Goldtel
SOLUTIONS

hosted pbx
CLOUD SERVICES

User Guide

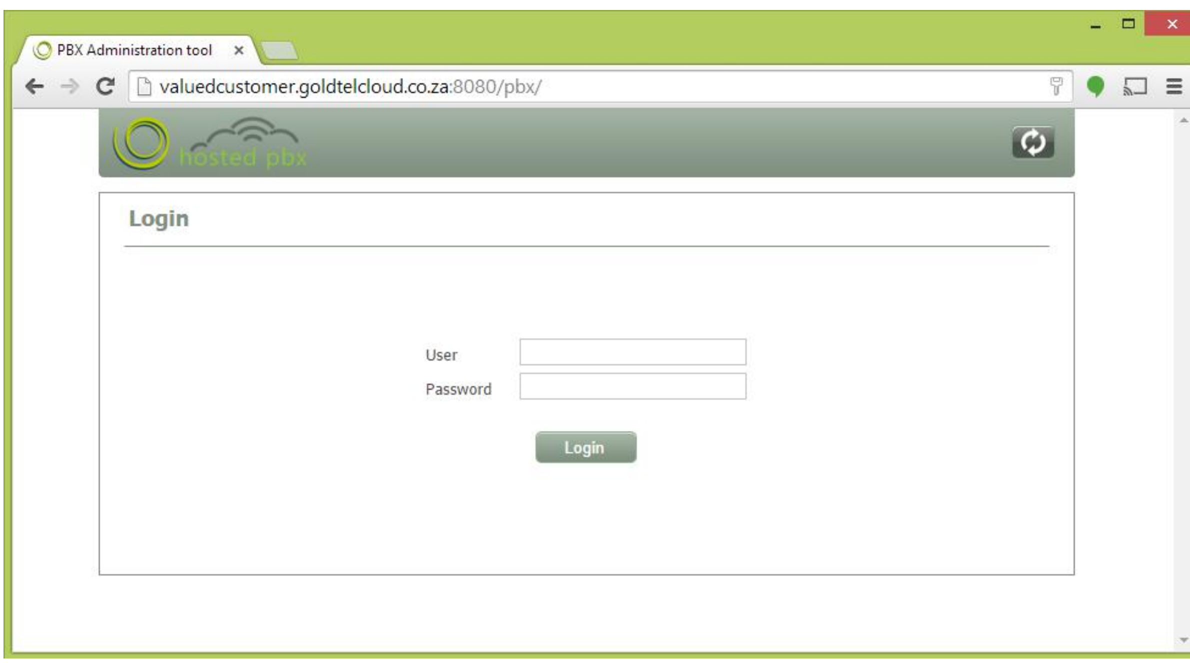
1. Login to Hosted PBX Admin Portal

Use the Hosted PBX Admin Portal to configure user settings:

1) Using your web browser application, access the Hosted PBX Admin Portal user login page. You can obtain the URL from your system administrator or Goldtel Solutions. (The URL will have this format: ***http://valuedcustomer.goldtelcloud.co.za:8080/pbx***.)

2) On the login page, enter your extension username and password, all of which should be provided by the system administrator or Goldtel Solutions. After logging in to your account, you can change your login password and other personal settings on the [Account] page. There is a detailed explanation for each field in the “User Settings” section.

3) After logging in, you will automatically be placed on the page that was open the last time you logged out. You can change the settings as your need. For details about each field, refer to section “User Settings”

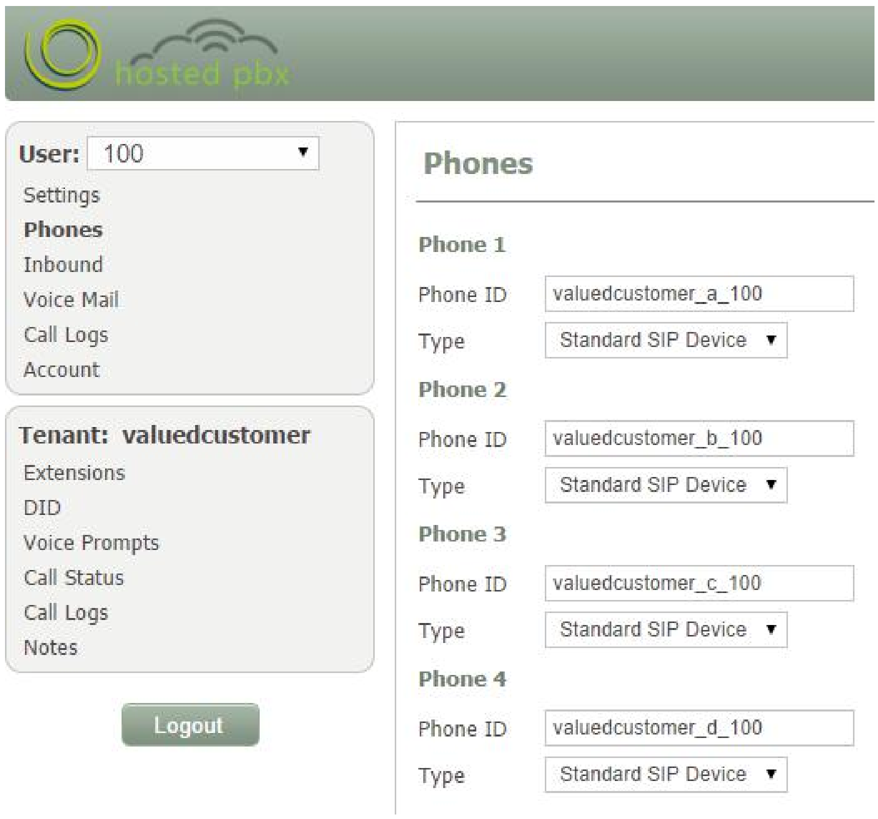


2. Basic PBX Functions

2.1. Making Calls Between Users' Phones

Up to four phones can be assigned under each Hosted PBX user extension on the [Phones] page in the [Phone n] > [Phone ID] field. These phones can call each other by dialing the number 1 to 4 in [Phone n].

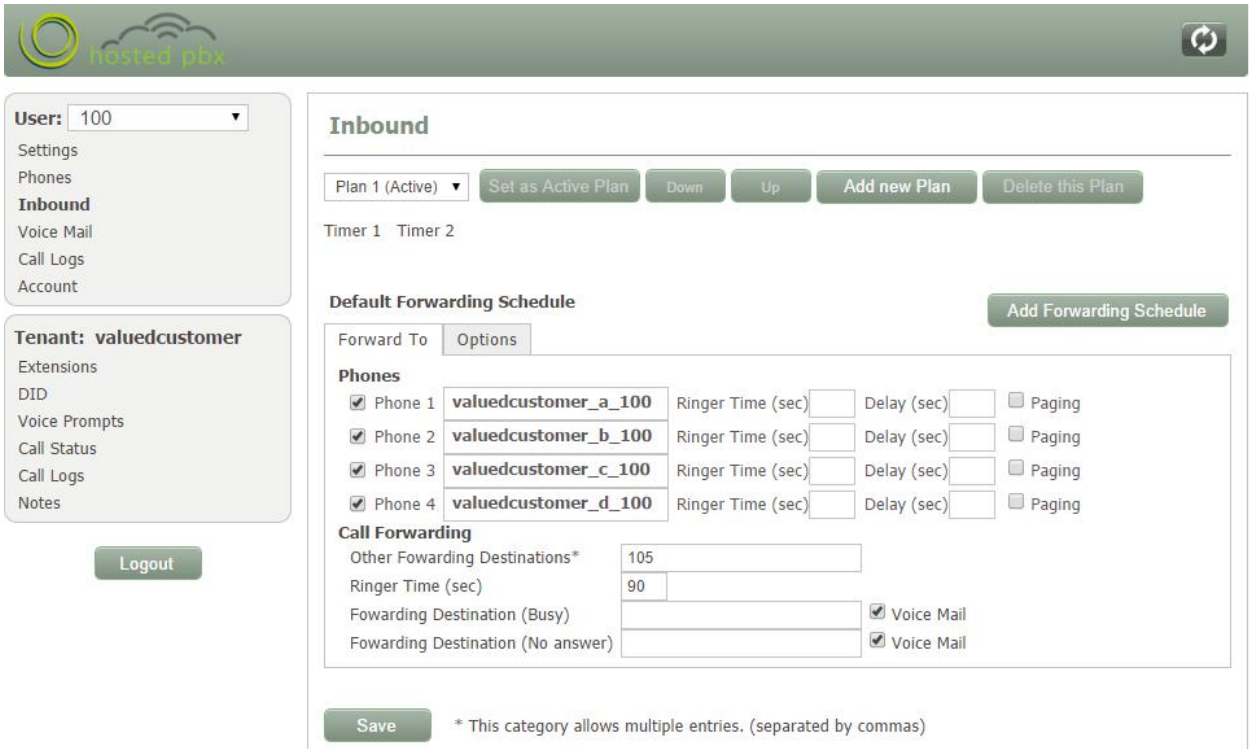
For example, Hosted PBX user 100 has the following assigned phones at the [Phones] page:
User 100 > [Phones]



The phone with SIP ID "valuedcustomer_a_100" can call the phone in the Phone 2 ("valuedcustomer_b_100") field by dialing 2 ABC. The other assigned phones can call "valuedcustomer_a_100" by dialing 1.

2.2. Making Calls Between Extensions

Dial an extension number to make a call between extensions. Calls between different Goldtel Hosted PBX customers will be regarded as external calls. When dialing a user extension, all assigned phones under that extension will ring(default setting). In addition, the clients set in this user's [Other Forwarding Destinations*] field will ring at the same time. When one of these users answers the call, the other phones will stop ringing. If a call is not answered by any client in the time set in [Call Forwarding] > [Ringer Time (sec)], or if all clients are busy, the call will be routed to the number set in the [Call Forwarding] > [Forwarding Destination (Busy/No answer)] field.



2.3. Holding Calls

2.3.1. Placing a Call On Hold

On most SIP phones, pressing the “Hold” button will place a call on hold. Alternatively, dialing #9 will also place a call on hold. When you place a call on hold, you will hear a steady beeping sound and the person on hold will hear music.

2.3.2. Taking a Call Off Hold

To resume a call, follow the instructions for your SIP phone on how to resume calls. Depending on the manufacturer, the steps to resume a call may differ. When you put a call on hold by dialing #9, you can resume the call by pressing *.

2.3.3. Resuming a Call That Is on Hold After a Hang Up

If you hang up your phone while a call is on hold, the Hosted PBX will automatically ring back the phone that placed the call on hold. If you answer the call, you will resume the conversation with the person on hold. If you do not answer, the call will be disconnected when the ringing times out.

In some cases when the “Hold” button is used, this call back notification feature may not be available.

2.4. Attended Transfer

An attended transfer is a transfer made after notifying the receiving party of the transfer. For example, if you have a call that you wish to transfer to a colleague at extension 123, the following steps will walk you through transferring the call using the keypad command (#9).

If you are using a SIP phone that has a Transfer button, please consult your system administrator for instructions on how to use the Attended Transfer feature.

1) Dial #9 to place the call on hold.

2) Dial the intended recipient's number, followed by #. For example, to transfer to extension 123, press 123#. (By pressing *, you can return to the original call.)

3) Let the recipient know that you are transferring a call to him, then hang up your phone. The transfer is complete and the caller is connected to the recipient. (Note that by dialing #9 before hanging up during a transfer to the recipient, the transfer will be canceled and the call will go back to the hold state.)

2.5. Unattended Transfer

2.5.1. Making an Unattended Transfer

An unattended transfer is defined as a transfer made without notifying the receiving party before transferring the call. For example, you can transfer a call to your colleague at extension 123 by going through the following steps.

If you are using a SIP phone that has a Transfer button, please consult your system administrator for instructions on how to use the Unattended Transfer feature.

1) Dial #9 to place the call on hold. (By pressing * in this mode, you can return to the original call.)

2) Dial the intended recipient's number, followed by #. For example, to transfer a call to extension 123, dial 123#.

3) Hang up the phone before the recipient answers the call. The transfer is complete when extension 123 answers the call.

2.5.2. Returning to the Original State Before an Unattended Transfer is Complete

Before the recipient of an unattended transfer answers the call, you can dial ** (Star twice) from the phone you hung up in order to return the call to the state that it was in before being placed on hold.

*For many SIP phones, you will need to press the **Send** button after dialing ** (Star twice).*

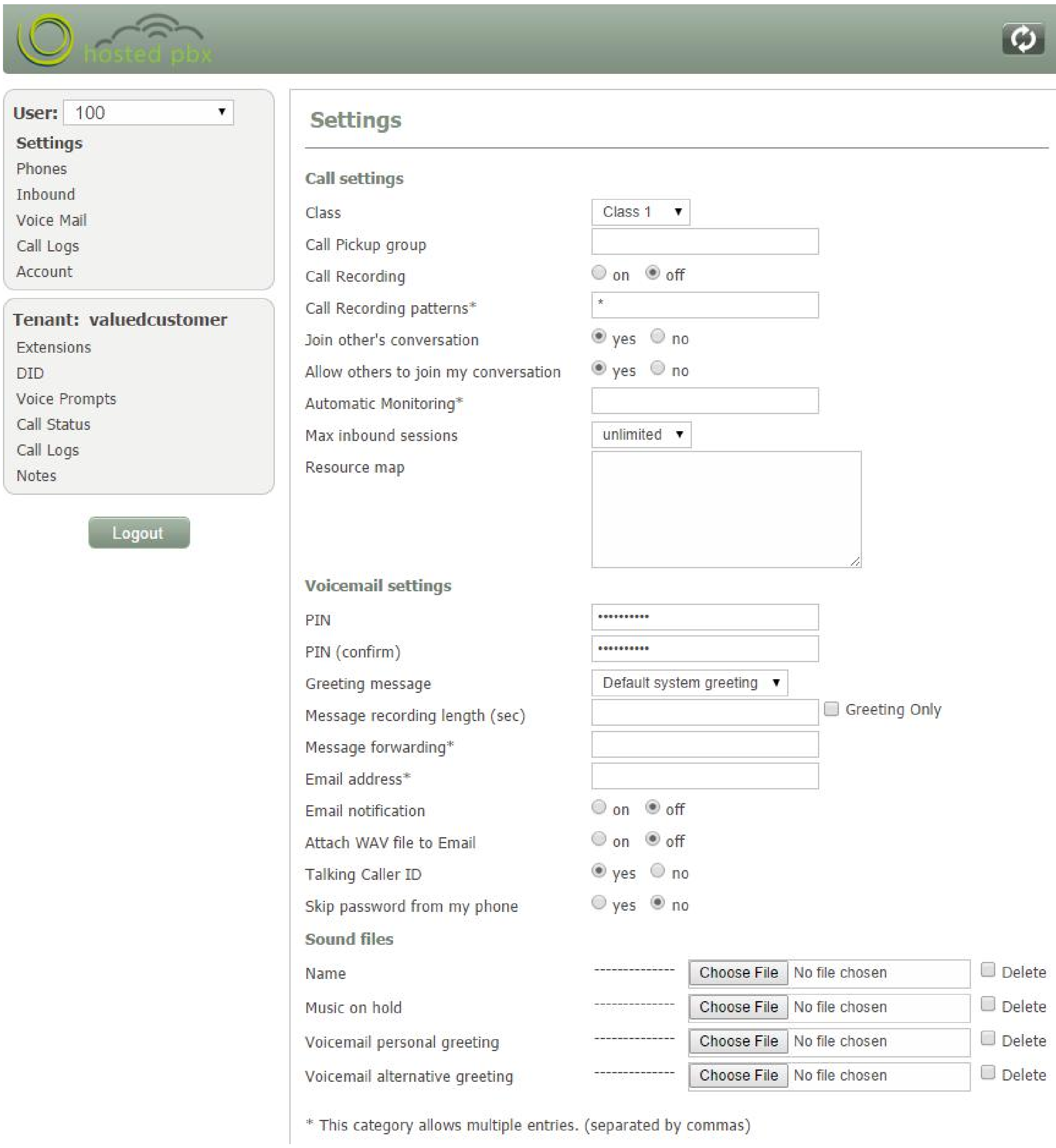
2.6. Voicemail Access and Settings

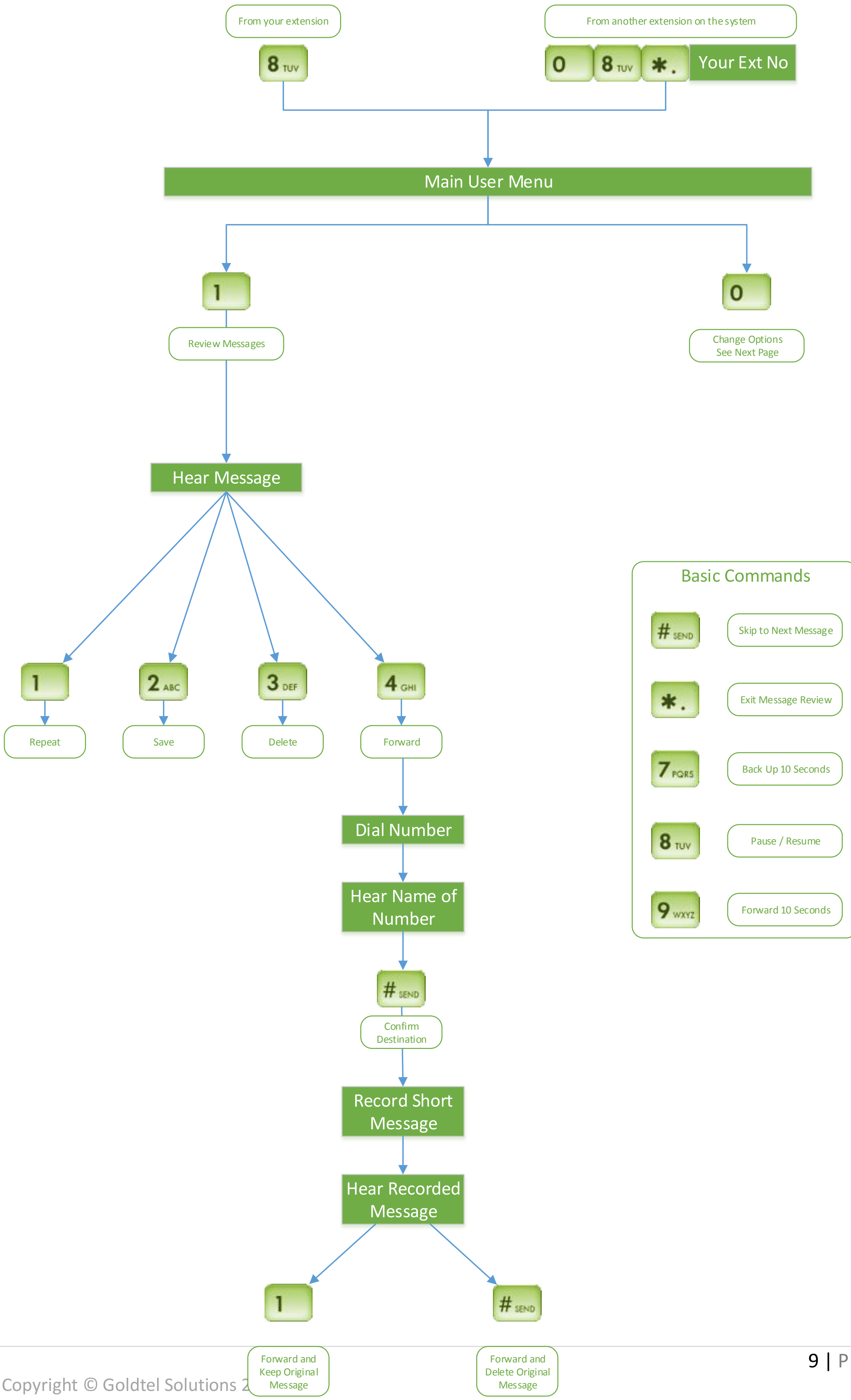
In order to use Hosted PBX voicemail, you will first need to get your extension number and password from your system administrator or Goldtel Solutions. By default, dialing 8 from your extension's phone or dialing 08* followed by the extension number from another phone in your system will allow you to reach the voicemail settings menu. By calling your voicemail access number, you can retrieve messages and change your voicemail settings and personal options.

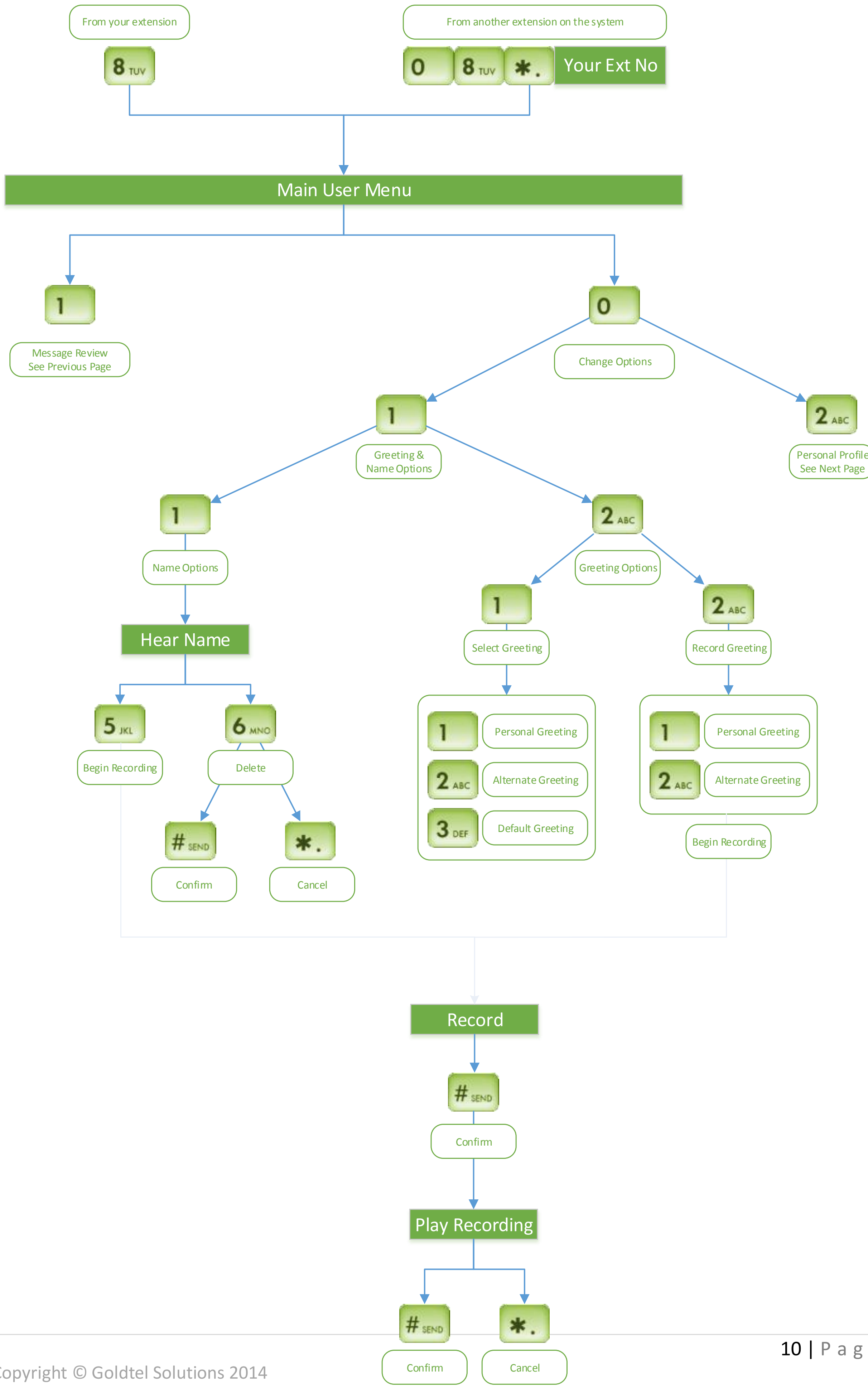
To listen to new and saved voicemail messages, dial your voicemail access number. After listening to a voicemail message, you can choose to Save, Erase or Forward the message to other extensions.

You can also rewind and forward a message while listening to it. Use the “Voicemail Navigation Map” document to see how to perform any of these tasks.

From the user account [Settings] page, you can change your voicemail PIN and other voicemail settings.




From your extension
8 TUV
From another extension on the system
0
8 TUV
*
Your Ext No
Main User Menu
1
Review Messages
0
Change Options
See Next Page
Hear Message
1
Repeat
2 ABC
Save
3 DEF
Delete
4 GHI
Forward
Dial Number
Hear Name of Number
SEND
Confirm Destination
Record Short Message
Hear Recorded Message
1
Forward and Keep Original Message
SEND
Forward and Delete Original Message
Basic Commands
SEND
Skip to Next Message
*
Exit Message Review
7 PQRS
Back Up 10 Seconds
8 TUV
Pause / Resume
9 WXYZ
Forward 10 Seconds


From your extension
8 TUV
From another extension on the system
0
8 TUV
*.
Your Ext No
Main User Menu
1
Message Review
See Previous Page
0
Change Options
1
Greeting & Name Options
2 ABC
Personal Profile
See Next Page
1
Name Options
Hear Name
5 JKL
Begin Recording
6 MNO
Delete
SEND
Confirm
*.
Cancel
2 ABC
Greeting Options
1
Select Greeting
1
Personal Greeting
2 ABC
Alternate Greeting
3 DEF
Default Greeting
2 ABC
Record Greeting
1
Personal Greeting
2 ABC
Alternate Greeting
Begin Recording
Record
SEND
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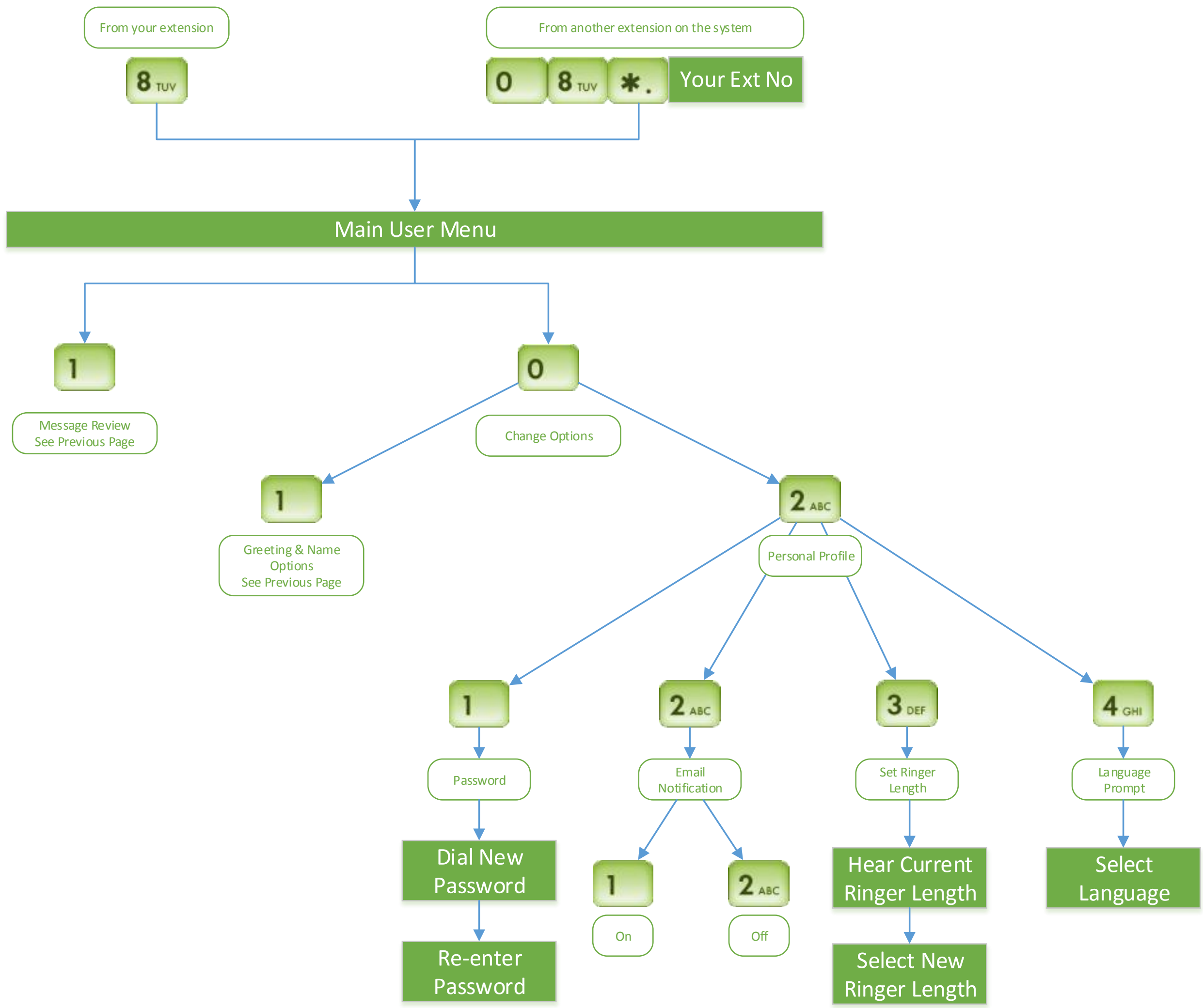
Play Recording
SEND
*.
Confirm
Cancel



2.7. Call Pickup

The Call Pickup feature allows you to answer incoming calls that are directed to other user extensions. Configuration of this feature is done through the [Users] menu in Hosted PBX Admin Portal. Refer to the "User Settings" section in this document to learn how to configure this feature.

*If you are using a SIP phone that has a **Call Pickup** button, please consult your system administrator for instructions on how to use the Call Pickup feature.*

2.7.1. Call Pickup from Within a Call Pickup Group

Call pickup from within a call pickup group allows you to pick up any incoming call directed to a [Groups] extension.

1) Set a [Groups] extension number at the user's [Settings] page in the [Call settings] -> [Call Pickup group] field.

2) When any user extension in the [Groups] extension rings, press 

to answer the incoming call.

2.7.2. Call Pickup from a Non-Group Extension

Call pickup from a non-group extension allows you to pick up any incoming calls directed to any user extension within the system.

1) From any extension, dial *. <user extension number> to answer an incoming call.

2) For example, if user extension 511 rings, dial *. 5 JKL 1 1 from any other Hosted PBX user extension in your system to pick up the call.

2.8. Call Parking

Call Parking allows you to place a call on hold with a number that you assign, which anyone can then pick up from a different extension.

If you are using a SIP phone that has a Call Park button, please consult your system administrator for instructions on how to use the Call Park feature.

2.8.1. Parking a Call

1) Dial # SEND 8 TUV to put the call on hold.

2) Input a park number followed by # SEND. The system will confirm the park number by stating it out loud.

3) Now the call is parked and ready to be resumed from any extension by dialing the park number that you just input.

2.8.2. Pick Up a Parked Call

There are several ways to pick up a parked call:

♦ By dialing the park number that was input when the call was put on park.

♦ By dialing *. *. from the extension where the call was parked.

♦ By dialing 8 TUV *. **<extension number>** from any user extension.

♦ By dialing 8 TUV *. **<group number>** from an extension in the ring group where the call was parked.

♦ When a [Call pickup group] is set for the extension that will pick up the parked call, dialing 8 TUV *. allows that extension to pick up the parked call from any extension in the call pickup group.

♦ If the phone supports the Shared Call Appearance (SCA) feature, the call can also be parked by pressing the hold button on the SIP Phone, and can then be picked up by pressing the shared line keys from another phone that supports the SCA feature.

If you are using a SIP phone that has a Call Park button, please consult your system administrator for instructions on how to use the Call Park feature.

2.8.3. Resuming a Parked Call

There are several ways to resume a parked call:

♦ If you put the call on hold by dialing # SEND 8 TUV, then pressing *. will return you to the original call.

♦ If you forget the park number, you can retrieve the parked call (return to the state the call was in before you parked it) by dialing *. *. from the extension you used to park the call.

♦ You can retrieve a parked call by dialing 8* **<extension>** or 8* **<group number>** of the user extension where you parked the call.

♦ If the extension or the group number is set in the [Call Pickup group] field in your [User settings], you can pick up the parked call by dialing 8*.

*For many SIP phones, you may need to press the **Send** button after dialing* **.

2.9. Conference Calls

2.9.1. Entering a Conference Room

Ask your system administrator for a conference room number. If your system administrator has set extension 1000 as the conference number, you can join the call by dialing 1000.

2.9.2. Holding a Conference Meeting Instantly

There are two ways to hold a conference meeting instantly:

♦ If your administrator set the conference attendees' numbers in the [Forwarding destinations*] of the conference number (1000), then any attendee on that list can join the conference instantly by dialing 1000.

♦ Dialing the conference number (1000) followed by the attendees' numbers separated by stars (*) will invite those extensions to join the call.

Example:

1000*101*102*103.

A star * *or a caller's extension should be displayed on the conference room extension edit page in the [Applied to (Caller numbers)*] field. When a star is entered in this field, there is no restriction on which extension can convene a conference meeting.*

2.9.3. Joining a Conversation

To join a conference meeting or a three-way call, dial 0* **<extension number>**. For example, if users 101 and 102 are talking, then dialing either 0*101 or 0*102 from another extension will allow you to join their conversation.

2.9.4. Inviting Others to an Existing Meeting

To invite others to join an existing conference meeting:

1) Dial #9 to place the call on hold and then dial the new member's number. You will then be talking with the new member privately.

2) Dial #0 to invite that person to your meeting. If you dial #7 instead of #0, the new member will only be able to listen to the conversation.

If you are using a SIP phone that has a Conference button, please consult your system administrator for instructions on how to use the Conference feature.

[Join other’s conversation] and [Allow others to join my conversation] fields in user extension [Settings] page can be used to set restriction to conference calls or call supervising.

Refer to “Quick Reference Chart” for the commands which can be used during video conference call.

2.10. Call Supervising

If you would like to supervise a conversation, use one of the following options:

♦ Dial 9 * **<extension number>.**

♦ To monitor a specific user extension automatically, enter the supervisor’s extension number in the monitored user’s [Settings] page [Automatic Monitoring*] field.

Join other’s conversation] and [Allow others to join my conversation] fields in user extension [Settings] page can be used to set restriction to conference calls or call supervising.

2.10.1. Supervising Call Example

This example shows what happens to each party during a call when a supervisor joins the call:

1) A and B are talking.

2) C joins in the conversation as a supervisor by pressing 9 * **<A's user extension number>**.

3) A places the call on hold or initiates a transfer.

4) B hears music on hold while C hears the same as A. C has the same controls as A, such as transferring calls, recording calls, etc.

When A terminates the call, C's session will also be terminated under supervising mode. It may be easier to think of a supervisor as an alias of A.

2.10.2. Automatic Monitoring

This example shows what happens to each party during a call when a supervisor automatically joins the call to monitor it by using the [Automatic Monitoring] feature:

1) Specify C’s extension number in extension A’s [Automatic Monitoring*] field.

2) B calls A (or A calls B).

3) C’s phone will ring.

4) C can speak and listen to the conversation between A and B. C has the same controls as A, such as transferring calls, recording calls, etc.

2.10.3. Tutor Mode

Tutor mode is a special version of Call Supervision. If a supervisor joins a conversation in Tutor Mode, his/her voice can be heard only by the extension under supervision. However, the supervisor can hear all parties to the conversation.

There are two ways to use tutor mode:

♦ Dial 9 9 * **<user’s extension number>**.

To tutor a specific user automatically, in the [Automatic Monitoring*] field of the user extension, enter either two tildes with the supervisor’s extension number, such as “**~~<supervisor’s extension>**” or **<supervisor’s extension> {tutor}**, such as **100{tutor}**.

2.11. Call Recording

All options for Call Recording can be configured on a user-by-user basis in Hosted PBX Admin Portal.

Your system administrator can set the default Call Recording state in [Call Recording]: on/off at the user's setting menu in Hosted PBX Admin Portal. The total number of recorded messages and voicemail messages cannot exceed the number set in the [Max stored messages] field in the [Options] menu. Recorded messages can be stored in the same location as the user extension's voicemail messages or in user call logs.

2.11.1. Recording While a Call Is in Progress

To start recording while a call is in progress, dial #6. To stop recording, dial #6 again. You will hear a beep sound when call recording starts and a buzz sound when it stops. The buzz sound will not be heard by the other party.

2.11.2. Recording Monitored (Supervised or Conference) Calls

While you are monitoring (supervising) someone's conversation, or participating in a conference call, you can start recording by dialing #6, but only you and the person you are supervising will hear the buzz sound.

2.11.3. Recording Option When Initiating a Call

If you want to specify the Call Recording option when initiating a call, dial 6***<user extension number>** for Record=on and dial 60***<user extension number>** for Record=off. When answering a parked call, supervising a call or joining a conference, dial the recording numbers after 8*, 9*, 0*, * or **.

2.11.4. Call Recording Patterns

With the settings in [Call Recording patterns*], the Hosted PBX users can restrict call recording only to certain extensions. For example, if you enter **1*** in this field, only calls to or from the numbers with prefix 1 will be recorded.

2.12. Mute

You can use the Mute function during a conference call or a one-on-one call. With the Hosted PBX Mute function you can set who will be muted in the conversation.

2.12.1. Mute Callee This feature mutes incoming callers to a conference call, making it a "listen only" call.

- If you dial #7 to let someone join in a conference call, the new attendee can only listen to the conversation in the conference.
- When **~(tilde)<extension number>** is set in [Inbound] > [Other Forwarding destinations*] under User Settings, that extension user's call will be in "Mute" mode. By setting **~(tilde)<extension number>** for all of the members in a conference, you can create a one-way broadcast conference.

2.12.2. Mute Caller

This feature is used to mute your own phone during a conversation.

- By dialing 7***<extension number>** or (a 8*, 9*, 0*, * or **) + 7***<extension number>**, you can mute yourself in the conversation.
- Dialing 9*7***<extension number>** will let you monitor (supervise) someone's call silently.

2.13. Broadcasting

With the Hosted PBX broadcasting function, you can set a conference call to be sent to multiple extensions in listen-only “Broadcast” mode. When the call is ended by the host, the attendees’ call will be disconnected as well.

- ♦ Set **~(tilde)<broadcast number>** in the conference extension’s [Forwarding destinations*] field and select “**yes**” in the [Exit all when host leaves] field, then when you dial in to this conference number, your call will be in “Broadcast” mode.
- ♦ Set **<broadcast numbers>** in the conference extension’s [Forwarding destinations*] field and select “**yes**” in the [Broadcast] and [Exit all when host leaves] fields,

2.14. Confirm Call

Confirm Call is a function that ensures an outbound call is answered by a person, and not by a voicemail system. The callee will be prompted to press a preset confirmation key (set by system administrator) to confirm that the call has been answered by a person.

2.14.1. Pick Up a Confirm Call

This example shows what happens when a PSTN phone gets a confirm call from Hosted PBX:

1) Set up Confirm Call using the ARS rule for outbound calls.
2) Dial to a PSTN phone number through the ARS rule from a user extension.
3) When the PSTN phone receives the incoming Confirm Call, there are three possible scenarios for how to handle Confirm Calls:
    - Answer the call and enter the preset confirmation key when the voice prompt plays. If the confirmation key is accepted, the call will be connected.
    - If there is no confirmation key entered, the call will be disconnected after the voice prompt.
    - If the call is not answered, it will be disconnected after the ringer times out. The caller will not be connected to the PSTN phone’s voicemail.

2.15. Paging

Paging1 is also called Auto Answer. The Hosted PBX can invoke a SIP phone’s paging function. The phones will answer incoming calls automatically without taking the handset off-hook when they receive the paging information sent in the SIP header from the Hosted PBX.

Use one of the following options to make a call with paging:

- ♦ Dial 2 ABC 2 ABC *. **<user's extension>**.
- ♦ To page a specific user’s extension automatically, enter **<user’s extension>{p}** or **<user’s extension>{page}** on that extension’s [Inbound] page.

For other parameters used with paging, refer to the “Parameters and Syntax” section.

2.15.1. Automatic Paging

1) Set user extension B’s [Other Forwarding destinations*] as **B{p}** or **B{page}**.
2) A calls B.
3) The phone(s) assigned to this user that support paging will be connected without ringing.

2.16. Busy Lamp Field

With Busy Lamp Field (BLF2), when there is a call to the monitored phone, the corresponding key lamp on the monitoring phone will flash and the call can be picked up from the monitoring phone.

2.17. Presence

The Hosted PBX can handle presence[2] SUBSCRIBE requests from phones, and return NOTIFY responses about the status of the monitored phones, such as "available" or "on the phone."

[1] A list of SIP phones that work with the Hosted PBX paging function and a sample configuration are available from Goldtel Solutions.

[2] A list of SIP phones that work with the Hosted PBX BLF and with the Presence and SCA function, as well as a sample configuration are from Goldtel Solutions.

2.18. Shared Call Appearance

With Shared Call Appearance (SCA[2]), Hosted PBX users can monitor statuses of ARS route lines, select an available line to place an outbound call or answer an incoming call.

3. User Settings

The following sections describe how to modify your individual settings for optimal performance and convenience. After making any changes, click [Save] to save them.

3.1. Settings

3.1.1. Call Settings

Name	Default value	Description
Class	Class 1	User class type
Call pickup group		Enable one-touch call pickup for the preset group extensions by assigning the group number.
Call recording	off	Enable or disable call recording. Options: on/off
Call recording patterns*	*	Wildcard setting used to filter incoming/outgoing calls that are needed to enable call recording. A star (*) and a question mark (?) can be used for matching meta-characters. For multiple entries, use a comma to separate each number.
Join other's conversation	yes	Allow (yes) or forbid (no) this user to join another user's conversation. Options: yes/no
Allow others to join my conversation	yes	Allow (yes) or forbid (no) other users to join this user's conversation. Options: yes/no
Automatic Monitoring*		Allow other users to "monitor" this user's conversation. For multiple entries, use a comma to separate each number.
Max Inbound Sessions	unlimited	Maximum concurrent sessions Options: unlimited, 0-6
Resource map		Map clients' custom parameters to the Hosted PBX parameters.

** This setting section is disabled from user access by default.*

3.1.2. Voicemail Settings

Name	Default value	Description
PIN		PIN to retrieve voicemail messages.
PIN (confirm)		Input field to confirm your PIN.
Greeting message	Default system greeting	Specifies the greeting message that will be used with your voicemail inbox. Options: Default system greeting Personal greeting (customizable) Alternative greeting (customizable)
Message recording length(sec) Greeting Only		Length of time for recording a voice message, in seconds. *When this box is checked, only the greeting voice prompt is played and the caller cannot leave a voicemail message.* *Options: check/uncheck*
Message forwarding*		Hosted PBX user number(s) to which you would like to forward your voicemail messages.

		If you use this setting, all of your messages will be automatically forwarded to the number(s) specified here. For multiple entries, use a comma to separate each number.
Email address*		Specifies an email address for email notification. For multiple entries, use a comma to separate each email address.
Email notification	Off	Enable (on) or disable (off) email notification, which sends an email to a specified address when a new voicemail is received.
Attach WAV file to email	Off	Enable (on) or disable (off) attachment of voicemail messages in WAV format to email notifications.
Talking caller ID	Yes	Enable (yes) or disable (no) display the number of the caller who left a voicemail message.
Skip password from my phone	No	Enable (yes) or disable (no) requesting voicemail password when voicemail box is accessed from numbers that are on the user's > [Phones] page.

3.1.3. Sound files

Name	Description
Name	The name associated with the voicemail inbox. (When you leave a message for another Hosted PBX user, the recipient will hear, "You have a new message from xxxx.")
Music on hold	An audio file containing music for callers to listen to while on hold.
Voicemail personal greeting	Voicemail inbox greeting message created by user.
Voicemail alternative greeting	Alternate voicemail inbox greeting message created by user.
Download	To download a recorded sound file, click on the file size. The sound file will be downloaded to your PC as a WAV file.
Delete	To delete a recorded sound file, select the [Delete] check box(es) at the end of the line. The selected file(s) will be deleted when you click the [Save] button on the user setting page.
Upload	To upload a file, click the [Browse] button at the line to which the file will be saved. Select the file you want to upload then click [Save] and the upload will start.

3.1.4. Format of Sound Files Uploaded sound files must be formatted as shown below:

Format	WAV
Sample rate	8000Hz
Bit-depth	16-bit
Channels	Mono

3.2. Phones On the [Phones] page, up to four phone numbers can be assigned to each Hosted PBX user.

Name	Default value	Description
Phone ID	<tenantID_phoneID_extensionID>	Phone SIP IDs that the phones have registered at the Hosted PBX SIP Server. *Can be obtained from your system administrator or Goldtel Solutions.*
Type*	Standard SIP Device	Default phone type or phone type defined by system administrator.

* Choose the proper phone type for each assigned phone for codecs, audio/video, or RTP relay features.

3.3. Inbound

3.3.1. Plan

Name	Description
Plan selection list	Select a plan from a list of created plans. Default: Plan 1 (Active)
Set active plan	Select a plan and set as active to apply it to incoming calls. The active plan can also be changed by dialing the [Switch Plan] extension from the user's phone.
Down	Switch position with the plan right below the current one.
Up	Switch position with the plan right above the current one.
Add new plan	Add a new plan.
Delete this plan	Delete the currently selected plan.
Add forwarding schedule	Add a schedule setting to the current plan.
Save	Save the current plan settings.

3.3.2. Plan [n] > Default Forwarding Schedule > Forward To Phones

Name	Default value	Description
Check boxes	All are selected	The selected phones will ring when there is an incoming call.
Phone [1-4]	<tenantID_phoneID_extensionID>	The same phone ID set in the user [Phones] page. When the box is checked, incoming calls will be forwarded to the selected numbers. When the box is unchecked, the phone will not receive any incoming calls. Options: check/uncheck
Ringer time(sec)		Length of time your phone rings when receiving a call. When the specified time is reached, incoming calls will be forwarded to the extension set in [Call Forwarding] > [Call forwarding (busy/no answer)]. *If there is no setting for a phone's ringer time field, the setting in [Call Forwarding] > [Ringer Time (sec)] will be applied.*
Delay(sec)		Delay time before ringing the selected phone when receiving a call.
Paging		Check paging checkbox related to the phone which supports intercom/auto answer.

Call Forwarding

Name	Default value	Description
Other forwarding destinations*		Numbers to forward incoming calls to, in addition to the phones selected in the [Phones] section. Incoming calls will ring any number(s) set in this field, as well as number(s) set in the [Phones] section. For multiple entries, use a comma to separate each number.
Ringer time(sec)		Length of ringing time applied to destinations in [Other Forwarding destinations*] field or user-assigned phones. When the specified time is reached, incoming calls will be forwarded to the destination set in [Call forwarding (busy/no answer)]. This setting will take effect when: • No ringing time is set in [Phones] section. • There is a setting in [Other Forwarding destinations*].
Forwarding destination (no answer) *Voicemail box*		The destination to which unanswered calls will be forwarded. *If this box is checked, the call will be automatically forwarded to the user's voicemail box.* Options: check/uncheck
Forwarding destination (busy) *Voicemail box*		The destination to which busy response calls will be forwarded. *If this box is checked, the call will be automatically forwarded to the user's voicemail box.* Options: check/uncheck

3.3.3. Plan [n] > Default Forwarding Schedule > Options

Call Waiting / Knock Knock

Name	Default value	Description
Beep on incoming call	No	Enable (Yes) or disable (No) playing notification sound when there is an incoming call while the user is in conversation. Options: Yes/No
Knock Knock *Only for internal extension*	0 Checked	The length of time to play the call waiting sound prompt to the caller. *If checked, this setting will only be applied to incoming calls from other Goldtel Hosted PBX users. Otherwise, this setting will be applied to all incoming calls.* Options: check/uncheck

Follow Me

Name	Default value	Description
Call next phone if phone stops ringing	No	If set to yes, the next available assigned phone will start to ring as soon as the previous phone stops ringing, ignoring the [Delay] setting.

3.3.4. Plan [n] > Forwarding Schedule [n] > Conditions

Caller

Name	Default value	Description
Filter	Matched	Wildcard setting used to filter incoming calls. A star (*) means zero (0) or more characters and a question mark (?) means one character. By adding a * (wildcard) after a number, you can specify all numbers that begin with that number. When the field is left blank, all numbers will be applied to the schedule. Options: Matched/Not Matched
Route	From any route	Apply the schedule to all incoming calls, or to calls from external routes (or not). Options: From any route From external line Not from external line

Date/Time

Name	Default value	Description
Term		Schedule start and end year and date
Days		Days of week, date(s) included and excluded.
Time		Time duration.

3.3.5. Forwarding Schedule [n] > Buttons

Name	Description
Add Forwarding Schedule	Create a forwarding schedule.
Copy	Make a copy of the current schedule.
Delete	Delete the current schedule.
Up	Switch position with the schedule right above the current one.
Down	Switch position with the schedule right below the current one.

3.3.6. Timer 1 / Timer 2

Name	Default value	Description
Term		Schedule start and end year and date.
Days		Days of week, date(s) included and excluded.
Time		Start time to set a plan to be active, which applies this plan's setting to incoming calls.

3.4. Voicemail

Name	Description
Messages	At the upper-right corner, the number of new and saved messages is shown.
Delete	To delete recorded messages, select the check box of each message you want to delete, or check the top box to select all recorded messages and then click the [Delete] button to delete the selected messages.
Date and time of call	Day of the week, date and time when the message was left.
Status	Status of the message (New/Saved).
Other party	Caller information.
Type	The type of message (Voicemail/Call Recording).
Size (bytes)	Size of each message.
Play/Download	Play a message with a media player or download a message to your PC as a WAV file by clicking on the arrow.

3.5. Call Status

Name	Description
Total	Total number of active sessions in the system.
ID	Call ID.
Status	Call status: In progress, Talking.
UAs	The users' phone numbers in the current session.

3.6. Call Logs

Name	Description
Date	Select to show calls on a specific day.
Start time	Select to show calls that start at a certain time and later.
Max records	Max records shown on one screen.
View	The users' phone numbers in the current session.
CSV	Export selected call record to a file in CSV format.

Call Log Record

Name	Description
ARS	ARS rule name applied to the call.
Type	Call type.
URI(UA)	SIP URI of the user phone related to the call.
URI(PBX)	SIP URI of the other party related to the call.
Connected	Time the call is answered.
Call duration	Length of the conversation.

3.7. Notes

A list of notes can be read or accessed at the user level.

Name	Description
Name	Note name.
Description	Description of the note.
User access level	The access level of the note (Read-only or Read/Write).

3.8. Account

Name	Default value	Description
Name		User display name. *This field is optional.*
Descriptions		Description of the user. *This field is optional.*
Language		The selected language will be used for the user's administrative tools and the voicemail prompt. Options: English/Japanese
Login password		Password for logging into Hosted PBX user's administrative tool.
Login password (confirm)		Input field to confirm your password.

4. Quick Reference Chart

4.1. Specifying the Type of Call When Making a Call

4.1.1. Dial Prefix

Prefix	Description
8*	Retrieve a parked call
9*	Supervising mode
99*	Tutor mode
22*	Paging
0*	Join in the conversation of a conference
*	Call pickup
**	Return to a call on hold after hanging up the phone

The Dial Prefix must come first in the dialing number. You can't use multiple Dial Prefixes at a time.

4.1.2. Call Attributes

Attribute	Description
6*	Recording = on
60*	Recording = off
7*	Mute mode

*You can use multiple Call Attributes at a time. Example: 9*6*7*<destination number>*
Example: 60<destination number >*

4.2. Commands to Use During a Conversation

Prefix	Description
#6	Start/Stop recording
#9	Place a call on hold
*	Take a call off hold
#8	Park a call
#0	Let someone join the conference
#7	Let someone join the conference in mute mode

4.3. Commands to Use During a Video Conference Conversation

Prefix	Description
#5	Start video control mode
1	Other attendees will see what you are seeing
2	Other attendees will see you
4	Go to next view
6	Go back to previous view
*	Exit video control mode

4.4. Parameters and Syntax

4.4.1. Parameters

Letter Parameter	Description
page / p	Paging function
mic-off / m	Turn microphone off; user cannot talk
speaker-off / s	Turn speaker off; user cannot hear
tutor / t	Tutor mode

Symbol Parameter	Description
~	User cannot talk; listen only
^	User cannot hear; speak only
~~	Tutor mode

4.4.2. Syntax

- ♦ Letter parameters must be enclosed in curly brackets, such as {and}.
- ♦ When multiple letter parameters are in the same curly brackets, separate with a semi-colon (;). Example: {p;m} means to turn on paging and turn off microphone.
- ♦ Specify <extension number> before letter parameters. Example: 400{p; s} means turn on paging and turn off phone's speaker for extension number 400.
- ♦ The symbol parameters ~, ^ and ~~ are set before <extension number>. Example: ~<extension number>