to IP calls will also use STUN. Alternatively, follow the instructions below to make a direct IP to IP call.

To make a direct IP to IP call

- 1 With the phone off hook, press the **MENU** key.
- 2 Enter a 12-digit target IP address to make the call.
- 3 If the port is not the default 5060, then destination ports can be specified by using “*4” (encoding for “:”) followed by the port number.

Examples:

- If the target IP address is 192.168.0.10, the dialing convention is
MENU_key 192 168 000 010
 followed by pressing the **SEND** key or wait for seconds in the No Key Entry Timeout.
- If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:
MENU_key 192168001020*45062
 followed by pressing the **SEND** key wait for seconds in the No Key Entry Timeout.

Quick IP Call Mode

The BT200 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-number.

This is possible only if both phones are in under the same LAN/VPN. This simulates a PBX function using the CMSA/CD without a SIP server. Controlled static IP usage is recommended.

To enable Quick IP calls, the phone has to be setup first, through the web-setup function.

To set up the phone to make Quick IP calls

- 1 Access the web-setup function.
- 2 In the Advanced Settings page, set the Use Quick IP-call mode to YES. When #xxx is dialed, where x is 0-9 and xxx <=255, a direct IP call to aaa.bbb.ccc.XXX is completed. “aaa.bbb.ccc” is from the local IP address *regardless* of the subnet mask. The numbers #xx or #x are also valid. The leading 0 is not required, but is okay.

For example:

192.168.0.2 calling 192.168.0.3 -- dial #3 follow by SEND or #

192.168.0.2 calling 192.168.0.23 -- dial #23 follow by SEND or #

192.168.0.2 calling 192.168.0.123 -- dial #123 follow by SEND or #

192.168.0.2: dial #3 and #03 and #003 results in the same call -- call 192.168.0.3



NOTE: If you have a SIP Server configured, a Direct IP to IP still works. If you are using STUN, the Direct IP to IP call also uses STUN. Configure the Use Random Port to NO when completing Direct IP calls.

2.3 Answering Calls



NOTE: To forward an incoming call directly to voice mail, press **MUTE/DEL** to reject the call and forward it to voice mail.

To answer a call

- Pick up the handset to answer the call normally.



TIP: Alternatively, press the **SPEAKER** button to answer a call in either speakerphone or headset mode.

Using call waiting to answer a call

If call waiting is enabled, then while you are on a call, you will hear a stutter tone if there is another incoming call.

To answer an incoming call while on another call

- 1 Press the FLASH button to put the current caller on hold and switch to the other call.
- 2 Press the FLASH button again to toggle between two active calls.

2.4 Holding and Resuming Calls

To place a call on hold

- While on a call, press the **Hold** button to put the remote end on hold.

To resume a call on hold

- Press the **Hold** button again to release the previous Hold state and resume the call.

To park a call at a selected park number

- 1 Press the FLASH button.



NOTE: If your phone has more than one identity assigned to it, then **Transfer** the call instead of putting it on hold.

- 2 Dial ***99**.

- 3 Enter the number that you want to use as the park number.

The hold position is used by the person who answers the parked call. For example, if you enter 1234, then the call is parked at 1234, and can be picked up at 1234.

- 4 Press **DIAL**.

The call is removed from the hold position, and put in the park position.

To answer a parked call

- 1 Typically, a company announcement informs you that a call is on hold for you.

- 2 Dial ***98**.

The system prompts you for the park hold position.

- 3 Dial the number of the park hold position.

The system disconnects the current call and presents a new call.

- 4 Answer the new call to speak to the parked caller.



NOTE: If nobody answers the parked call, the call rings the extension that originally put the call on system park.

2.5 Muting and Unmuting Calls

You can mute an active call so that the other party cannot hear you.

To mute a call

- 1 Press **MUTE/DEL** to mute a call.
- 2 Press **MUTE/DEL** again to resume the conversation.



NOTE: When dialing a number, pressing MUTE/DEL deletes the last digit entered. When receiving an incoming call, pressing MUTE/DEL rejects the call and forwards it to voice mail.

2.6 Conferencing

The BudgeTone 200 phone supports a 3-way conference. Follow the instructions below if you are on a call with one party, and you wish to bring a second party into the call as a conference call.

To initiate a conference call

- 1 Press the **CONFERENCE** button.
This puts the first party on hold and gets a dial tone.
- 2 Dial the number for the second party, and then press **SEND** to make the call.
- 3 If the second party answers the call, then press the **CONFERENCE** button to bring both parties into the conference.
- 4 If the second party does not answer the call, then press **FLASH** to resume the call with the first party.



NOTE: During the conference, if either party drops the call, then the remaining two parties can continue talking. However, if the conference initiator hangs up, then all calls are terminated.



NOTE: For information on the UC Server conference bridge see [“Using the UC Server Conference Bridge” on page 22.](#)

2.7 Transferring Calls

Two transfer operations are supported, blind and attended transfers.

Blind Transfer Transfers an active call to a third party without announcing the call.

Attended Transfer Transfers an active call to a third party with an announcement.

To transfer a call blindly

- 1 Press the **TRANSFER** button.

If the other voice channel is available (that is, there is no other active conversation besides the current one), you will hear a dial tone.

- 2 Dial the third party's phone number then press **SEND**.



NOTE: The Enable Call Feature has to be configured to Yes in the Web configuration page in order to make this feature work.

You can hold on to the phone and wait for one of the three following behaviors:

- A quick confirmation tone (temporarily using the call waiting indication tone) followed by a dial tone. This indicates the transfer was successful. You can hang up or make another call.
- A quick busy tone followed by a restored call (on supported platforms only). This means the transfer failed due to the failed response sent from the server, and the phone will try to recover the call. The busy tone indicates to the transferrer that the transfer has failed.
- Busy tone keeps playing. This means the phone has failed to receive the final response and has timed out. This does not indicate the transfer has been successful, nor does it indicate the transfer has failed.

To make an attended transfer

- 1 Press the **FLASH** button.

You will hear a dial tone.

- 2 Dial the third party's phone number then press the **SEND** button.

- 3 If the call is answered, speak to the recipient, and then press **TRANSFER** to complete the transfer and hang up.

- 4 If the call is not answered, press the **FLASH** button to resume the original call.



NOTE: When an attended transfer fails, if you hang up, the BudgeTone phone rings you back again to remind you that the caller is still on the call. Pick up the phone to restore conversation with the caller.

2.8 Ending Calls



Phone status icon

To end a call

- To end a call, hang up the phone.

The phone status icon on the display will be off, when the handset is on-hook.

To end a call on speakerphone

- Press the **SPEAKER** button.

2.9 Redialing

To redial the last number dialed

- Press the **SEND** button.

2.10 Forwarding Calls



NOTE: The Grandstream call feature codes are disabled by default. Only if the call feature codes have been enabled by your system administrator will the call forwarding features work as per the instructions below. If the Grandstream call feature codes are enabled, then you can forward all calls, or forward only those calls when busy or delayed.

To forward all calls

- 1 Dial ***72**.
- 2 At the dial tone dial the number that you want to forward your calls to.
- 3 Press **#**.
- 4 At the dial tone hang up.

To cancel forwarding all calls

- 1 Dial ***73**.
- 2 At the dial tone hang up.

To forward calls when busy

- 1 Dial ***90**.
- 2 At the dial tone dial that the number that you want to forward your calls to.
- 3 Press **#**.
- 4 At the dial tone hang up.

To cancel forwarding calls when busy

- 1 Dial ***91**.
- 2 At the dial tone hang up.

To forward calls when delayed

- 1 Dial ***92**.
- 2 At the dial tone dial the number that you want to forward your calls to.
- 3 Press **#**.
- 4 At the dial tone hang up.

To cancel forwarding calls when delayed

- 1 Dial ***93**.
- 2 At the dial tone hang up.

To forward an incoming call directly to voice mail

- When receiving an incoming call, press **MUTE/DEL** to reject the call and forward it to voice mail.

2.11 Changing the Volume



Handset and
Speakerphone/Headset
volume icon

Use the arrow keys to adjust the volume.

To change the ring tone volume when the phone is idle

- Press the arrow keys up or down to adjust the ring tone volume from 0 (off) to 7 (maximum).

To adjust the handset, speakerphone, and headset volume when on a call

- Press the arrow keys up or down.

On the phone display the volume icon shows a 0-7 volume scale.

2.12 Selecting a Ring Tone

Different ring tone options are available from the BT200 phone menu. You may want to select a different ring tone to distinguish between your phone and a co-worker's phone when ringing.

To select a ring tone

- 1 Press the **MENU** button.
- 2 Press the arrow keys to select option 10 ring 0.
- 3 Press **MENU** to hear the selected ring tone.
- 4 Press the arrow keys to select a ring tone.

- 5 Press **MENU** to confirm the selection.

2.13 Accessing Call Lists

You can view a list of numbers previously dialed. You can also view a list of callers.

To view a list of dialed numbers

- 1 Take the phone off hook.
- 2 Press **CALLED** to display the dialed numbers.
- 3 To call a displayed dialed number, press **SEND**.

To view a list of callers

- 1 Take the phone off hook.
- 2 Press **CALLERS** to display a list of incoming Caller IDs.
- 3 To call a displayed number, press **SEND**.

2.14 Accessing Voice Mail

To access voice mail

- When BudgeTone-200 is on-hook, press the **MESSAGE** button to retrieve voice mails.

The MWI (Message Waiting Indicator) LED flashes in red when the voice mail server sends the message waiting information to BudgeTone-200.

3 Using UC Server Features

This chapter outlines the phone features that are available with UC Server. The UC Server call feature access codes are summarized in the following section:

- [Using the UC Server Call Feature Access Codes](#)

UC Server supports the following features listed below.

- [Parking Calls and Picking up Parked Calls](#)
- [Picking up Calls Ringing at Another Extension](#)
- [Using the UC Server Conference Bridge](#)
- [Queuing Calls](#)
- [Paging](#)

3.1 Using the UC Server Call Feature Access Codes

To access various call features, key press sequences are provided by UC Server.



NOTE: Some features may be unavailable depending on the particular system configuration. See your system administrator for more information.

UC Server Call Features	Key Codes	Examples and Notes
Transferring Calls <i>Transfer a call to another user's voice mail</i> Allows you to transfer a current call to another user's voice mail/PA/PBA service. This features is useful when the intended recipient is away from their desk or out of the office. Note: The extension dialed must be at least 3 digits long.	*86 + extension	To transfer to extension 300's voice mail, dial: *86300

UC Server Call Features

Transfer a call to a call queue
 Use this feature to transfer a caller to a preconfigured departmental call queue. If no agents logged into the queue are available to take the call, the caller is put on hold until an agent is available.

Key Codes

*80 + queue#

Examples and Notes

To enter queue 500, dial:
*80500

To transfer a caller to queue 500:
 1. Press **Transfer**
 2. Dial *80500
 3. Press **Send**

Queuing Calls

Log into a call queue
 Allows you to start receiving calls for the specified queue.

*81 + queue#

To login as an agent to receive calls directed to queue 500, dial:
*81500

Log out of a call queue
 Allows you to stop receiving calls for the specified queue.

*82 + queue#

To logout as an agent to no longer receive calls for queue 500, dial:
*82500 **Note:** If you are using a Grandstream phone, then this feature code may conflict with the *Caller ID* feature code of Grandstream, if your system administrator has enabled the Grandstream feature access codes. If this is the case, then your system administrator may have changed the *Log out of a call queue* feature code number. See your system administrator for more information.

Temporarily Stop Accepting Queued Calls
 Temporarily makes yourself unavailable for accepting queued calls, while not resetting the longest idle agent status.

*83 + <1 - 9>

Dial:
*83
 Press a number from 1 to 9 to indicate your reason for being unavailable.

Reset agent status to available
 Make yourself available again to accept queued calls

*83 + 0

Dial:
*83
 Press "0" at the prompt.

Check agent status in a call queue
 Check your agent status to verify whether you're currently logged into a call queue, temporarily unavailable, or currently logged out of a call queue.

*84 + queue#

To check your agent status in a call queue with queue number 500, dial:
*84500

UC Server Call Features	Key Codes	Examples and Notes
Picking up Other Calls		
<p><u>Picking up Calls Ringing at Another Extension</u></p> <p>Ringing call pickup allows you to pickup a call that is ringing at a different extension.</p> <p><i>Pickup a connected call</i></p> <p>Connected call pickup allows you to pickup a call that is currently connected to a different extension.</p>	<p>*78 + extension</p> <p>*27 + extension</p>	<p>To pickup a call that is ringing at extension 300, dial: *78300</p> <p>To pickup a call that is currently connected to extension 300, dial: *27300</p>
<u>Using the UC Server Conference Bridge</u>		
<p><i>Conference Server</i></p> <p>Note: The bridge number must be 4 digits long.</p> <p><i>Conference Server - Prompt Caller</i></p> <p>This feature is used for external callers to connect to a conference bridge.</p>	<p>7050 + bridge#</p> <p>7050</p>	<p>To automatically connect to the conference server bridge number 1234, dial: 70501234</p> <p>To connect to the conference server, dial: 7050 You will then be prompted to enter a bridge number.</p>
<u>Parking Calls and Picking up Parked Calls</u>		
<p><i>Park a call using a specific park number</i></p> <p>Allows you to park a call so that you can pickup the call using a different phone. This feature is useful if you need to switch phones while on a call.</p> <p><i>Park a call at a park number selected by UC Server</i></p> <p>This feature is the same as above, except that the call is parked at a number chosen by UC Server instead of you choosing the number to park the call. This is useful if you are uncertain whether another call has already been parked at a particular number.</p> <p><i>Pickup a parked call</i></p> <p>Enables a user to pick up a parked call</p>	<p>*99 + park#</p> <p>*99</p> <p>*98 + park#</p>	<p>To park a currently connected call at park number 1234:</p> <ol style="list-style-type: none"> 1. Put the current call on hold. 2. Establish a new call. 3. Dial *991234 <p>The call is automatically parked at 1234.</p> <p>To park a currently connected call:</p> <ol style="list-style-type: none"> 1. Put the current call on hold. 2. Establish a new call. 3. Dial *99. <p>The system will tell you what number the call is parked at.</p> <p>To pickup a call parked at 1234, dial: *981234</p>

UC Server Call Features

Key Codes

Examples and Notes

[Paging](#)

Contact your system administrator for instructions on sending recorded pages.

Sending a Live Page
You can send a live page.

*72

To send a live page
1. Dial *72 (or *PA)
2. After you hear the tone, begin speaking.
3. When you are finished with the page, hang up or use the end call function on your phone.

3.2 Parking Calls and Picking up Parked Calls

You can park an active call, and then have the parked call picked up on another phone. This feature is useful if you want to change phones during a call.



NOTE: Call park/pickup is a feature that must be enabled by your system administrator for use on your phone.

There are two methods of parking calls. In one method, UC Server selects the next available park number. Alternatively, you can select your own park number. Having UC Server select the park number is useful if you are uncertain whether a call is already parked at a particular number.

If more than one identity (for example, more than one extension number) is assigned to your phone, then transfer the call rather than putting the active call on hold.



NOTE: If a person whose call has been parked then puts the call on hold, the person retrieving the parked call will hear dead air or silence until the other party resumes the call.

To park a call at the next available park number

- 1 Put the current call on hold.



NOTE: If your phone has more than one identity assigned to it, then **Transfer** the call instead of putting it on hold.

- 2 Dial *99.
- 3 Note the **hold position** that plays over the telephone.

The hold position is used by the person who answers the parked call. UC Server selects the park number for the caller starting at 1, or the next available higher number.

4 Press **DIAL**.

The call is removed from the hold position, and put in the system park position.

To park a call at a selected park number**1** Put the current call on hold.

NOTE: If your phone has more than one identity assigned to it, then **Transfer** the call instead of putting it on hold.

2 Dial ***99**.**3** Enter the number that you want to use as the park number.

The hold position is used by the person who answers the parked call. For example, if you enter 1234, then the call is parked at 1234, and can be picked up at 1234.

4 Press **DIAL**.

The call is removed from the hold position, and put in the park position.

To answer a parked call**1** Typically, a company announcement informs you that a call is on hold for you.**2** Dial ***98**.

The system prompts you for the park hold position.

3 Dial the number of the park hold position.

The system disconnects the current call and presents a new call.

4 Answer the new call to speak to the parked caller.

NOTE: If nobody answers the parked call, the call rings the extension that originally put the call on system park.

3.3 Picking up Calls Ringing at Another Extension

Ringling call pick up allows you to answer other ringing telephones, which means that you no longer have to run to answer another person's telephone. You can divert a ringing call from another telephone, typically within earshot, to your own telephone to answer.

To use ringling call pick up, you must know the extension of the phone that is ringling.

This feature also allows for night bell support. Many small companies have an audible bell that rings after business hours. Night bell support can also allow employees to contact security after hours. This feature can allow employees, for example on manufacturing floors, who are not stationed at a particular desk to answer any telephone.



NOTE: To use ringing call pick up, your phone must be configured to receive incoming calls. Make sure that such features as **call forwarding** and **do not disturb** are *not* enabled on the phone.

To answer a call on another user's phone

- 1 When you hear a phone that you want to answer, dial ***78<Extension>**.
Make sure that you dial, on your own phone, *78 plus the extension of the phone that is ringing.
- 2 If the phone has stopped ringing by the time you dial the *78 code, the system prompts you to enter your pickup ID.



NOTE: If the pickup ID that you enter is invalid, the system prompts you with "That is not a valid pickup ID," and asks you to enter a valid pickup ID.

3.4 Using the UC Server Conference Bridge

UC Server has the ability to support a limited-sized conference bridge application. The number of participants is limited to the number of licenses available, however additional conference expansion licenses can be purchased. Use your SIP phone during the conference to access several options, such as muting or adding participants.

To access the conference bridge

- 1 From an internal telephone, dial **7050**.



TIP: You can also dial 7050 to access the conference bridge from the **dial by extension** or **dial by name** directory.

- 2 Enter the 4-digit **conference ID**.
- 3 Follow the prompts to access the conference bridge.



TIP: To access the list of options, you can **press *** while the conference is in progress.

Conference option list

When you press * during a conference, the following options are available. The available options depend on at which point during the conference you press *. Only relevant options are available.

- **Press 1** to mute yourself. Response: “You are now muted.”
- **Press 2** to unmute yourself. Response: “You are now unmuted.”
- **Press 3** to disable entry and exit notifications. Response: “Entry and exit notifications have been disabled.”
- **Press 4** to mute all other participants. Response: “All others have been muted.”



NOTE: When you press 4 to mute all participants, this does not apply to new callers who join the conference. To mute new callers, press 4 again.

-
- **Press 5** to unmute all other participants. Response: “All others have been unmuted.”
 - **Press 7** to toggle between closing and opening the conference to additional participants. Response: “The conference is now closed to additional participants.” Response: “The conference is now reopened to additional participants.”
 - **Press 9** to end the conference.
 - **Press 0** to report the number of participants. Response: “Including yourself, there are x participants.”
 - **Press #** to return to the conference.

3.5 Queuing Calls

Call queuing allows callers to remain in a queue while they wait for their call to be answered. If callers know the queue number, they can dial directly into the queue.

The company agent is typically associated with a particular area of expertise, for example, sales or support. A support agent who is ready to answer calls can log into the support agent queue and receive support calls in order of arrival. Call queuing allows calls to be distributed evenly among the agents that are logged into the queue.

For more information on queuing calls see the following sections:

- [Logging in and out of a call queue](#)
- [Temporarily stop accepting queued calls](#)
- [Checking your agent status in a call queue](#)

Logging in and out of a call queue



NOTE: The call queuing codes, in particular, the code to *Log out of a call queue* (*82) conflicts with one of the Grandstream feature codes, *Send Caller ID*. Therefore your system administrator may have changed the call queuing codes from their default values listed below. For more information see your system administrator.

To log in to a queue as an agent

- 1 Dial *81<QueueNumber>.
- 2 Hang up.

To log out of a queue as an agent

- 1 Dial *82<QueueNumber>.

As previously noted, your system administrator may have changed the code number from the default value of *82. For more information see your system administrator.

- 2 Hang up.

To dial directly into a queue as a caller

- 1 Dial *80<QueueNumber>.
- 2 Hang up.



NOTE: Contact your system administrator to obtain the queue number for your queue.

Temporarily stop accepting queued calls

If you are logged into a queue to accept incoming queued calls, there may be times when you wish to temporarily stop accepting queued calls, without logging out of the queue. You can manually set your status to unavailable to prevent accepting further incoming queued calls, without logging out of the queue.

Temporarily setting your status to unavailable does not change your "longest idle" position in the agent group. When you make yourself available again by logging back into the queue, your idle time is based on the time since your last call, including the time that you were unavailable.

To set your agent status to unavailable

- 1 Dial *83.
- 2 Press a number from 1 to 9 to indicate your reason for being unavailable.

- 3 Hang up.

To set your agent status to available

- 1 Dial *83.
- 2 Press "0" at the prompt.
- 3 Hang up.

Checking your agent status in a call queue

You can check your agent status to verify whether you're currently logged into a call queue, whether your status is set to temporarily unavailable, or whether you're currently logged out of the call queue.

To check your agent status

- 1 Dial *84<QueueNumber>.
- 2 Ensure that you enter the correct queue number.
The system will indicate your current status as:
 - Logged in
 - The agent is not available, service code <#>
 - Not Logged in
- 3 Hang up.

3.6 Paging

You can send a live page, which allows you to instantly broadcast a page. For instructions on sending recorded pages, see your system administrator.



NOTE: In order to send a live page, your system administrator must have enabled access for you to do so. For more information see your system administrator.

Sending a live page

To send a live page

- 1 From your phone, dial *72 (or *PA).
- 2 After you hear the tone, begin speaking.
- 3 When you are finished with the page, hang up or use the end call function on your phone.

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