

# **User Manual**

## **Real PBX Service Version 4.0**

**Real Telecommunication Corporation** 

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# About this Manual

This manual guides users to using some of the **Real PBX Service** features. Below is a summary of the features:

	Standard PBX Features	
<ul> <li>Send and receive calls</li> </ul>	<ul> <li>Caller ID</li> </ul>	<ul> <li>Message waiting</li> </ul>
<ul> <li>Extension dialing (2, 3, or</li> </ul>	<ul> <li>Call Transfer – blind</li> </ul>	indicator
4 digit)	and consultative	<ul> <li>Call detail logging</li> </ul>
Local and long-distance	<ul> <li>Three-way calling</li> </ul>	<ul> <li>Auto attendants and</li> </ul>
calls	(conference)	hunt groups
<ul> <li>Call Forwarding all, no</li> </ul>	<ul> <li>Block caller-ID</li> </ul>	<ul> <li>Voicemail</li> </ul>
answer, busy	<ul> <li>Mute</li> </ul>	<ul> <li>Call Waiting</li> </ul>
<ul> <li>Call Hold</li> </ul>	Do Not Disturb	
Last number redial & call	(DND)	
return		
	Advanced PBX Features	
<ul> <li>Call Park and Call Park</li> </ul>	<ul> <li>Click to dial inbound</li> </ul>	Automatic Call Distributor
Retrieve	call pop ups	or Call Agent Groups
<ul> <li>Call pickup</li> </ul>	<ul> <li>Send/receive faxes</li> </ul>	<ul> <li>Call recording of all calls</li> </ul>
<ul> <li>Direct Inward Dial</li> </ul>	<ul> <li>Call barge, call listen,</li> </ul>	<ul> <li>Integrated Voice</li> </ul>
<ul> <li>Follow Me Find Me</li> </ul>	call teach mode	Response call processing
<ul> <li>Intercom and paging</li> </ul>	<ul> <li>Ad hoc call recording</li> </ul>	<ul> <li>Cell Phone integration</li> </ul>
<ul> <li>Block Anonymous Calls</li> </ul>	<ul> <li>Cell phone support</li> </ul>	prepay support
Anonymous Call Intercept	<ul> <li>Personal address book</li> </ul>	<ul> <li>Call Reporting and</li> </ul>
<ul> <li>Call screening</li> </ul>	<ul> <li>Email voice messages</li> </ul>	Billing
Call Hold – music on hold	aswav files	<ul> <li>Company or system wide</li> </ul>
<ul> <li>Shared Line appearances</li> </ul>	<ul> <li>Email on missed calls</li> </ul>	address book
<ul> <li>White/Black Lists</li> </ul>	<ul> <li>Call cell phone on new</li> </ul>	Secure communications
<ul> <li>Remote Phone Support</li> </ul>	voice mails	with encryption
<ul> <li>Wake up calls</li> </ul>	<ul> <li>Time of Day routing</li> </ul>	<ul> <li>Daily email with system</li> </ul>
<ul> <li>Hot Desking</li> </ul>	with service flags	call logs
<ul> <li>Calling Card – Direct</li> </ul>	<ul> <li>Conference Server</li> </ul>	
inward station access		



We are constantly working on new features and enhancements to make the service easier to use.

Some of the new features are:

- Make your cell and office phones ring for all incoming calls.
- Request emails for all missed calls.
- Request that voicemails go to your email account.
- $\circ\;$  Receive calls while traveling as though you were still in the office.
- $\circ\;$  Retrieve you email from anywhere via your Cell Phone or other phone.
- $\circ\,$  Make calls through your extension from outside your office.
- Multiple phones on the same extension.
- New paging system that is more effective with large number of extensions.
- Improved security feature that better protect against unauthorized attempts to access the system.
- Improved web access for users

The administrator of the system must enable and configure each extension to be able to use the features.

### Introduction

Real PBX Service is designed to provide customers the highest possible flexibility in *moving and managing voice communication*.

Real PBX Service is high quality features rich. It enables customers to make and receive calls from anywhere and enjoy features that are unique to its architecture.

Users can move and manage voice communication with ease using web access, function keys (\* keys) to enable and disable various PBX activity, and integrate email and cell phones to their extensions.



The Real PBX Service supports using any IP phone, ATA, or SIP Phone. In fact, you may use a Softphone on your computer to do all your communication thus eliminating the need for hard phone on your desk.

## What are Star Codes?

Star codes are two-digit numbers prefixed with a star (\*) and are used to tell the system what to do. Each star code maps directly to an action of the system, allowing you to access PBX features more easily. By using star codes, you can activate a wide range of PBX functionality, such as activate and deactivate Do Not Disturb (DND), forward calls to your cell phone, and use Teach Mode to help a new employee make his first sale. Figure 2-1 lists the full set of features codes.

*Note*: The star codes in this chapter are the default codes. If your system administrator has assigned different codes for any of the available features, you will need to request a modified list.

## Making Calls

The IP Phone devices vary in functionality and features. Some require a confirmation to process the outbound call by pressing a confirmation button such as check mark. Please remember your phone must be Registered (connected to the system) before you call and you follow the calling policy established by your company.

### **Dialing an Internal Number**

To dial another extension, just dial the number. For example, if you want to reach Extension 123, just enter "123" followed by the confirmation button and start the call. **123** 

There are several services (like auto attendant and conference mixing) that can also be reached by dialing an extension number.



## Dialing an Outside Number

To dial an outside number, just enter the number that you want to reach. Please note, depending on your company's policy, you may require a prefix to dial an outside number (for example, a "9"). For example, when you enter 914161234567 it will detect this is a long distant number in North American numbering plan and dial the number. Real PBX Service dialing plan require the caller to dial 1+ the number for North America destinations. Other destination must start with the country code like 44 for the UK and need not to enter 011 before it. This feature reduces the need to enter 011 for every international call.

## Hold a Call

To hold a call, press the **Hold** button on your phone (no star code is available for holding a call). If you do not have a **Hold** button but you have a SNOM phone, you can configure one of the buttons on the telephone keypad as a **Hold** button.

Hold can be used for transferring a call, picking up another incoming call, or for providing privacy when a mute button is not available. Hold is also useful for initiating star codes from an IP phone (e.g., to park a call, the call must be placed on hold before activating \*85).

## Redial a Number (\*66)

If your phone does not have Redial feature button, you may dial \*66 and press the Confirm button on the telephone keypad. You can also call the last number that was dialed from your phone by using the Call Log from the web interface (see Chapter 3, "Using the Web Interface," Viewing Important Lists > Call Log).



## Call Return (\*69)

To dial the number of the last call, dial \*69 and press the Confirm button on the telephone keypad. The number will be available until you establish a call to that number (not necessarily connect, however). This can help prevent you from calling back the same person twice. (However, if you use \*69 and discontinue the call during its ring state, \*69 will not be able to access that number again, unless the caller calls again.)

*Notes*: If you reach an external mailbox, the system will handle your Call Return as if the call had been successfully established and will clear the number. If you reach a mailbox of another extension on the system, the Call Return number will not be cleared and you will be able to make successive attempts to reach that extension by dialing the Call Return code. The system will store a Call Return number only for calls that contain a valid caller-ID. Telephones that share the same extension number will share the same redial and Call Return number.

## Intercom (\*90)

The Intercom feature mimics the old secretary-boss intercom systems. Intercom can be thought of as a one-part page with a two-way audio. This feature is used for communicating with one other person (also calling the extension will accomplish the same). The other party must have a separate extension number, as the Intercom feature will not work between two phones that are registered to the same extension. Also, the Intercom feature will not work when multiple registrations are involved with the receiving extension.

*Note*: Permission is needed to use the Intercom feature. If you are unable to activate this feature, see your administrator. Not all vendors support the Intercom feature. Check the user guide that came with your phone for more information.



To activate:

- 1. Dial \*90 and the extension you would like to intercom.
- 2. Press the Confirm button on the telephone keypad.

Example: \*90123 for extension 123

You can now begin conversing with the other party.

Paging (one-way, one-to-many communication) is another feature that can be used to contact someone. See your administrator to have this set up.

## Remote Dialing \*91

The system allows the administrator to assign a dial plan per user. Depending on your company policy, this feature may be used to restrict certain extension from placing outside calls or to place calls to expensive numbers.

If you are not in your office and your company has a strict telephone bill policy, you might want to tell the system that you want to bill an outgoing call to your extension number. This feature is useful if you are located in a room that has no permissions to place outside calls, for example in the kitchen. In order to do this, dial \*91. The system will prompt you for your extension number, your access code and the destination number. If the call established, it will show up in the call data record database as if it has been placed from your office. The call will also use your personal dial plan with the specific restrictions.

## Transferring Calls

Calls can be transferred using either the Transfer button on your phone or star codes. The system supports two types of call transfers: blind transfer and attended transfer.



Blind Transfer

When you activate a blind transfer, the call is transferred with no knowledge as to whether the receiving party will be available to receive the call. A blind transfer can be used for transferring calls to your voicemail or your cell phone.

- 1. Press the Transfer button on your phone, or press Hold and dial \*77.
- 2. Enter the extension number (or phone number) of the recipient.

A blind transfer offers no confirmation of call connectivity. If the number is busy or does not exist or no one picks up, you will not be notified.

Attended Transfer \*77

When using attended transfer, you must speak with the party who is receiving the call to ensure that the call is wanted.

- 1. Put the caller on hold (using the phone's Hold button). The LCD will display the call that is on hold.
- 3. Enter the destination number.
- 4. Once the party on the receiving end is prepared to take the call, press the Transfer button twice (the first Transfer connects the call; the second Transfer clears the music so that the parties can speak. (If your phone does not have a Transfer button, dial \*77 to transfer the call.)

You do not need to press the Hold button to free the call. If the person is not available to take the call, you can press the Hold button to reclaim the call and discuss the available options with the caller.



Transfer a Call Directly to Someone's Voicemail

If your systems administrator has enabled a direct dial number, which allows you to call your voice mailbox directly, you can "blindly" transfer a call to someone else's voice mail. The default for direct dial is "8." If you are unable to use this default, see your administrator.

- 1. Put the call on hold using the Hold button on your telephone keypad.
- 2. Press the Transfer key. (If your phone does not have a Transfer button, dial \*77 to transfer the call.)
- 3. Enter the mailbox prefix key.
- 4. Enter the extension number.

The caller will be transferred to the extension's voicemail. Because the caller will bypass the voicemail timeout, the caller will not hear any ringing.

This method does not need to involve an outside caller. To call a person's mailbox directly, dial 8123.

Example: 8 1 2 3

Call Park (\*85) and Call Park Retrieve (\*86)

Call Park allows you to send a caller to someone whose whereabouts (within paging proximity) are not known. The receiving party uses Call Park Retrieve (\*86) to retrieve the call.

Scenario: A call comes in for Diane, but you do not know where she it, so you put the customer on hold, dial \*85, and press the checkmark button. You then page Diane, "Diane, you have a call on extension 509" (509 is the extension on which the call was received). Diane executes the Call Park Retrieve (\*86) feature, dials 509, and retrieves the call. Park orbits can also be used for parking calls (rather than using your extension). When using a park orbit, enter the extension number of the park orbit after \*85.



Park a call on your own extension

1. Put the call on hold (music plays in the background). If your phone does not have a Hold button, refer to the user guide that came with your phone.

2. Press \*85 to park the call, then press the Confirm button on your telephone keypad.

Example: Hold \* 8 5

An announcement will indicate that the call has been parked. At this point, the call can be retrieved by any extension by dialing \*86 and your extension number.

Example: \* 8 6 1 2 3

If the call is not picked up within a specified time (default is 1 minute), you will receive a call from the system reminding you that a call is waiting.

Park a call on a specific extension:

See your system administrator to determine which extensions are available for parking calls. The system administrator may configure specific park orbits (e.g., 800, 801, 802, and 803) on the system so that all users can park calls to the same place. This is useful for monitoring parked calls and buttons, as well as for applications that need to park calls. Park orbits also ensure that voicemail does not pick up.

- 1. Put the call on hold (music will play in the background).
- 2. Press \*85 to park the call.
- 3. Enter the extension number.

Example: Hold \*85 123

An announcement will indicate that the call has been parked. At this point, the call may be retrieved by any extension (which has permission) by dialing \*86 and the extension number on which the call has been parked.



Example: \*86 123

Call Pickup (\*87)

Call Pickup allows you to pick up a phone that is ringing. It can be used to pick up calls ringing into a specific extension or for ringing calls in general, including calls ringing into a hunt group or agent group.

Note: When compared to Call Park Retrieve, Call Pickup has a sense of urgency associated with it. (You must pick up the ringing phone before the call goes to voicemail.) Call Park Retrieve allows you to take your time, as the caller is already in the system.

Before you can use Call Pickup, you must first put your last call on hold so that the system does not interpret and execute the star key during calls made to external systems that also use star keys.

Directed Pickup

1. Dial \*87.

2. Dial the extension on which the call is ringing.

3. Press the Confirm button on the telephone keypad Example: \*87 123

Note: The Sipura devices from Linksys by default are not able to support star codes with more than two digits.

General Pickup

Call Pickup can also be used without specifying an extension. To activate, dial \*87 and press the Confirm button on the telephone keypad.



Example: \* 87

The system searches for calls to pick up in the following order:

Hunt groups: For a call that is headed for a specified hunt group, the system will stop the hunt group and send the call to the extension that dials the pickup code. If several calls ring the hunt group, then the system will pick up only the first call that went to that hunt group.

Extensions: When the account number is an extension, the system searches for calls that go directly to the specified extension. These calls can come from an auto attendant or through direct extension dialing.

Calls that ring the extension because it is part of a hunt group or an agent group will not be considered by Call Pickup.

Agent groups: As with the hunt group, the system picks up a call from the specified agent group. For calls in the ringing state, the system picks the first call that is in that state. Otherwise, the system picks the call that entered the queue first.

## Mailbox

Leaving Voice Mailbox Messages

When a user is not available, the mailbox system may pick up the call. In this case the caller will hear an announcement to leave a message after the tone. To finish the recording, just hang up.

If the caller press the star (\*) or pound key (#) during the recording, the system will stop the recording. The caller then has three possibilities. If the caller presses the digit 1, the recording will be deleted. If the caller hangs up, the mail box owner will not receive a messages waiting indication. If the caller presses 2, the caller may record a message again. If the caller presses 3, the message will be marked as urgent.



If the mailbox is full, the caller will hear an announcement that makes this clear. If the mailbox is full, but there are saved messages, the PBX will make room for a new message by deleting the oldest saved message.

#### **First Call to Your Mailbox**

Each Extension may have a mailbox. The mailbox picks up messages when you are not available. Depending on the Extension setup, the message system will either store the message in the system or forward it as email message to the email address assigned to the Extension.

To enter the voicemail system, just dial \*97 or your extension number or press the Mail key on your phone.

When you call your mailbox the first time, it will ask you if you want to leave your name. This name will be used in announcements, for example when a caller is asked to leave a message. If you want to record your name, press 1. If you want the system just uses your extension number, press 2. You may record your name later.

If you decide to record your name, wait until the system plays a beep tone, say your name and then press pound. After recording your name, you will have four choices. If you press 1, you can listen to your recording. If you press 2, the system will use the recorded name. If you press 3, you may record the name again. And if you press 4, you tell the system to use the standard announcement and discard the recording.

## Picking Up Mailbox Messages $*_{97}$

If you have a voice message on your mailbox and the message has not been forwarded to your email, your phone should indicate that a message is waiting for you (message waiting indication, MWI). Usually this is done by a message on your phone.

Depending on your phone type, you can press the message retrieval button, which is just a shortcut for dialing the mailbox number (for example, by \*97). The system will automatically start reading out the message.



During message playback the following keys are available:

- "1" will start reverse playback like on a tape recorder. If pressed again, the reverse playback will be faster.
- "2" will stop/resume playback.
- "3" will start fast forward the playback. If pressed again, the playback will be faster.
- $\circ$  "4" will read the message again (without envelope information).
- "5" will read the message again, including the "envelope information". This means the time and caller will be included in the message announcement.
- "6" can be used to move the message into another mailbox. The PBX will ask for an extension number and after a verification move the message to another mailbox.
- $\circ$  "7" will delete the message and move to the next message.
- "8" will leave the mailbox and call the number that has left the message (if that number is not an anonymous number). The PBX will prompt for verification, pressing a star will get the user back to the message readout.
- $\circ$  "9" will save the message in the mailbox.
- "0" will play a help text

After all your messages are played, you will hear the main menu for voicemail box.



## Main Mailbox Menu

From the main mailbox menu you can listen to your messages; change the access code (Mailbox PIN) for your voice mail; or record your name or record a personal greeting.

- "1" will read new and saved messages.
- "2" makes it possible to change the PIN code for the mailbox
- "3" will ask for a new name recording
- "4" will ask for a new personal greeting message.
- "\*" will start dialling a star code. For example, this way you can record a new announcement for the auto attendant of the office while you are in your mailbox
   (a a fram a coll phane while you are traveling)

(e.g. from a cell phone while you are traveling).

Logging into your Mailbox

If you call your mailbox from somewhere outside your office, you will hear the mailbox announcement. The system will assume that you are someone who might want to leave a message.

To authenticate yourself with the system, just enter your access code. If the code matches, the system will start reading out messages.

Usually the mailbox is set up without access code. In this case you must call from your extension into the mailbox and set the access code in the main menu.



Forwarding Messages to Email

If your system administrator has set up your email address, you may forward voicemail messages per email. To turn this feature on, call \*95. Ask someone to leave a message on your mailbox to check if the forwarding works. If there are problems, ask your administrator to

check the email address and the settings for the email server.

If you want to receive the messages via message waiting indication on your phone, call \*96 to disable the email forwarding.

Please note that once a voicemail message has been forwarded via email, it is not available on the system any more.

Call Forwarding

There are several events that may trigger the redirection of an incoming call. The system differentiates between the following cases:

Always: All incoming calls are forwarded. This case can for example be used when you are on holiday and a colleague is answering your phone or you want to redirect all calls to your cell phone.

No answer: Calls are redirected after a certain time when the extension does not pick up.

Typically, you use this to redirect incoming calls to an assistant or to your secondary phone in case that you are not sitting next to your telephone.

Busy: Calls are forwarded if the extension is busy. When several extensions are used, the busy condition checks if all devices are busy.

Typically you will program this number to redirect calls to a team assistant or to a colleague, so that incoming calls can be answered by



someone when you are busy in another call. The busy condition is also used for anonymous calls, see the description below.

The cases are handled independently. That means, you can program all three forwarding events; the system will use the settings whenever they apply.

To turn the call forward always on, dial \*71. You will hear an announcement that asks you to enter the call forwarding number. Enter the number and press the pound key. The system will repeat the number and then hang up. If you want to change the number, just call \*71 again.

Instead of programming the redirection number by interactive voice response, you can also dial the code plus the redirection number and then start the call. This has the advantage that you can check the number on the display of your phone before starting the action and you may preprogram that number of your extension.

To disable call forward always, call \*72. You will hear an announcement that call forwarding has been turned off.

Programming call forwarding on busy works in a similar way. Call \*73 to turn it on and to program the forwarding number. Like with the other codes, you can also put the number directly behind the star code.

Call \*74 to turn call forward on busy off.

To turn call forward on no answer on, dial \*75. Like with the other codes, you can also put the number directly behind the star code. The time after which the system forwards the call cannot be set by pressing keys, it must be changed by the system administrator. Please contact your system administrator if you want to change this value.

To disable the call forward on no answer, dial \*76.

Do Not Disturb

Do not disturb (DND) is similar to the call forward described in the previous section. However, the typical case is that you



temporarily don't want to be disturbed (for example, because you have a meeting). To turn DND on, dial \*78.

To turn DND off, dial \*79.

If you are member of a hunt group, the call forwarding conditions for your extension will not apply (because those calls goe to the hunt group, not to your extension). But if you turn DND on, even calls that go to a hunt group will leave your phone silent.

Most VoIP phones have a button that act in a similar way. However if you are using a analog terminal adapter, this function might not be available. And a phone is usually not able to distinguish between urgent calls and regular calls. The system can differentiate between these calls, therefore the DND programming on the PBX has advantages against the DND setting locally on the phone.

Redial and Call Return

To redial the last number, just call \*66. This functionality is also available on most VoIP phones, but the system functionality will survive reboot cycles of the phone.

Call return (\*69) will dial the number of the last call that you have missed. The number will be available until you establish a call to that number. This will avoid that you call the back the same person twice, however if you are hitting an external mailbox the system will believe that the call was established successfully and clear the number. If you are hitting a mailbox of another extension on the same system, the number will not be cleared and you will be able to try to reach that extension later by dialing the call return code. The system will store only calls that contain a valid caller-ID.

Please also note, that if you have registered several endpoints to the same extension, these devices will share the same redial and call return number.



Caller-ID Treatment

Caller-IDs are usually telephone numbers. By providing your caller-ID to the party that you are calling, the other party is able to look up the address book, to initiate a callback or just to see who is calling.

Sometimes you want to reject calls that come from callers who do not want to reveal their identity.

A typical example is a call from a sales person that is making cold sales pitches. A caller-ID is anonymous, if it does not come from a local extension number and contains a non-numerical digit in the SIP ID. Most VoIP systems use the name "anonymous" in the caller-ID in such a case.

To treat these calls, dial \*88. If you have set the call forwarding on busy condition, the call will forward anonymous calls to that destination (for example, your assistant). This way, you can make sure that anonymous calls get screen first before you take the call. If you don't have set the call forward on busy, the system will play an announcement to the caller that tells him that the call can not be taken because they block their caller-ID.

To allow anonymous calls again, dial \*89.

If you want to place a call without showing your caller-ID, dial \*67. You will hear an announcement that the caller-ID will be blocked for all future calls.

To enable caller-ID again, dial \*68.

Please note that the caller-ID will always be presented for internal calls.



#### Call Data Record

Sometimes you receive a call from someone that you have to call back. If your system administrator has set up your extension with your email address, instead of asking the person on the phone to spell the phone number and scribble it down on a notepad, you can direct the PBX to send you an email with the call details.

To receive a "data record", dial \*63 after the call. The system will send you an email with the Caller-ID, the duration of the call and the time of the call.

#### Call Pickup

If you want to pickup a call that is ringing another extension, use the \*87 code.

There are two ways or picking up calls. The directed call pickup is used to pick up a call from a specific account, while the undirected call pickup searches for a call that can be picked up.

The directed call pickup uses the call pickup code together with the account from which a call should be picked up. For example, \*87123 will tell the PBX to pick a call up from account 123. Please note that the code must be dialled as one number. The call pickup can be done for the following account types:

• Hunt groups. If there is a call that goes to the specified hunt group, the PBX will stop the hunt group and send the call to the extension that dials the pickup code. If there are several calls that ring the hunt group, then the PBX will pick up the first call that went to that hunt group.



• Extensions. When the account number is an extension, the PBX will search for calls that go directly to the specified extension. These calls can come from an Installing the IP-PBX Appliance Page 13 auto attendant or through direct extension dialling. Calls that ring the extension because it is part of a hunt group or an agent group will not be considered for call pickup.

• Agent groups. Like with the hunt group, the PBX will pick up a call from the specified agent group. If there are calls in the ringing state, it will pick the first call which in that state. Otherwise, it will pick the call that entered the queue first. The undirected call pickup goes through the list of calls and searches for calls that can be picked up. It will use the following preference for this:

- It will first try to pick up calls from a hunt group.
- $\circ~$  If no call has been found, it will try to pick up a call for an extension.
- If that also is not possible, it will try to pick up a call for an agent group.

For all pickup attempts, the extension that attempts the pickup must have the Dialog Permissions for seeing the state of the account. If this is not the case, then the PBX will not allow picking the particular call up.

## **Call Park and Retrieve**

While in many cases you may dispose of a call by transferring it to a specific extension, in certain situations you will want to "park" a call so that it can be picked up by an unspecified extension.

Parking a call so that it may be picked up by any extension is a two-step process. First put the call on hold; then, dial \*85 to park the call. You will hear an announcement that the call has been parked. At that point, the call can be picked up by any extension that dials \*86.



Again, your telephone system administrator may have configured your PBX with a more strict pickup policy. In that case, you will specify a group to which you are parking the call (\*85610 for example for group 610). Any member of that group can then pick up the parked call by dialling \*86.

Attended and unattended transfer

If your telephone has a transfer button, the PBX will receive the transfer signal and switches the call to the provided destination. There are generally two types of transfer. The simplest transfer is called a "blind" transfer. The other type is often referred to as an "attended" or a "consultative" transfer. In an attended transfer, you speak with the party to whom the call will be transferred to ensure that the call is wanted. In a blind transfer, you simply transfer the call with no knowledge of whether the person called will be available to take the call.

To initiate a "blind" transfer, just press the transfer button on your phone and dial the extension to which you are transferring by pressing the extension number keys and the call start key. To initiate an attended transfer, first put the caller on hold (using the hold button on the phone); then dial the number to which you will transfer the call.

If the person on the receiving end of your call is prepared to take the call you can just press the transfer key and the call that is on hold will be transferred. You do not have to press the hold button to free the call. If the person is not available to take the call, you can press the hold key to reclaim the call and discuss the caller's options with him or her.



Transfer with \*77

Many mobile devices do not have options to transfer calls. However they are able to put a call on hold. The PBX supports blind transfers for such devices in the following way:

- Put the call that should be transferred on hold;
- Then dial \*77 followed by the destination number. This number must be dialled as one number. For example you would dial \*771234, not \*77 and then press 1234 by DTMF.

Then the PBX will put the call off hold and redirect it to the destination. Please notice that a blind transfer does not check if the call will be connected. If the number is busy, does not exist or just does not pick up, the PBX will not send automatically the call back.

Transfer requires that you dial the star code together with the transfer destination. That means you cannot first dial \*77 and then the destination number. You must dial the transfer destination together with the star code, for example \*771234. If you want to transfer a call to an external location, you must also have a dial plan for the extension that initiates the transfer. Either you must have a default domain dial plan or you must assign a dial plan directly to the extension.

Also it should be noted that most SIP devices support the REFER method to initiate a transfer. This method does not need a star code.

### **Recording Prompts**

The attendant and the agent group support the recording of customized greetings. You can record those greetings from any telephone in your network.



Consult the system administrator manual if you want to restrict the extensions that are allowed to record greetings. In order to record a greeting, dial \*98 followed by the account number (e.g. \*98123 if 123 is the number of your auto attendant). The PBX will prompt you for your new prompt. If you just want to delete the old prompt and use the standard prompt, press the star key during the announcement.

The agent group has up to ten announcements. The announcement with the number 0 is the initial announcement; the announcements 1 to 9 are placed in a loop. In order to record those announcements, you need to dial \*98 + the group number + \* + the announcement number (for example, \*98123\*4 to record announcement 4 in the account 123).

## Call Mixing

One of the benefits of the PBX architecture is that existing calls can be interrupted and monitored. All three modes, and especially the listen in mode, are severely affecting the privacy of the calls on the PBX. Therefore, those modes are only available to extensions that are specifically allowed to use those features. In the permissions tab of the respective extension, the domain or system administrator has to enable the features. Please consult corporation and government regulations if turning these features is allowed in your environment. Illegal listening to phone calls is a severe crime, and system administrators must be aware about that.

To see which calls are active, you can monitor the extension's state of the state of a CO-line. For example, you can do this by using the LED key of a SIP phone with the associated display.



## Call Barge-In

In call barge in, two persons that are talking to each other are put into a kind of conference with a third person. Typically, this third person is a secretary reminding ("saving") the boss about another appointment. Both parties will hear the third person come into the call ("knock knock") and both parties will be able to hear what the third person has to say. The existence of the barge in call depends on the existence of the underlying conversation.

In order to barge into a call, dial \*81 followed by the extension number that you would like to interrupt, then press dial.

## Call Teach-Mode

In teach mode, only one side of the call can hear the third party. This is typically useful in a call center when a trainer wants to give tips to a new agent, so that the customer does not know about the teacher in the background. This mode is also sometimes called whisper mode, because the agent's phone must have a real good echo cancellation so that the customer does not hear some background echo.

In order to start the teach mode, dial \*82 followed by the extension number that should hear your voice, and then press dial.

### Call Listen-In

The listen mode is similar, but completely stealth mode. The two persons talking to each other are not notified about the listen in and cannot hear what the third person says on the phone (e.g. breathing). In order to start listening to calls, dial \*83 followed by the extension number that you would like to monitor.



### Hot Desking

"Hot Desking" makes it possible that employees temporarily or permanently change the routing of all their calls to a specific physical device. Then all calls to that extension, also as part of the hunt group or agent group, get routed to that extension. Hot Desking means that the person takes ownership on the phone. That means that outbound calls from this phone will show his caller-ID. It is not expected that other significant inbound traffic goes to the originally registered extension.

This fact suggests that offices use "virtual" and "real" extension numbers:

- "Real" extension numbers are used for employees with a fixed location (e.g. switch board, management)
- "Virtual" extensions don't have any registrations.

They are just used for routing calls to a specific user. They use real extension numbers that are not assigned to any other person, so that there is no conflict between identities on a specific physical device. In cases when an employee just wants calls to his extension being routed to a colleague's office, it is better to use the unconditional redirection feature of the PBX. This feature must be turned on before the user leaves his office.

### Logging In

When you want to log in, enter the Hot Desking star code (typically "\*70"). The PBX will prompt for the extension number and the PIN code for that extension number and acknowledge the Hot Desking with a "the service is active now".



## Logging Out

In order to log out, you just need to call the Hot Desking star code from a location that is currently registered as hot desk or from the phone that holds a registration for extension in question. The PBX will answer with a "the service is inactive now".

## Limitations

Hot Desking has limitations. Because the configuration of the device does not change during Hot Desking, you will not be able to move telephone preferences (like ring tones, address book programming, etc.) to another desk.



## **Domain Administration**

Domain administrators can manage the Domain by accessing the Web based interface.

REAL TELLED	GE	
Login		
Please ente	r your login information:	
Account:	101@real	
Password:	••••••	
Login Type:	Automatic -	
	Login	
🔲 Remember	login information.	
Copyright © 2005 the license agreer	-2007 Real Telecommunication Corpo nent for more information.	oration. All rights reserved. See

To log in as domain administrator, you must enter the username and domain name in the "user@domain" form and enter the password, for example "123@test" and the password is the Domain Administrator's extension (123) password.

The domain administrator flag is used to control the permissions of the extension. If the flag is set to true, the web interface will accept the user's login (same as extension registration) and allow them to change the settings of the domain.





We will review each option listed above. Each enables you to perform various activities that enable more functionality and more features for users of the domain.



AL TELLEDGE	ettings Lists Status				Help Lo
G	neral Redirection Mailbox Email	Instant Message			
General settin	gs for account 101				
first name (e.g. John):		Rashid			
last name (e.g. Smith):		Jamalalideen			
SIP Password:		•••••	]		
SIP Password (repeat):			]		
Web Password:			]		
Web Password (repeat):			]		
PIN (e.g. 1234):		•••••			
PIN (repeat):		•••••	]		
limezone:		Default Time Zone	_		
VR Language:		English	-		
Web Language:		English	•	Desures	
Vakeup time (e.o. 7:15):			]	(_browse)	
Natch the calls of the foll	owing extensions (* for all):		1		
Watch the presence of th	e following extensions (* for all):				
Explicitly specify park orb	it preference:		í		
Dialog Permissions:			í		
Block outgoing caller-ID:		€ noy@s			
Save					

The settings "First name" and "Last name" store the display name of the extension. This name will be used for internal calls. Phones that are able to display the names will show it on the screen. These names will also be searched by the auto attendant if the caller chooses the dial by name feature.

The "Password" is used for SIP and HTTP traffic. It should be reasonable safe; just a few digits of digits will not be enough. However, it is up to the user to choose a safe password. When the user enters a new password, the current HTTP session will stay valid until the next login. Permanent cookies will be invalidated on the next login.



The "PIN" is used in several dialogs of the PBX, for example when entering the mailbox. Although it is also possible to do brute force attacks on PIN codes, it takes more time to perform such attacks. We recommend using at least 5 digits, so that an attacker has at least 100,000 combinations to choose from.

The "Cell phone number" is associated with the extension. When someone tries to call the extension directly, the PBX may include that number in the list of devices that is being called. The setting "When calling the extension" controls if and when the cell phone is being called. For more information, see Cell Phone Integration for more information.

The "Timezone" tells the PBX in which time zone the user is. This setting is used for example in the mailbox (reading out timestamps), but it is also used when the PBX generates configuration files for attached phones.

The "IVR Language" settings controls which language the user prefers. This is setting is used for example, when the user calls the mailbox or when the caller calls into the PBX from the associated cell phone.

The "Web Language" is used when the user logs into the web interface of the PBX. This setting is also used when emails are sent to the user.

The "Watch the calls of the following extensions" is used for automatic generation of the configuration files of the phones. The PBX will try to put the listed extensions into the settings for the phone. This setting depends on the used device. For this setting you can use Wildcard Patterns. For example, if you use just a star, the PBX will try to put all extensions of the domain into this list.

The setting "Watch the presence of the following extensions" is similar, but it does not provision calls, but asks the device to subscribe for the presence state of the listed extensions. The PBX itself will just forward the presence state of the phones; the phones themselves are responsible for generating the presence information.

If the "Limit own visibility to this list" is set, the PBX allows only the listed users to watch the extension. It also uses Wildcard Patterns.



If you upload a picture in the "Upload a picture" setting, the PBX will automatically insert a SIP header that can be used by SIP phones to show the picture of the extension. This is similar to the display name of the extension, with the difference that in addition of a text a picture is rendered.

The picture must be in BMP format that the endpoint understands.

The setting "Block outgoing caller-ID" is used when making outbound calls. If the setting is set to "yes", the PBX will ask the gateway or SIP provider to hide the Caller-ID for this call.

User Redirection Settings

Do not disturb:	⊖ on ⊙ off
Incoming anonymous calls:	Pretend to be busy
Hot Desking at:	
Call forward all calls to:	
Call forward calls when busy to:	
Call forward on no answer to:	9784353234
Call forward no answer timeout:	15 sec 💉
Save	

The "Do not disturb" (DND) settings is used to keep the user from incoming calls. If this setting is turned on, then this extension will not receive incoming calls. Only extensions that have the permission to override DND will be able to call the extension. However, the extension can a all times place outbound calls.



- The setting "Incoming anonymous calls" controls how incoming calls to the extension are treated. This setting does not affect calls that come from a hunt or agent group or internal calls.
- The "no special treatment" mode turns this feature off and sends all calls directly to the extension.
- The mode "reject call" will inform the caller that the extension does not accept anonymous calls. Calls will not be put through to the extension.
- $_{\odot}$  The mode "pretend to be busy" also does not put the call through, but does not tell the caller why. The PBX will not offer camp on.
- The mode "ask for name" will prompt the caller for the name and then put him on hold while calling the extension. After reading out the recorded name, the extension may decide what to do with the call. Either the call can be accepted, or it can be sent to the mailbox or it can be rejected.
- The mode" ask for name even if the caller-id is present" will always ask for recording the name, even if the caller-id is available. Only callers which are on the white list are directly put through.
- The "Hot Desking" setting is described in the separate page Hot Desking.
- There are several ways of doing call forwarding.
  - If you want to temporarily redirect all calls to another extension you can use the "Call forward all calls" setting. This setting affects only calls that are going directly to the extensions. If the extension is part of a hunt group or an agent group or the call origins from a paging group, the call is not being redirected.
  - The "Call forward calls when busy" forwards calls only when the extension is busy. This condition is either true if



the phone itself signals it is busy or the "Lines" parameters has been set for the extension and this number has been reached (see Registrations).

 The Call forward on no answer" kicks in when the phone rings, but there is no answer. The waiting time is defined in the domain and can be overridden by a user setting ("Call forward no answer timeout"). If the mailbox picks up earlier, this setting has no effect; if the call is being redirected, the mailbox timeout is cancelled.

All redirection settings can have one number, either internal or external. If the call is being redirected to an external destination, the PBX will use the dial plan of the extension for placing this call and it will charge the extension for this call.

User Mailbox Settings

Mailbox Enabled:	⊙ yes 🔘 no
Time until mailbox picks up:	Default value 💙
Announcement Mode:	Name Announcement
Send message waiting indication:	⊙ yes ◯ no
Call cell phone when new message arrives:	🔿 yes 💿 no
Share the mailbox with the following extensions:	502 503
Save	

The user may decide if the mailbox should be enabled or not. For this purpose, the web page offers the setting "Mailbox Enabled". As with the redirection setting, the user may override the default setting until the PBX redirects the call to the mailbox. This can be done in the setting "Time until mailbox picks up".

The "Announcement Mode" can have the following values:

• If the user selects the "Anonymous Announcement" mode, the PBX will just read out the number of the extension.



- If the user selects "Name Announcement" the PBX will pay back the recorded name of the extension, if present.
- The setting "Personal Announcement" will read out the recorded announcement of the extension, if present.
- If the PBX should send a notification to registered phones, the setting "Send message waiting indication" must be turned on.

If the PBX should call the cell phone after a message has been recorded, the setting "Call cell phone when new message arrives" needs to be turned on. The PBX will charge the extension for this call.

If the mailbox should be shared with other extensions, those extensions can be listed (separated by space) in the setting "Share the mailbox with the following extensions". If the mailbox does not require a PIN code, those extensions can directly dial into the mailbox and listen to messages.

Those extensions will also receive the message waiting indication, if the sending has been activated and the phones register for MWI events.

### User Email Settings

Email Address (e.g. abc@company.com):	greg@burbonizer.com
Send a mailbox message by email:	Send message as attachment to email
After sending a message:	Mark message as read
Send email on missed calls:	⊙ yes ◯ no
Send email on status changes:	⊙ yes 🔿 no
Save	

The setting "Email Address" tells the PBX where to send email messages.

The domain administrator must enable Email, so that this setting actually enables sending out emails.



The setting "Send a mailbox message by email" defines how the PBX sends emails out:

- "Send emails without attachments" will just send a notification to the user by email. The voicemail message itself is not sent and must be retrieved either through the web interface or by calling the mailbox. This mode has the advantage that the emails are relatively short; this way it is a useful feature when using mobile devices that support reading emails.
- "Send message as attachment to email" will also send an email, but put the voicemail itself as attachment to the email. This is a good choice if you are using email during the whole day and you are using a personal computer for processing emails.
- "Do not send an email" is a good choice if you are just using your SIP phone to receive voicemail messages.

You can decide what you want to do with the message after an email has been sent. "Keep the message as new message" will keep the message in your mailbox and it will count as a new message. This mode has the potential danger that your mailbox eventually might become full. If you choose the mode "Mark message as read", the PBX will still keep the message in your mailbox, but when the mailbox is becoming full, it will drop the oldest message to make room for a new message and your mailbox will not become full. However, in this mode most SIP phones will not turn on their message waiting indicator (because only new messages trigger this behaviour). If you decide to "Delete the message", the PBX will delete the message after sending the email. This keeps your mailbox clean, but the system relies on a reliable transport of the voicemail.

If you want to receive emails when you miss a call, turn the setting "Send email on missed calls" on. The PBX will send this email only if the call went directly to your extensions; calls to a hunt group or agent group do not count.



The PBX can also send you emails when your status changes "Send email on status changes". This email is being sent when the DND status or the redirection changes.

#### User Instant Message

Destination:	502 503			]	
	Hello,	can you	see	this?	^
Massagas					
Message.					
					~
Create					

Usually you would use a SIP endpoint to send instant messages (IM) to another extension. However, most of the SIP devices do not support that feature today. Therefore, the PBX offers a possibility to do that from the web interface.

Sending an IM is simple. Just enter the destinations in the "Destination" field and the text that you want to send in the "Message" field, then click on the "Create" button. You can send the message to more than one destination by listing the extensions separated by spaces. The PBX will then send the message to all registered devices for the extensions that you selected.

The PBX will tell you how many messages have been sent in total.



#### Scheduling Conferences

Users of the PBX may schedule conferences. In order to do this, the domain administrator has to set up at least one conference room that allows scheduled conferences.

Creating a new conference

REAL TELLEDGE Settings Lists Status Mailbox Conference Missed	l Calls Call Log Address Book	Domain Addresses	Неір
Conferences organized by 1	01		
Name Start Poom Moderator DI	N Participant DIN A	tion	
	in Farticipant FINA		
Create a new conference			
Name:	Marketing Meeting		
Date (MM/DD/YYYY or MM/DD, no entry = today)	01/29/2008		
Start time (HH:MM):	11:00 AM		
End time (HH:MM):	12:00 AM		
Participants (e.g. "101; 102; user@domain.com"):	10; 102, 201@real		
Record the conference:	© on		
Poom:	Real (300) 🔻		
Koom.			

In order to set up a conference, you need to specify a conference name, the start time and the end time. The name is just used for describing the conference, so that the participants know the subject of the conference.

The start time must be in the format MM/DD HH:MM (month and day, hours and minute). The PBX will use the selected time zone of the extension that is logged in. The end time must in the same format, however you may leave out the day information. In this case the PBX will automatically choose the day information of the start time stamp.



You may also list the participants. If your domain has been set up for sending out emails, the PBX will send out an invitation to the listed participants. The list of the participants must be separated by semicolons (this is different than in most other places of the system). You may either specify the email address of the participant or the extension number. If the extension number has an email stored, the PBX will use this email address to send out the email. The email contains a calendar event, which can be read by most scheduling tools. The tool will then popup a conference reminder with the relevant conference information.

## Editing existing conferences

Currently, it is only possible to delete an existing conference. To do this, just click on the delete icon in the conference list.

Conferences are automatically deleted after 24 hours. If there are still

participants in the conference, they may continue talking.

## In a conference

If you are the creator of a conference and also participate in the conference, you can use the following special codes to control the conference:

• \*9 removes all other participants of the conference. This is useful when you want to make sure that the conference is cleared up before you leave the conference yourself.

### Address Book

The address book stores associations of numbers with names, speed dial entries and types. Numbers are generally telephone numbers. The PBX does not support SIP URI numbers in the address book. Numbers may contain readability characters.

The PBX internally converts the numbers into purified numbers, so that matches with other numbers become more easy and consistent. If your domain has selected a specific dial plan scheme (for example, North American dial plan), the PBX also internally automatically



converts the number into a consistent presentation. If the PBX requires that a name must be presented as one string (for example, in the SIP display name), it will automatically put a space character between the first and the last name if both names are not empty. Names consist of the first name and the last name. The purpose of this separation is to make searches easier; sometimes the user searches for first and sometimes for the last name. Otherwise the PBX treats the names transparently.

Names should be encoded in UTF-8 format. The web browser usually performs the necessary conversations, so that the end user does not have to deal with this problem.

### Speed Dial

The speed dial entry is a two-digit star code (for example, \*12). The range should be in the lower feature code range, so that they do not overlap with the other feature codes that are available on the system. If there is an overlap, the speed dial numbers have a higher priority.

Speed dial numbers are useful if you want to store numbers that you are using frequently, especially if your telephone does not support an address book. Used in the domain mode, they also might be useful to hide the number that is being dialled (however, that kind of security is very weak).

### Black List and White List

An address book entry also has a type flag. This flag can be set to "white", "black" or unset. If it is "white", that address book entry is on the white list, if it is black, it is on the black list. The white list usually contains the contacts that are trusted and therefore get a preferred treatment. The black list usually contains the list of contacts that are known and unwanted. For example, it makes sense to put a family member's number on the white list, while the caller-ID of an aggressive sales person might end up in the black list.

If that flag is set to "white", the number is part of the white list. That



means, when a caller has the type "white", he will never be intercepted by the auto attendant to record the name.

Callers on the white list are allowed to receive a call back when the extension becomes available.

If a caller is on the black list, the behavior of the PBX depends on the settings of the user for anonymous call treatment. If the caller should be blocked, then the PBX will block that call. Otherwise the PBX will always ask the person to leave his name before calling the extension. Callers on the black list will also no be allowed to camp on an extension.

### **Personal Address Book**



The address book entries in the personal address book are only visible to the user. To see the personal address book, you must be logged in as user in the web interface of the PBX. In this screen, you can edit and delete address book entries by clicking on the edit and delete button.



#### Domain Address Book

The domain address book entries are visible to all members of the domain.

That means they can search the domain address book and calls coming to that domain will automatically see the name of the calling party.

All names in a domain are automatically included in names searches in the domain. You can also use the address book to indicate which DID number has been dialled. If you add an address book entry that matches the specific DID, the PBX will add the display name to the caller-ID, and the phone then can display the text associated with the caller-ID.

Address Book Import

Most address management programs support the export of a CSV (comma separated value) file. The PBX can read this format, if the below guidelines are kept.

• The field separator must be a semicolon.

• The first column must contain the first name; the second column must contain the last name.

• The third column must have the number. The number must have numeric characters. The characters "-", "(", ")", " " (space), "/" and "." are ignored and may be included for readability. The first character may be a ,+', so that global telephone numbers may be used. For example, the numbers "(978) 543 6545" and "+49 (30) 386-12345" are valid numbers, while the text "WIRELESS CALLER 9785436534" is not a valid number.

• The fourth column may have a speed dial number. The number must contain a star character in the beginning. If



the star code was already used by another entry, the PBX will clear the other star code entry during the import process.

• The text must be encoded in UTF-8 format.

Because the number must have numeric characters, it is ok if the first contains the field description. For example, most programs use a name like "Number" to identify the column, and because that name is not a phone number the system will not include that row in the address book.

The following example shows an address book that can be imported:

First Name;Last Name;Number;Speed Dial Fred;Feuerstein;(978) 123 4567 Carl;Clever;(212) 324 4334;\*12 Franky;Fahrenheit;(515) 234 3334 "A;B;C";;(123) 543 3453

User Status

The user status page shows essential information about the state of the account.

You can see the settings for call redirection, the current state of do not disturb (DND), and the last number that have been dialled or received from that extension.