OnDO PBX

Version 1.3

Administrator's Guide

Brekeke Software, Inc.

Version

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1. Introduction

OnDO PBX is a software-based IP-PBX that is compliant with the IETF standard protocol Session Initiation Protocol (SIP). This product has the basic functions that a conventional PBX has, including Call Transfer, Call Forwarding, Ring Group, Auto Attendant, Voicemail, etc.

This document explains the installation and basic configuration of OnDO PBX.

2. System Requirement

OnDO PBX runs on the following environments:

OS	Microsoft Windows XP/2000, Red Hat Linux 7.x/8.x
Java	JDK 1.4 or newer

✓ *You need to install Apache Tomcat 4.1.2 or later if you use an OS other than Windows.*

3. Installation (for Windows)

For instructions on how to update OnDO PBX, please refer to [5. Upgrade].

3.1. Step 1: Installation of J2SE

Access the website http://java.sun.com/products/, search for Java2 Platform, Standard Edition (J2SE). Then download and install the latest version of Java2SDK for the type of OS you are running.

3.2. Step 2: Installation of OnDO PBX

Obtain the file “pbx<version_#>.exe” from Brekeke Software Inc., and start the installer by double-clicking the file. Continue the installation by following the installer’s instruction. OnDO PBX and OnDO SIP Server for PBX will be installed automatically. If you check [Run OnDO PBX] at the last stage of the installation and click the [Finish] button, OnDO PBX HTTP service will start automatically.

3.3. Step 3: Starting OnDO PBX HTTP service

If you didn’t check [Start OnDO PBX] at the last stage of the installation, please start OnDO PBX HTTP Service by the following method.

- ✓ Open [Control Panel]**>** [Performance and Maintenance]**>** [Administrative Tools]**>** [Services], then select and start [Brekeke OnDO PBX].
- ✓ Restart the machine. OnDO PBX HTTP service will start automatically.

3.4. Step 4: Starting OnDO PBX Administration tool

1) Select [Start]**>** [Program]**>** [Brekeke]**>** [OnDO]**>** [OnDO PBX 1.x]**>** [OnDO PBX Admintool].
2) You will be asked to enter OnDO PBX Product ID. Enter the 16 digit long product ID in the given space. Entering the same product ID on multiple machines is not allowed.
3) You will see the Login screen of OnDO PBX Admintool.
4) Enter User ID and Password. Then click the [Login] button.

   [Default] User ID & Password (Case sensitive)

User	sa
Password	sa

5) If you wish to change the display language of the Admintool, please do the following: Select [User Setting] **>** click on the Edit button of Admin user **>** Select desired language

from [General setting]-[Language].

6) Click the menu [Start/Shutdown]. If you see “RUNNING” for both PBX Status and Media Server, the OnDO PBX server was started successfully. If you see [Inactive], the OnDO PBX server failed to start.

✓ *You can install OnDO PBX on Windows Operation Systems using the same method explained in the chapter 4.Installation (for Red Hat Linux). In this case, you’ll need to have the files msvcp60.dll, msvcrt.dll in the Windows system directory (usually C:\windows\system or C:\winnt\system32).*

4. Installation (for Red Hat Linux)

For instructions on update OnDO PBX, please refer to [5. Upgrade].

1.1. Step 1: Installation of J2SE SDK

Access the website http://java.sun.com/products/, search for Java2 Platform, Standard Edition (J2SE). Then download and install the latest version of Java 2SDK for the type of OS.

4.1. Step 2: Installation of Tomcat

1) Access the website http://jakarta.apache.org/site/binindex.cgi and download the binary file of Tomcat version 4.1.2 or later for the type of OS you are running.
2) Set J2SDK Install directory for the environment variable JAVA_HOME.
3) Install using the download file.

4.2. Step 3: Installation of OnDO PBX and OnDO SIP Server (bundled with OnDO PBX)

1) Obtain files “pbx.war” and “oss.war” from Brekeke Software.
2) Copy those 2 files directly into the \webapps directory which is under Tomcat install directory.

4.3. Step 4: Starting Tomcat

1) Start Tomcat.
2) Open a web browser and specify http://localhost:8080 as a URL (If you chose a port number other than default “8080”, specify the appropriate port number in the URL.)
3) Tomcat has started successfully if the Apache Jakarta Project page is displayed.

4.4. Step 5: Starting OnDO PBX Administration tool

1) Open a web browser and specify the URL http://localhost:8080/pbx. (If you chose a port number other than default "8080", specify the appropriate port number in the URL.) You will see the Login screen of OnDO PBX Admintool as below.
2) Enter values for User ID and Password. Then click the [Login] button.

[Default] User ID & Password (Case sensitive)

User	sa
Password	sa

3) If you wish to change the display language of the Admintool, please do the following: Select [User Setting] **>** click on the edit button of Admin user**>** Select desired language from [General setting]-[Language]
4) Click the menu [Start/Shutdown]. If you see "RUNNING" for both PBX Status and IVR / Voicemail, the OnDO PBX server was started successfully. If you see [Inactive], the OnDO PBX server failed to start.

5. Update

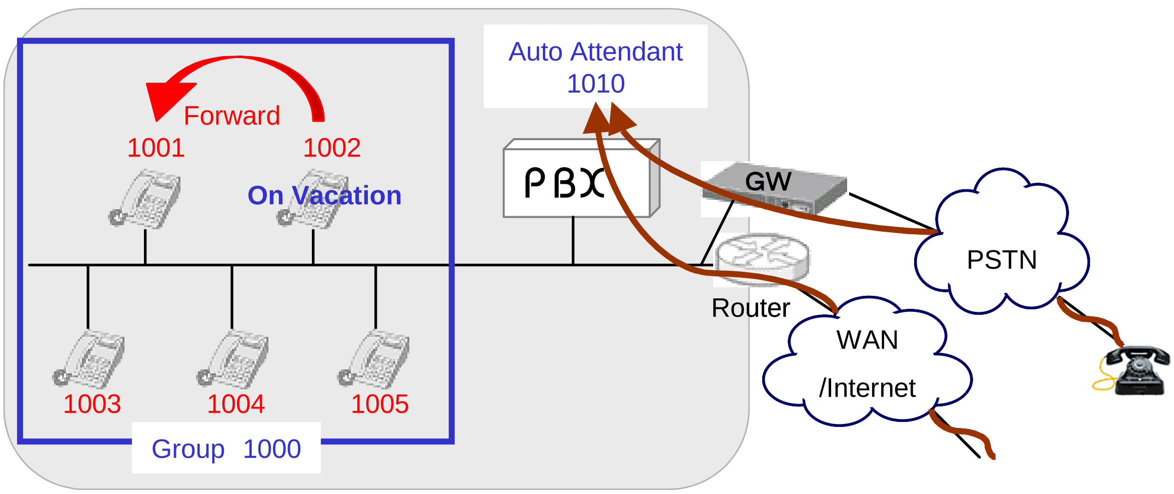
Updating OnDO PBX can be done through [Update software/Service pack] without losing your current customized settings.

1) At the menu [Start/Shutdown], confirm that OnDO PBX Status is [Inactive].
2) Open the menu [Option], scroll down to the bottom of the screen to find [Update software/Service pack].
3) Press [Upgrade] button to open another screen.
4) Press [Browse…] button to select the appropriate file (ex. *.war) to update.
5) Restart your computer.

6. An example of how to set up OnDO PBX

The following is an example to illustrate what you can achieve using **OnDO PBX**.

- *5 employees in the office (assigned extensions: 1001-1005)*
- *Mr. Smith at extension 1002 will be absent from work while he is on vacation*
- *Calls that come in to Mr. Smith will be forwarded to Mr. Burns at extension 1001*
- *Ring Group*[i] *is set for extension 1000. When calls are received at extension 1000, all employees' phones will ring at the same time.*
- *Extension 1010 is set for Auto Attendant. Calls to extension 1010 will be answered by the automatic operator.*



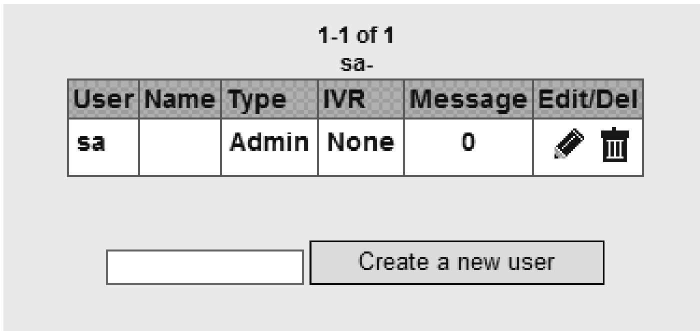
[i] A group of local extensions that ring in unison. When calls are made to the group number, any available group member can pick up the call.

6.1. Registering users

This section explains how to set up extensions for users.

OnDO PBX Admintool > User Setting

1) Choose the **[User Setting]** menu and open the **[User Setting]** window.



2) Enter “1001” (User ID) in the entry field and click [Create new user]. A new user will be added to OnDO PBX. Continue adding other users for extensions 1002 to 1005.

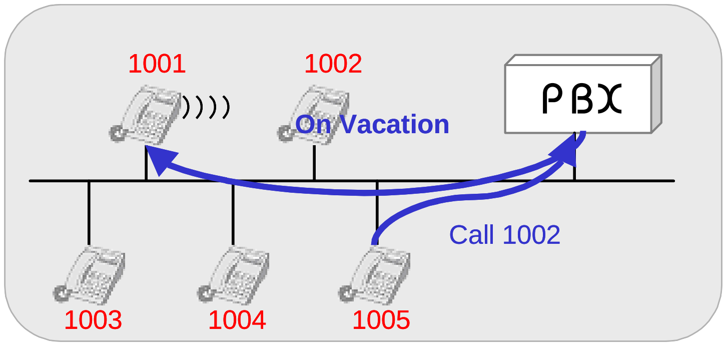
The default values of the user settings are:

Setting item	Details
Language	Same Language as the administrator
Password	Same password as the administrator
Ringer time	90 seconds
Call forwarding (No answer/Busy)	Forwarded to user’s Voicemail[ii]
Call Pickup group	Same group as the administrator
Voicemail greeting message	Default system greeting
Email notification when a new message arrives	No

✓ *To change the default setting, click on the Edit button to change the default settings.*

[ii] Access number for users’ voicemail is set in the **[Phone number setting]** menu under the **[Option]** menu. The default prefix for voicemail is set at 7. (e.g., 71001 for extension 1001 user)

6.2. Unanswered call forwarding setup

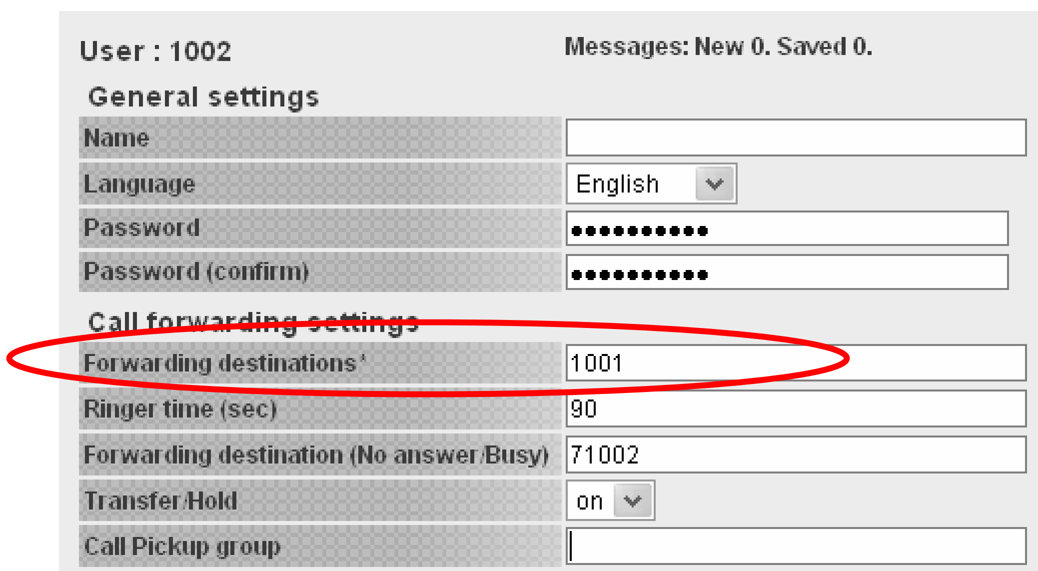


The user at extension 1002 will be temporarily out of the office. Calls to extension 1002 will be forward to extension 1001.

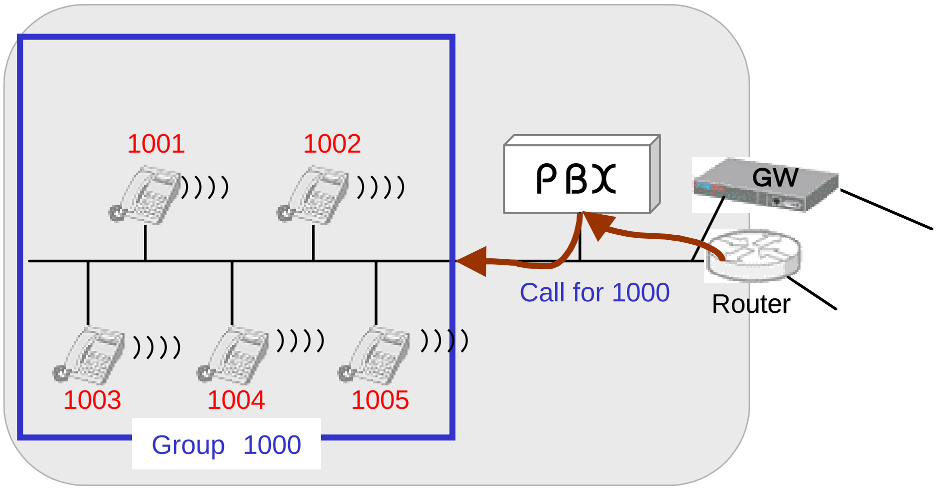
OnDO PBX Admintool > User Settings > Edit

1) Click the menu **[User Setting]** to open the **[User Setting]** window.
2) Click on the **[Edit]** button for user 1002 and open the edit window for the extension.
3) Enter the extension number you want to forward the call to, in this case extension "1001", in the entry field **[Forwarding destinations].**
4) Click the **[Save]** button to save changes.

With this setting, when extension 1001 receives calls, they will be forwarded to extension 1001.



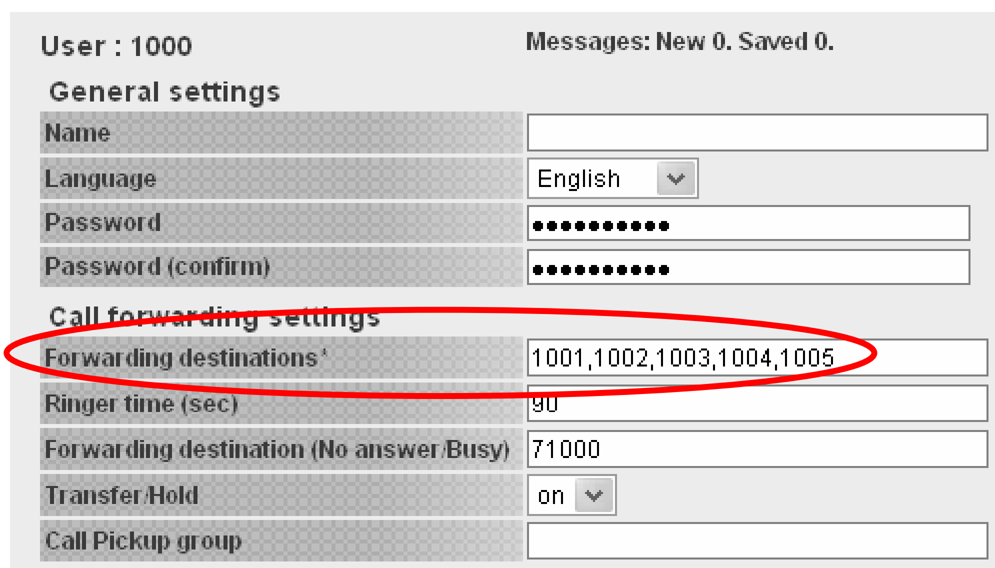
6.3. Ring Group Setup



In this example, Ring Group is set up for extension 1000. When calls are received by any extension in the group, all extensions (1001-1005) will ring simultaneously.

OnDO PBX Admintool > User Settings > Edit

1) Choose the **[User Setting]** menu.
2) To create a Ring Group extension, enter *1000* in the entry field, click the **[Create a New User]** button. User 1000 will be added.
3) Click the edit button for user 1000 to open the [User Setting] window.
4) Enter 1001, 1002, 1003, 1004, 1005 in the [Forwarding destination] field.
5) Click the [Save] button to save changes.



6.4. Setting a Call Pickup Group

Call Pickup is a function that allows users to answer incoming calls from any OnDO PBX phone by dialing a pre-set number. When an OnDO PBX phone rings, dial *** + extension** to answer the call. For example, when extension 1001 rings, dialing *** + 1001** will enable you to answer the call from any OnDO PBX user's phone.

Here are some other special ways of using the Call Pickup feature:

◆ *Using Call Pickup feature to answer calls that are directed to a Ring Group extension*

Calls directed toward a Ring Group extension can be answered from any extension using Call Pickup. (For details for how to set up Ring Group, please refer to section 6.3 Ring Group Setup)

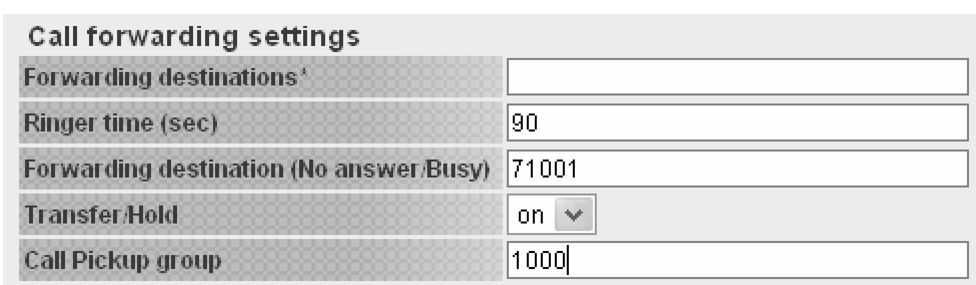
Extension: 1000

Ring Group	1001,1002,1003,1004,1005

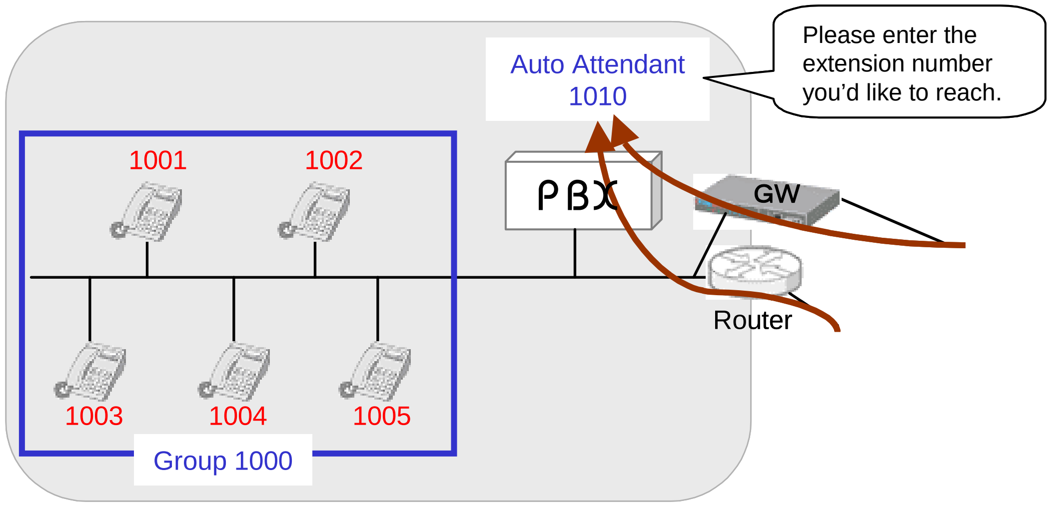
Dialing ***1000** (Ring Group extension number) enables one to pick up any call that is directed to any of the extensions in the Ring Group. Incoming calls can still be answered by dialing * + extension for each number; however, using the group extension number does not require you to remember each extension in the group. This feature works even when a call comes directly to a single extension (e.g. 1003), or comes through the Auto Attendant.

◆ *One touch Call Pickup for specified Call Pickup Group*

OnDO PBX lets you set up your Call Pickup Group number, so that you only need to dial ***(Star)** to pickup the incoming calls directed to any of the extensions that belong to your custom group number. You may specify a Call Pickup Group at **[Forwarding destinations settings]** in OnDO PBX Admintool (shown below).



6.5. Auto Attendant Setup



The Auto Attendant will answer all calls that come in to 1010.

OnDO PBX Admintool > User Settings > Edit

1) Choose the **[User Setting]** menu.
2) To create an Auto Attendant extension, enter 1010 in the entry field, then click the [Create a New User] button. User 1010 will be added.
3) Click the edit button for user 1010, to open the edit window for the user 1010.
4) Enter a number for 1010's Auto Attendant* in the [Forwarding destinations] entry field. The Auto Attendant number should have a prefix. If you are using a default IVR prefix, the number set is 61010.
5) Select [Auto Attendant] from the list of [IVR] in the [Administrative setting]. (Depending on your needs, the items inside [Auto Attendant setting] can be updated and an audio file for Auto Attendant greeting can be uploaded. For more detailed information, refer to Setting Items 2. User Setting.) Please refer to the section 12.2 Prefix for more information about how prefix setting works with OnDO PBX.
6) Click the [Save] button to save changes.

7. Connecting to outside lines

It may not be quite accurate to use the terms “extension” and “outside line” for IP telephony, as VoIP does not make a clear distinction between inside and outside lines as traditional telephony system would. Here “extension” indicates a self-defined, OnDO PBX number such as office extensions, while “outside number” or “outside line” refers to a number assigned by a third party. Analog lines connected through PSTN Gateways, or lines connected by VoIP service providers are examples of “outside lines”.

7.1. Connecting with PSTN Gateways

7.1.1. PSTN Gateway Settings

IP address	192.168.0.111
Recipient Number	111
SIP Proxy address	OnDO SIP Server’s IP address

✓ *For details of how to set up your PSTN Gateways, please refer to your Gateway’s manuals.*

There are two types of PSTN Gateways available. The first enables one-step calls by entering (dialing) a SIP URI (sip:<the final destination number>@192.168.0.111). The other enables calls by two-step dialing, which require first dialing a SIP URI format (sip:<the number specifying the gateway’s port>@192.168.0.111>, then dialing the destination number using dial buttons after a dial tone.

This example uses the type of PSTN Gateway that only requires one-step dialing using a SIP URI.

At the PSTN Gateway setting, incoming calls through the PSTN Gateway will be directed to extension number 111. (sip:111@<OnDO SIP Server’s IP address>)[iii]

If the PSTN Gateway has Caller ID detection features, Caller ID information can be carried through to your phones or displays. In order for Caller ID information to be displayed, the From header must have “sip:<caller id information>@<OnDO SIP Server’s IP address>”. Your phones or SIP UAs also must have been able to display Caller ID information.

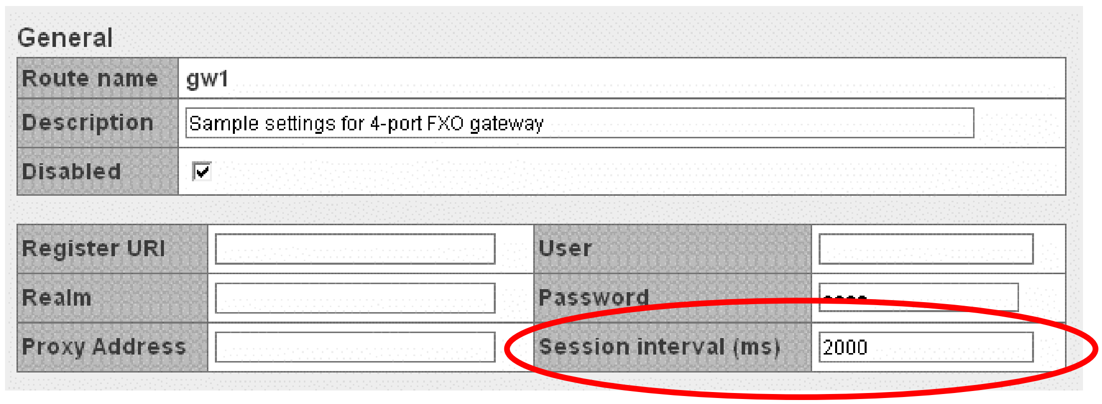
[iii] You may specify multiple numbers for receiving incoming calls.

7.1.2. Setting ARS for PSTN Gateway

OnDO PBX Admintool > ARS > Edit

- *General*

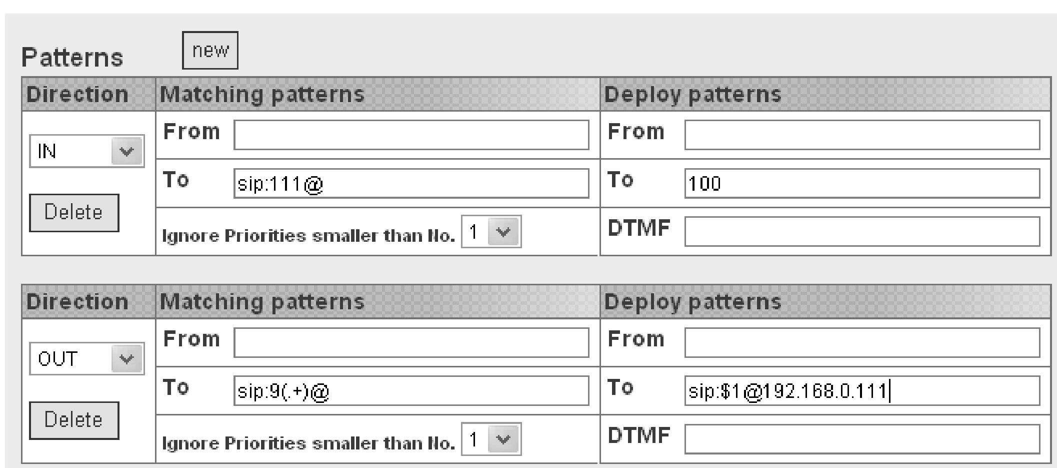
  Changing these settings is not required here unless you use authorization for connecting to PSTN Gateway. Many PSTN Gateways have a short interval between sessions. With this example we have added 2000 milliseconds for session intervals.



- *Patterns*

  Here you set up the patterns when a call is initiated (OUT) and received (IN) through PSTN Gateways. In the pattern (IN), leave "From" entry blank to carry over Caller ID information.

  In the example below, when a call comes through the PSTN Gateway, extension 111 is set to ring. On the other hand, outgoing calls that start with 9 will be directed through the PSTN Gateway.



- *Priorities*

  It is useful to set priorities when there are multiple options for making calls, such as when you have multiple PSTN Gateways or when you subscribe to multiple VoIP service providers. Maximum session number can be set in the **[Max Sessions]** field. Priority can be defined in the **[Priorities]** field. Lower numbers hold higher priorities.

7.2. Connecting with VoIP service providers' SIP Server

7.2.1. Account information for third party SIP Server

Acquiring the information shown below is necessary to connect with a third party SIP Server.

Phone number	6504106636
SIP Server IP address	sample proxy
User ID	6504106636
Password	6636

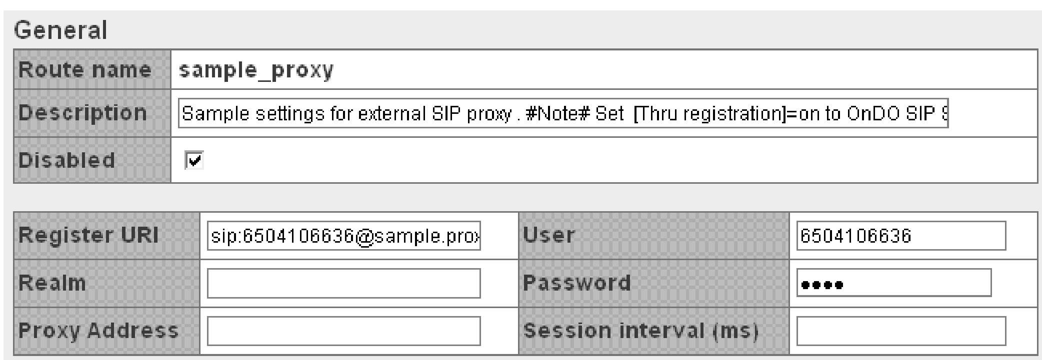
✓ *Depending upon the provider there may be restrictions for connecting to services, such as available information or equipment used to connect. Please contact your VoIP service provider for more details.*

✓ *Please note that we do not guarantee connection with third party products.*

7.2.2. Setting ARS for VoIP service providers

OnDO PBX Admintool > ARS > Edit

Input third party account information in the **[General]** section of **[ARS]**. "Register URI" must be filled in using the SIP URI format (sip:user@domain.com)."Proxy Address" can be omitted if the address is the same as the one in "Register URI". "Realm" can be left blank if the realm setting is not necessary.



◆ *Patterns*

Use these settings to define patterns for when a call is initiated and received through third party SIP Server.

In the example pattern [IN] below, extension 100 is set to ring when a call comes through the third party SIP Server. Leaving the "From" field blank carries over the Caller ID information.

Settings for the "OUT" pattern define patterns for converting SIP URI to match your VoIP provider's header format requirements. In the example below, dialed numbers beginning with 0 will be directed through the VoIP service provider. To ensure that the recipient's caller ID display will function, the From header will be changed according to the rules of the provider.

✓ *Some VoIP service providers restrict the connection when FROM or TO header information is different from their own header format.*

◆ *Priorities*

It is useful to set priorities when there are multiple options for making calls, such as when you have multiple PSTN Gateways or when you subscribe to multiple VoIP service providers. Maximum session number (such as port numbers of Gateways or subscribed line numbers of VoIP services) can be set in the **[Max Sessions]** field. Setting "-1" specifies an unlimited number of sessions. Priority can be defined in the **[Priorities]** field. Lower numbers hold the higher priorities

8. Setup Items

8.1. Option menu

The following list displays the settings you can set up under the **[Option]** menu. This menu is only available to system administrators.

8.1.1. General setting

Name	Default value	Description
Start up	auto	Auto: OnDO PBX starts up automatically with Tomcat (OnDO PBX HTTP Service). Manual - Start up manually. Options: Auto/Manual

8.1.2. SIP setting

Name	Default value	Description
SIP Proxy address	localhost	Defines the IP Address or Hostname of the SIP Server the OnDO PBX uses as a SIP Proxy.
Register expires	3600	Interval for sending REGISTER requests to third party SIP Servers using ARS feature

8.1.3. Phone number setting

Name	Default value	Description
IVR prefix	6	By using this prefix before an extension, a caller can reach that user's IVR
Voicemail prefix	7	Using this prefix before an extension allows a caller to reach that user's voicemail inbox directly to leave a message.
Voicemail review/ Setting prefix	8	Using this prefix before an extension allows access to that extension's voicemail inbox to check messages. The caller will be asked for a password before access is granted.
Call Pickup prefix	*	Dialing this prefix allows users to answer incoming calls directed to other call pickup group users. Prefix + ringing extension number Default = * + ringing extension number
Park number (min)	60	The minimum assigned number for retrieving parked calls. (The number is assigned randomly.)
Park number (max)	89	The maximum assigned number for retrieving parked calls. (The number is assigned randomly.)

8.1.4. PBX system setting

Name	Default value	Description
Port number	15060	The port number that OnDO PBX will use. (This port number cannot be modified in the SmallOffice Edition)
Max simultaneous sessions	4	The maximum number of simultaneous sessions that OnDO PBX can handle. (This number cannot be modified in the SmallOffice Edition)
Max number of user agents	10	The maximum number of user agents (SIP UAs) that PBX can handle. (This number cannot be modified in the SmallOffice Edition)
Minimum Port	11000	Minimum port number the RTP Protocol uses for sending voice data.
Maximum Port	11999	Maximum port number the RTP uses for sending voice data.
RTP relay	off	Specify RTP packets are handled by Media server or not. This setting will be in effect when there are no RTP relay specifications made in ARS or User settings.
INVITE Timeout (ms)	60000(ms)	Timeout value for awaiting a response to an INVITE request.
Ringing Timeout (ms)	120000(ms)	Timeout value for awaiting an answer from the dialed party after ringing starts.
Talking Timeout (ms)	0(ms)	The maximum length of time a call can last. Value 0 signifies infinite.
BYE Timeout (ms)	60000(ms)	Timeout value a BYE request waits for an answer.
RTP Session Timeout (ms)	600000(ms)	Timeout value for awaiting the next RTP packet after the system received the last one.

✓ *ms = 0.001 second*

8.1.5. Media Server system settings

Name	Default value	Description
Port number	25060	The port number that Media Server system uses. (This port number cannot be modified in the SmallOffice Edition.)
Max Inbound session limit	4	The maximum number of simultaneous sessions to the voicemail system and IVR that can be handled by OnDO PBX. (This port number cannot be modified in the SmallOffice Edition.)
Max messages	50	Maximum number of saved voicemail messages and any recorded file for each user's voicemail inbox.
Message recording length (sec)	600(sec)	Maximum length of recording time for a voicemail message.
Days to keep unsaved messages	30	The number of days before unsaved messages is deleted automatically from each user's voicemail inbox.
Minimum Port	12000	Minimum port number the RTP uses for sending voice data.
Maximum Port	12999	Maximum port number the RTP uses for sending voice data.
Invite Timeout (ms)	60000(ms)	Timeout value for awaiting the response to an INVITE request.
Ringing Timeout (ms)	120000(ms)	Timeout value for awaiting an answer from the dialed party after ringing starts.
Talking Timeout (ms)	0(ms)	The maximum length of time the call can last. Value 0 signifies an infinite value.

- ✓ *ms = 0.001 second*

8.1.6. Email setting

Name	Default value	Description
SMTP server	Blank	The SMTP Server Address for sending email notifications when the user receives a new voicemail message.
POP3 server	Blank	The address of the POP3 server.
User	Blank	Account user name for the above SMTP server.
Password	Blank	Password corresponding to the account user name above.
Password (confirm)	Blank	Input field for confirming the above password.
Email address (from)	Blank	Email notifications sender’s address.

8.2. Automatic Route Selection (ARS)

OnDO PBX automatically selects the optimum call route from preset routing options. This feature can be used for Least Cost Routing, traffic management and load balancing of PSTN gateways or PBX.

To add a new route:

1) Choose the **[ARS]** menu from.
2) Type the name of a rule in the input field at the bottom of the screen
3) Click **[Create a new route]** to add the route.
4) To edit the rule's settings, click the **[edit]** button.

OnDO PBX Admintool > ARS > Edit

8.2.1. General

Name	Default value	Description
Route name	Blank	The name for the route
Description	Blank	The description for the route
Disabled	OFF	Disable this setting

Name	Default value	Description
Register URI	Blank	SIP URI that is used to register OnDO PBX with a registrar server
Realm	Blank	Realm that is used for authentication. This field is optional.
Proxy Address	Blank	IP address of the registrar server. This field is optional when the proxy address is the same as the address set in the Register URI field
User	Blank	User ID for authentication account. This field is optional field when authentication is not being used.
Password	Blank	Password for authentication account. This field is optional when authentication is not being used.
Session interval (ms)	Blank	Set interval period between sessions for some VoIP FXO Gateways that require pausing between sessions.

8.2.2. Pattern

Name	Default value	Description
Direction	Disable	Choose direction for the patterns you wish to specify. IN – for incoming calls to OnDO PBX OUT – for outgoing calls from OnDO PBX
Matching patterns(FROM/TO)	Blank	Specify address patterns for FROM and TO header using regular expressions. When the field is left blank, all calls will be considered matching calls.
Deploy patters (FROM/TO/DTMF/ RTP relay)	Blank	Specify replace patterns for FROM and TO header using regular expressions. For when DTMF needs to be issued after calling gateway (2 step calling), you can specify the DTMF string using some part of [TO] Matching Pattern.

8.2.3. Priorities

Name	Default value	Description
Direction	IN/OUT	Specify direction of the priority.
Max Sessions (-1=unlimited)	-1	Specify the number of sessions (including RINGING and BYE sessions) that are allocated to the priority.
Priorities	100	Lower numbers hold a higher priority.

8.3. Call Log

By specifying a date, you can view call log information for that date. By default, call log information is displayed in html on the browser, but by clicking the **[csv]** button, you can download a log file to your local machine in .CSV format.

Individual Call logs for each user are available through **[User Settings] > [Edit] > [Call log]**.

8.4. User Setting

OnDO PBX Admintool > User Settings > Edit

Click the menu **[User Setting]** on the menu bar, select a user, and click the edit button for that user to set the following items:

8.4.1. Messages

Name	Default value	Description
Messages	(Not visible)	If there are voicemail messages for this user, this item will appear on the screen. The type of message (new/saved), date and time the message was received, and file size will be displayed.
Download		To download the voicemail message as an audio file (WAV format), select the desired message from the pull-down list and click **[Download]** button. The file will be downloaded to your PC.
Delete		To delete a voicemail message, select the desired message from the pull-down list and click **[Delete]** button. The message will be deleted from the voicemail inbox.

8.4.2. General setting

Name	Default value	Description
Name	Blank	The description of this user. This field is optional.
Language	Ask your system Administrator[iv]	Selected Language will be used by the OnDO PBX Admintool and as the default voicemail guidance prompt. Options: English/Japanese
Password	Ask your system Administrator[v]	Password for the OnDO PBX Admintool login and accessing the user's voicemail inbox.
Password (confirm)	Ask your system Administrator	Reenter password for confirmation.

[iv] These options will be set as the Administrator who create this user
[v] These options will be set as the Administrator who create this user

8.4.3. Call forwarding settings

[Basic] is selected under Type of Call Forwarding (Administrator)

Name	Default value	Description
Forwarding destinations*	Blank	Enter phone number(s) or SIP-URI to directly forward all calls that are received at this extension number. Multiple forwarding destinations can be specified by separating entries with a comma (,) delimiter.
Ringer time (sec)	90	The length of time that the user's phone will ring when it receives a call. After the length of time set here, the call will be transferred to the destination that is specified in [Forwarding destination (No answer/Busy).] If no destination is set at [Forwarding destination (No answer/Busy)], the call will be terminated.
Forwarding destination (No answer/Busy)	Voicemail Prefix[vi] + extension #	Phone number or SIP-URI to which the call will be forwarded when Ringer timeout has occurred.
Transfer / Hold	on	Enable/disable this user to use call transfer/hold features. Options: on/off
Call Pickup group	Blank	Enable one touch Call Pickup for the preset group extensions by assigning the group number.

*** This category allows multiple entries. (divided with commas)**

[vi] This value is set in the [Option] menu

8.4.4. Call forwarding settings

[Round Robin/Top-down] is selected under Type of Call Forwarding

Name	Default value	Description
Forwarding destinations*	Blank	Enter phone number(s) or SIP-URI to directly forward all calls that are received at this extension number. Multiple forwarding destinations can be specified by separating numbers with a comma (,) delimiter.
Ringer time (sec)*	20	Ringing timeout for waiting for the recipient to answer. After the length of time set here, the call will be transferred to the destination that is specified in the [Forwarding destination (No answer/Busy)] field. If no destination is set at [Forwarding destination (No answer/Busy)], the call will be terminated.
Waiting time in the queue (sec)	0	The length of time for queued calls will be held on hold till forwarded to the user destination set at [Forwarding destination (No answer/Busy)].
Max number of calls in the queue	10	The maximum number of calls in the queue.
Call interval (msec)	3000	The interval period for calls in queue to ring a client that end the call session.
Single attempt	no	Enable/disable to retry calls when an initial try has unanswered. When this setting is enabled, the call will be transferred to the destination set at [Forwarding destination (No answer/Busy)].
Forwarding destination (No answer/Busy)	7 + extension #	Phone number or SIP-URI to which the call will be forwarded when Ringer timeout has occurred.
Mode	Round-robin	There are two modes for call forwarding: Round Robin: Calls will be distributed from the top of the list. When a call is received, it is forwarded to the extension following the last extension to have received a call. Top-down: Calls will always be distributed in the order listed in the field.
Transfer / Hold	on	Enable/disable this user to use call transfer/hold features. Options: on/off

Call Pickup group	Blank	Enables one touch Call Pickup by assigning the user to a preset Call Pickup group.

8.4.5. Call forwarding settings

[Schedule] is selected under Type of Call Forwarding by the administrator

For more detailed information on OnDO PBX's schedule function, please see section 14.5

Name	Default value	Description
Forwarding destinations*	Blank	Destination phone number(s) or SIP URI(s) to forward all calls that are received by this extension. Multiple entries can be specified by using a comma (,) delimiter. Calls will be forwarded to the appropriate destination based on conditions defined in the schedule.
Ringer time (sec)	90	The length of time that the callee's phone will ring. The appropriate Ringer time will be applied as the condition specified in **[Schedule]** or [**Applies to (Caller numbers)]** or **[Not Applies to (Caller numbers)]**.
Forwarding destination (No answer/Busy)	7 + extension #	Phone number or SIP-URI to which the call will be forwarded when Ringer timeout has occurred.
Schedule	Blank	Specify schedule information by which to forward incoming calls.
Applies to (Caller numbers)*	Blank	Enter applicable Caller ID information for the schedule settings. Multiple Caller IDs can be specified by separating entries with a comma (,) delimiter. 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.
Not Applies to (Caller numbers)*	Blank	Enter Caller ID information to be exempted from this setting. Multiple Caller ID numbers can be specified by separating entries with a comma (,) delimiter. By adding a * (wildcard) after a number, you can specify all numbers that begin with that number.
Transfer / Hold	on	Grants the user permission to use transfer and hold functions. Options: on/off
Call Pickup group	Blank	Enables one touch Call Pickup by assigning the user to a preset Call Pickup group.

8.4.6. Voicemail setting

Name	Default value	Description
Greeting message	Default system greeting	Select the greeting message for this user's voicemail. Options: Default system greeting /Personal greeting (user created)/Alternative greeting (user created).
Message forwarding*	Empty	The extension number(s) to which received voicemail messages will be forwarded. Multiple numbers can be specified using a comma (,) delimiter.
Email address*	Empty	The e-mail address to which the notification of the arrival of voicemail messages is sent. Multiple addresses can be specified using a comma (,) delimiter.
Email notification	off	Enables/Disables email notification. Options: on/off
Attach WAV file to email	off	Enables attachment of voicemail messages in wav format to email notifications.

*** This category allows multiple entries. (divided with commas)**

8.4.7. Administrative setting - This menu is only available to the system administrators

Name	Default value	Description
User Type	User	Specifies the type of user. Options: User/Administrator
Type of Call Forwarding	Basic	Specifies the type of Call Forwarding. Options: Basic, Round robin/Top-down/, Schedule
IVR	None	Setting for the usage of the Interactive Voice Response (IVR) system. The following options are available:**None**: No IVR service **Auto Attendant**: Calls will be answered by the Auto Attendant **Setup**: Enable mailbox management over IVR system. Note: For more details please refer to [set up menu] in voicemail navigation map

8.4.8. PBX settings - This menu is only available to the system administrators

Name	Default value	Description
RTP relay	default	Specify RTP packets are handled by Media server or not. “on” – This user’s RTP packets will be relayed by Media server. “default” – RTP relay setting in ARS will be valid.
Max sessions	unlimited	Specify the maximum received session numbers for the user.

8.4.9. Auto Attendant Setting - This menu is only available to the system administrators

Name	Default value	Description
Max input digits	4	Maximum number of input digits accepted by the Auto Attendant.
Max retry count	5	Maximum number of retries when an input error has occurred. After retries of the number of times set here, the call will be terminated.
Ring timeout (sec)	30(sec)	The length of time that a user’s phone will ring when a call is received via Auto Attendant.
Default operator	Empty	Default destination (phone number or SIP-URI) for an incoming call that hasn’t specified a call recipient.
Speed dial*	Empty	Set up “Speed dial” for Auto Attendant. Example: 0=0001,5=sip:sales@brekeke.com. In this case, instead of dialing 0001 to reach the user 0001, a caller can dial 0. Multiple items can be specified using a comma (,) as a delimiter.
Transfer to unregistered users	disable	Enables/disables call transfers to an unregistered user. Options: disable/enable

*** This category allows multiple entries. (divided with commas)**

8.4.10. Message files: Download/Upload

Button Name	Description
Download	To download a file, select a file type from the pull-down list, and click **[Download]**. See below for the types of message files.
Upload	To upload a file, select a file type from the pull-down list, and click the **[Browse]** button. Select the file you want to upload and click **[Upload]**. The upload will then start.
Delete	To delete a file, select a file type from the pull-down list and click **[Delete]**.

Types of message files

File Name	Description
Voicemail personal greeting	Personal Voicemail greeting message the user has created.
Voicemail alternative greeting	Another voicemail greeting message the user has created.
Name	A message file that contains the user’s name. (Ex., when you record a message for another user’s voicemail, the recipient will hear “There is a message from ‘name’”.)
Music on hold	An audio file that contains music/sound that will be used for music on hold.
Auto Attendant greeting message	Greeting message that is used for the Auto Attendant.
Auto Attendant retry message	A message to prompt the caller to reenter when an input error occurred.

✓ *The number in the parentheses after the name of above field shows the file size. When no file exists for the type of message file, (none) is displayed.*

8.5. Notes for sound file

Uploaded sound files must be formatted as below.

Sample rate	8000kHz
Bit-Depth	8bit
Channels	Mono

You may use sound recording applications, such as Windows Microsoft Sound Recorder. We recommend you to adjust the pause and sound level to suit your needs and requirements.

9. Uninstall (Windows)

Choose **[Start]>[Program]>[Brekeke]>[OnDO]>[OnDO PBX 1.1]>[Uninstall OnDO PBX]**. The uninstaller will uninstall OnDO PBX automatically.

- ✓ *If the uninstaller fails to delete the folder (C:\Program Files\Brekeke\OnDO\pbx) you will need to restart the PC and delete the folder manually.*

10. Uninstall (Red Hat Linux)

Delete the file “pbx.war” and the folder “pbx” in the directory \webapps which is located under the install directory of Tomcat.

11. SIP Server

11.1. The relationship between OnDO PBX and SIP Server

A SIP Proxy Server provides call session management in a VoIP network and processes SIP requests and responses. OnDO PBX receives call sessions from a SIP Server, and directs call sessions to the appropriate destination according to the users' settings; such as Forwarding destinations. When the Forwarding destinations setting is left blank, the call will not be transferred and be distributed to an original destination. The call sessions initiated from OnDO PBX will be directed to a destination SIP UA as defined in Dial Plan of SIP Server.

The PBX features of OnDO PBX depend greatly on how its SIP Server routes call sessions. Thus you will need to customize the SIP Server to work with OnDO PBX accordingly. The OnDO SIP Server (bundled with OnDO PBX) is already set to work with an OnDO PBX that is installed on the same computer. Since OnDO PBX is independent from SIP Server, it is possible to have another SIP Server in place of OnDO SIP Server; however, the SIP Server must be able to route call sessions as OnDO SIP Server's Dial Plan.

11.2. Setting for OnDO SIP Server

11.2.1. Installing OnDO PBX and OnDO SIP Server on the same machine

OnDO SIP Server that comes with OnDO PBX has the following default settings in the Dial Plan. These settings are based on the assumption that OnDO PBX will work with OnDO SIP Server on the same machine.

to Media Server

Matching Patterns	Deploy Patterns
$port=15062 $request=^INVITE $localhost=true to=(^.*sip:[678].{0\,5}@.*\$|^.*sip:media.*@.*\$)	$target=localhost:25060 to=%1 $auth=false

The calls will be routed to Media Server (Voicemail or Auto Attendant) when the following two conditions match:

1. The dialed number starts with 6, 7. or 8. 2. The dialed number is less than 6 digits long (including prefix)
2. The To header field will not be changed. Specify using $target. Authentication is diabled.

from PBX 1

Matching Patterns	Deploy Patterns
$registered=true $port=15062 $request=^INVITE $localhost=true	$auth=false

If the session coming from PBX has been directed to the OnDO SIP Server's registered user, the call will be routed to the Registered user. Authentication is disabled.

from PBX 2

Matching Patterns	Deploy Patterns
$request=^INVITE $localhost=true $port=15062 $outbound=true	$auth=false

If the session coming from PBX has a destination outside the host IP address, the call will be routed to the address.

from PBX 3

Matching Patterns	Deploy Patterns
$port=15062 $request=^INVITE $localhost=true	$action=404

If the above rules don’t apply for the session coming from PBX, an error 404 (Not Found) will be returned

to PBX

Matching Patterns	Deploy Patterns
$request=^INVITE to="(.+)	$target=localhost:15060 To=%1

The sessions that are not from PBX will be routed to PBX. To header field won’t be changed. Specified using $target.

11.2.2. Installing OnDO PBX and OnDO SIP Server on the separate machines

You might want to run OnDO PBX and OnDO SIP Server on separate machines, or you might want to have multiple OnDO PBX installations with one OnDO SIP Server to act as the Proxy Server. For these cases, you will need to set the Dial Plan appropriately to suit your situation. For details on Dial Plan configuration, please refer to “OnDO SIP Server manual” and “OnDO SIP Server tutorial –Dial Plan”. You can obtain related documents at the Brekeke Website: http://www.brekeke.com/en/download/download_en.html.

12. Process

Two processes will be initiated when OnDO PBX starts: the main process and the process for voicemail / IVR features. You will be able to change the process count number in the **[Option]** menu. (This feature is limited to OnDO PBX Standard Edition only. For the SmallOffice Edition, each process is limited to one.) Depending on the specifications of your hardware, operating system, and type of Java VM, increasing the process count might improve the performance of OnDO PBX. When you increase the number of process counts, it is also necessary to make modifications to the SIP Server routing (Dial Plan) to distribute SIP call sessions to the newly created processes appropriately.

12.1. Main Process

The OnDO PBX main process is the actual player who exchanges SIP sessions between SIP UAs. The main process will receive all SIP call sessions that are dispatched from the SIP UAs that are registered on the OnDO SIP Server. SIP UAs that receive incoming call sessions be called by OnDO PBX.

12.2. Media Server Process

This process handles RTP packet (sound data). It also enables PBX features, such as voicemail, Auto Attendant, and Setup. This process is separate from the OnDO PBX main process; however, with the current version of OnDO PBX you may not run this process on a separate PC from the main process.

12.3. About RTP relay

According to your environment and requirements, the RTP relay settings need to be specified. Below are some examples when RTP relay setting is necessary:

- When using SIP UA which does not support DTMF-via-INFO method, and wish to use OnDO PBX features such as Call Forwarding, Call Park, etc.
- Using SIP devices (SIP UA, SIP proxy server, etc) that do not support changing RTP sender information when request is sent via re-INVITE message.

RTP relay settings can be set under several locations in OnDO PBX: Option, ARS settings, and User Setting. Setting under [User Setting] has the highest priority. When the default is set at [User Setting], setting under [ARS settings] will be in effect. When both settings are set at default, the setting under [Option] will be valid.

RTP relay will be turned off only when both end points are set RTP relay off. When one of the end point require RTP relay, the media server will handle the RTP packets.

By decreasing the amount of RTP packet handled by media server, the maximum number of concurrent session will be increased. When there are no change in voice quality, turning RTP relay on at [Option] settings will decrease the chance of miscellaneous troubles.

Even when the RTP relay setting is set to “off”, some PBX features’ (music on hold and voicemail) RTP packets are handled by media server. Thus depending upon the type of usage and environment, the maximum number of concurrent session can by varied upon each setting.

13. Number System

A well-designed number system is indispensable for a user-friendly PBX system.

13.1. Definition of OnDO PBX User

OnDO PBX recognizes each extension number as an OnDO PBX User ID. An OnDO PBX user ID is the same as the SIP URI user name (e.g. In the SIP URI “sip:001@brekeke.com”, 001 is considered the OnDO PBX User ID). PBX users can use PBX features, such as call transfer, call hold, call forwarding, and voicemail.

13.2. Prefix

To provide easy access to users’ PBX features such as voicemail and Auto Attendant, OnDO PBX uses a prefix system. By adding the appropriate prefix to an extension number, users can directly access voicemail or Auto Attendant. This feature allows users to reach their own voicemail boxes without remembering a special number.

Prefixes are set in **[Phone number setting]** under the **[option]** menu. Assigning prefix numbers can only be done by the administrator. We recommend that you not change prefix settings once established, because changes to the prefix settings would require user to change his/her forwarding settings.

The table below shows the default prefix settings:

IVR prefix	6
Voicemail prefix	7
Voicemail review / Setting prefix	8
Call Pickup prefix	*

When you need to change the prefix (number or alphabet) value, you will need to change the OnDO SIP Server Dial Plan appropriately. The default OnDO SIP Server Dial plan is set with default prefix settings, and calls will be routed accordingly. **Thus with OnDO PBX/OnDO SIP Server default settings, you cannot create extension numbers starting with 6, 7, or 8.**

<u>Example:</u>
Call received for extension 7001 will be routed to user 001's voicemail inbox. Even if user 001 doesn't exist in the system, the call will not be routed to user 7001.

When you need to create user extension numbers that start with the same value as a reserved prefix, you will need to specify the appropriate routing in the OnDO SIP Server Dial plan. One way of achieving this is by routing by the number of digits. Ex.: All extension numbers are set to a 3-digit length. In your dial plan, you specify that all 3-digit length extension number should route to the SIP UAs even when the number starts with the same value as a prefix. If the dialed number is 4 or more digits in length and starts with a prefix value, then the call will be routed to the Media Server process.

<u>Example:</u>

- Extension (User ID 001
- IVR prefix 6
- Voicemail prefix 7
- Voicemail retrieval / Setting prefix 8

13.2.1. Using IVR prefix

When a caller dials "6001", s/he will reach User 001's Interactive Voice System. OnDO PBX offers voicemail management over IVR. For more details, please refer to **[set up menu]** in the Voicemail Navigation Map.

13.2.2. Using Voicemail prefix

Callers can dial "7001" to directly reach User 001's voicemail inbox. This feature is useful for leaving a message without disturbing the recipient party, since the call will be directly connected to the user’s voicemail inbox. The forwarding destination for unanswered calls is set in the **[Call forwarding (no answer)]** in the **[User Setting]** menu.

13.2.3. Voicemail retrieval / Setting prefix

User 001 can dial 8001 to access his/her voicemail inbox and modify his/her voicemail settings. The user will be prompted for a password before being allowed to access messages and settings.

14. Call forwarding / Call forwarding (no answer)

14.1. Call forwarding

Each user can set up call forwarding individually. When a call is received at the user’s extension, the call will be forwarded to the destination that was set in the **[Forwarding destinations]** field in the **[User Setting]** menu.

By default, the **[Forwarding destinations]** field is left blank and all incoming calls will be directed to the user’s extension. If there are destinations specified in the field, only those extensions will receive calls.

If multiple destinations are set, incoming calls will be forwarded to all destinations (Ring Group). Each extension’s phone will ring at the same time, and the first person to answer will be connected with the caller.

14.1.1. Example 1: Forwarding calls to a single extension

Extension: 001

Forwarding destinations*	002

With the setting above, every call that comes into extension 001 will be transferred to extension 002. Extension 001 will not receive any calls that are dialed for extension 001.

14.1.2. Example 2: Forwarding calls to the multiple extensions (Ring Group)

Extension: 001

Forwarding destinations*	001, 002

With the setting above, every call that comes into extension 001 will be transferred to extensions

001 and 002. If there is no forwarding set at the extension 002, both extension 001 and 002 will receive calls that comes into the extension 001. The first person that answers the phone will be connected with the caller.

14.1.3. Example 3: Forwarding calls to the multiple group extensions (Multiple - Ring Group)

Extension: 001

Forwarding destinations*	100, 200

Extension: 100

Forwarding destinations*	101, 102, 103

Extension: 200

Forwarding destinations*	201, 202, 203

With this setting, incoming calls to extension 001 will be forwarded to extensions 100 and 200. Extensions 100 and 200 in turn forward calls to extensions 101,102, 103, 201, 202, and 203. If the users on these extensions don't have call forwarding set up, these 6 extensions will ring. The first person that picks up the phone will be connected with the caller.

14.1.4. Example 4: Enter SIP URI in the Forwarding calls field

Extension: 001

Forwarding destinations*	sip:6636@brekeke.com

You can specify SIP URI in the **[Forwarding destinations]** field under the **[user setting]** menu.

14.1.5. Example 5: Ignore destination number's call forward settings. Forward call only to the specified extension here

Extension: 001

Forwarding destinations*	%002

With this setting, the calls directed to extension 001 will be forwarded to extension 002. All of the call forwarding settings at extension 002 will be ignored, and the settings at 001 will be applied. Also, forwarded calls will be recorded on extension 001's call log.

14.1.6. Example 6: Ignore

Extension: 001

Adding "!" before an extension instructs the system to ignore that extension. This setting is useful when a certain extension needs to be excluded from the setting temporally. Note that the call will be disconnected when all of the destination extensions have "!"

14.1.7. Example 7: Disconnection

Extension: 001

Forwarding destinations*	!

With this setting, all incoming calls to extension 001 will be disconnected.

14.2. Call forwarding (no answer/busy)

The cases in which this forward setting is used are:

- *The user’s phone is not connected (when there is no response from the phone)*
- *The UA send responses other than 200OK response (e.g. busy)*
- *After the ringer timeout occurs*

Each user can set up forwarding for unanswered calls. By default, unanswered calls are forwarded to the user’s voicemail inbox. (For the extension 001, “7001” is set for the **[Forwarding destination (no answer/busy)]** field.

Unlike regular call forwarding settings, you can only specify a single destination here. If you need to set up multiple destinations here, you can use the method described in [13.1.3 Example 3: Forwarding calls to the multiple group extensions (Multiple - Ring Group)].

14.2.1. Example 1: Forwarding calls to voicemail

Extension: 001

Ringer time (sec)	15

Incoming calls to extension 001 will be forwarded to the user’s voicemail inbox (extension 7001) if the call remains unanswered after ringing for 15 seconds.

14.2.2. Example 2: Forwarding unanswered calls to another extension

Extension: 001

Ringer time (sec)	15
Forwarding destination (no answer)	002

With this setting, the unanswered call will be forwarded to extension 002. You may specify a SIP URI instead of extension number.

14.3. More advanced forwarding setting

By combining Call forwarding, Call forwarding (no answer), and Ringer time, you may create more advanced call forwarding settings.

Here are a few rules that are useful to remember;

- *When Call forwarding is set for the extension, the Ringer time for the call depends on the setting on the extension that calls are forwarded to.*
- *If the forwarded call isn't answered by the extension(s) that the call was forwarded to, the call will be forwarded to the destination that is set under the original call recipient's [Forwarding destination (no answer/busy)].*
- *Only a single destination can be set in the [Forwarding destination (no answer/busy)] field*

By specifying a different extension number in the **[Forwarding destination (no answer/busy)]** in the settings of the forwarded extension number, you can create another forwarding schema.

14.3.1. Example 1: Ringer time & Forwarding destination (no answer/busy)

Extension: 001

Forwarding destinations*	101, 102
Ringer time (sec)	20
Forwarding destination (no answer)	7001

Extension: 101

Forwarding destinations*	
Ringer time (sec)	15
Forwarding destination (no answer)	7101

Extension: 102

Forwarding destinations*	
Ringer time (sec)	25
Forwarding destination (no answer/busy)	7102

1) All incoming calls to the extension 001 → forwarded to both extensions 101 and 102
2) Extension 101 rings for 15 seconds and Extension 102 rings for 25 seconds →no answer
3) The call will be forwarded to extension 7001

User 001's settings, Ringer time and Call forwarding (no answer), are treated as highest priority even when the user 101 and 102 have set up call forwarding (no answer) on all of his/her calls.

With the settings above, incoming calls to extension 001 will be transferred to both extensions 101 and 102. If the call is not answered by extension 101 or 102, it will be forwarded to the destination specified in User 001's **[Call forwarding (no answer)]** field. The forwarded call will ignore **[Call forwarding (no answer)]** settings for extensions 101 and 102.

Ringer time for extension 001 is set for 20 seconds. Even after User 101's ringer stops, extension 102 will continue ringing until 20 seconds have passed. Then the call will be forwarded to extension 001's voicemail inbox (extension 7001). When the call was forwarded to the voicemail inbox, the ringer will stop.

14.3.2. Example 2: Ring Group and Call forwarding

Extension: 000

Forwarding destinations*	001
Ringer time (sec)	5
Forwarding destination (no answer)	001A

Extension: 001A

Forwarding destinations*	100A, 200A
Ringer time (sec)	10
Forwarding destination (no answer)	001B

Extension: 001B

Forwarding destinations*	100B, 200B
Ringer time (sec)	10
Forwarding destination (no answer)	7001

Extension: 100A

Forwarding destinations*	101, 102, 103

Extension: 200A

Forwarding destinations*	201, 202

Extension: 100B

Forwarding destinations*	111, 112, 113, 114, 115, 116

Extension: 200B

Forwarding destinations*	211, 212, 213, 214

1) All incoming calls to the extension 000 are forwarded to extension 001
2) When a call is unanswered for 5 seconds, it will be forwarded to 001A
3) The call will be forwarded to 100A and 200A
4) Extensions 101, 102, 103, 201, and 202 will ring
5) If the call is unanswered for 10 seconds it will be forwarded to extension 001B
6) The call will be forwarded to 100B and 200B
7) Extensions 111, 112, 113, 114, 115, 116, 211, 212, 213, and 214 will ring
8) If the call is unanswered for 10 seconds, it will forwarded to extension 7001
9) When extension 7001 answers the call, all ringing will stop

Meanwhile all extensions that receive the call will keep ringing until their own Ringer time setting expires or the call has been answered.

With this example, we used alphabet character for the extension numbers in order to clarify different groups.
Note: If a caller is using a SIP UA that allows him/her to enter alphabet characters, he/she can make a call directly to a number such as 100B.

14.4. Round robin / Top-down Call Distribution

OnDO PBX features two types of call distribution: round robin and top-down. The round robin setting is useful when you wish to distribute calls equally within a specific group of extensions. The top-down setting is useful when you wish to distribute calls by certain order within the specific group of extensions. In either case, the system will ring each available extension sequentially until the call is answered.

Under round robin setting, the extension that rings first will keep changing in the order as specified. Every time the call will be forwarded to the next specified extension of the one that answered the last call.

Under top-down setting, calls will be always forwarded in a specified order. With OnDO PBX, the number specified in the most left in the field is given the highest priority. The same setting can be configured under the basic call forward settings, but top-down setting is simpler.

When Single attempt setting is set for “no” (default setting), the calls will keep ringing the next destination. The similar setting can be achieved by specifying multiple destinations under “Basic” setting under [Type of Call Forwarding]; however, using this feature is more simplified.

When Single attempt setting is set for “yes”, after one fairer to establish connection, the calls will be forwarded to the destination set at [Forwarding destination (No answer/Busy)].]

Using Call Queue Feature, calls can be kept on hold until specified user answer the call. It is true here that if Single attempt setting set for “yes”, the unanswered call will be forwarded to the destination set at [Forwarding destination (No answer/Busy) after one attempt.

14.4.1. Example 1: Round robin / Top-down

Extension: 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	10,5,5
Forwarding destination (No answer/Busy)	7100
Waiting time in the queue (sec)	0
Single attempt	No

- *Top-down setting*

1) Extension 100 starts ringing
2) If the call is unanswered at extension 100 after 10 seconds, extension 101 starts ringing.
3) If the call is unanswered at extension 101 after 5 seconds, extension 102 starts ringing.
4) If the call is unanswered at extension 102 after 5 seconds, the call will be forwarded to 7100.

- *Round robin setting*

Call forwarding order will be the same as top-down setting the first time; however, from the second call on, the call will be forwarded to the extension after the one that answered the most recent call.

14.4.2. Example 1: Round robin / Top-down

Extension: 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	10,5,5
Forwarding destination (No answer/Busy)	7100
Waiting time in the queue (sec)	0

Extension: 101 (Call forwarding setting [Basic])

Forwarding destinations*	200,201
Ringer times (sec)*	20

Call forwarding settings will be enabled at each extension individually, regardless of the settings of the extension the call was forwarded from. With the example above, when a call is forwarded to extension 101, it will be directed to extension 200 and 201 simultaneously. If the call is unanswered at both extensions after 5 seconds, extension 102 will ring. Extension 200 and 201 will keep ringing till they reach at the time specified in the Ringer times.

14.4.3. Call Queue

Extension 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	20
Forwarding destination (No answer/Busy)	7100
Waiting time in the queue (sec)	180
Max number of calls in the queue	10

Call Queue Feature will be used when the time more than 1 second is set at [Waiting time in the queue (sec)] menu.

Calls will be answered by the available extensions among 100, 101, or 102. The order the extensions answer will depends upon the mode setting. (Round robin, Top-down) If the call is unanswered, it will be placed in the queue.

When the calls in the queue exceed 10, the 11^{th} call will be forwarded to extension 7100. The callers in the queue will hear music on hold while waiting to be answered. If the call is answered after the caller wait 180 seconds in the queue, it will be forwarded to the extension 7100.

After the period set in the [Call interval (msec)], the call in the top of the queue will ring the available extension. When [Single attempt] setting is set for “no”, the call will keep calling till the waiting time for the queue (180 seconds) expire. To avoid missing calls, the call in queue are designed to call different extensions for each trial calls. Thus sometimes it results that the call in the top of queue will be answered first.

14.5. Schedule call forwarding

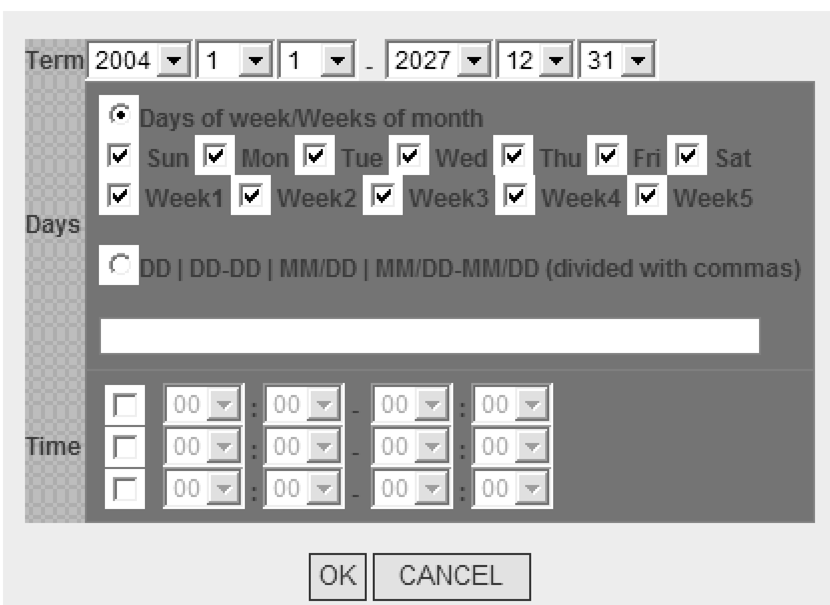
When **Schedule** is selected for the **[type of Call Forwarding]**, call forwarding rules can be defined using conditions based on date, time and Caller ID information.

14.5.1. Schedule setup

Schedule setting is applied when the condition (period, date, hour, or Caller ID) matches. (The action will be same as the basic call forwarding setting.)

To set up a schedule:

1) Select *schedule* as the **[type of call forwarding]** in the **[user setting]** menu.
2) Click the **[save]** button.
3) The menu items **[schedule settings 1]** and **[schedule setting 2]** will appear.
4) To define a schedule, click the **[...]** button. This will open the schedule menu as shown below.
5) Choose the dates and times during which you would like calls to be forwarded.
6) Set call forwarding destinations as you would with the other call forwarding settings.



14.5.2. Example: DD (Date of months)

5,10,15,20,25

This setting will be applied on the 5th, 10th, 15th, 20th, and 25th of every month during the specified term.

14.5.3. Example: DD-DD (Date-Duration)

5-10,15-20

This setting will be applied on between the 5th and 10th; and the 15th and 20th of every month during the specified term.

14.5.4. Example: MM/DD (Specific Date)

5/10,5/20,6/10,6/20

This setting will be applied on May 10th, May 20th, June 10th, and June 20th.

14.5.5. Example: MM/DD-MM/DD (Specific Date- Period)

5/10-6/20,8/10-8/20

The setting will be applied between May 10th and June 20th, and between August 10th and August 20th during the specified term.

14.5.6. Example: Time

08:30 – 13:30
23:30 – 00:30

This setting will be applied between 8:30 am and 1:30pm; and between 11:30pm and 12:30am.

14.5.7. Example: Caller ID information

Applies to (Caller numbers)*	3*
Not Applies to (Caller numbers)*	3001,3002

By adding * (wildcard) after a number, you can specify all numbers that starts with that number.

With the setting example above, all the coming calls that start with 3 except 3001 and 3002 will be forwarded according to this schedule call forwarding setting. The numbers specified in the “Not Applies to (Caller numbers)” field override “Applies to (Caller numbers)”. If the “Applies to (Caller numbers)” field is left blank, all incoming calls will be directed according to this schedule call forwarding setting.

14.6. Add/Remove forwarding destinations

Using this feature, user can add or remove their own number in [Forwarding destinations*] from their phone instead via Web Admintool screen.

14.6.1. Setting up Add/Remove forwarding destinations

Extension: 100

Forwarding destinations	6100
IVR	Add/Remove forwarding destinations
Target users*	300,400

When extension 200 call into extension 100, extension 200 will be added to the forwarding destinations for extension 300 and 400. When extension 200 is already existed in the forwarding destinations for those numbers, extension 200 will be invalid as forwarding destination (remain as forwarding destination with "!" in the front of the extension number).

15. Session Management

The maximum number of session that can be received by extension can be set in the [Max sessions] under PBX setting. Call session will be counted between the beginnings of the conversation till the phone is hanged up, or between ringing till the phone is hanged up.

15.1. Unlimited sessions

When Unlimited sessions are set at [Max sessions], incoming calls will ring the extension even if the extension is in session(s).

The call will be forwarded to the destination set at [Forwarding destination (No answer/Busy)] when it is unanswered by the extension after specified time (Ringer time) or conditions (SIP UA send response as “busy”, “reject”, etc.).

When [Type of Call Forwarding] is set for “Round robin/Top-down”, the call will only ring the available extensions.

15.2. Limited session (0-6)

If the callee’s extension has already hold the session number that set at [Max sessions], the incoming call will be in pending status.

When [Type of Call Forwarding] is set for “Round robin/Top-down”, the call will only ring the available extensions thus it will not be in pending status.

The calls will stay in pending status until the callee’s extension is available to receive sessions or [Forwarding destination (No answer/Busy)] setting is in effect.

15.3. When to use Maximum session setting

- *Preventing missing calls*

  By setting [Max sessions], users can avoid missing calls. When multiple calls are coming in the extension instantly, the calls except the first call will be forwarded to the destination set at [Forwarding destination (No answer/Busy).

- *Setting the preferred number of simultaneous calls*

  For the SIP UAs that have capacity to handle multiple sessions, users can limit the preferred number of simultaneous calls. And make other lines available for outgoing sessions.

16. Call hold / Call Transfer

OnDO PBX Users can use Call hold and Call Transfer features. (For information about operation, refer to “OnDO PBX User’s Guide.”

16.1. Call hold

A user who wishes to use the Call hold feature needs to be registered as a user of OnDO PBX. Call Hold permissions can be set in the **[User Setting]** menu, at the **[transfer/hold]** option.

16.1.1. Call hold – Activation/Deactivation

When OnDO PBX Users A and B are on the phone with each other. (Fig. 1), either user can place the call on hold by entering the hold command (fig 2). Taking a call off hold can only be done by the user who placed the call on hold, so the conversation won't be resumed until User A takes User B off hold.

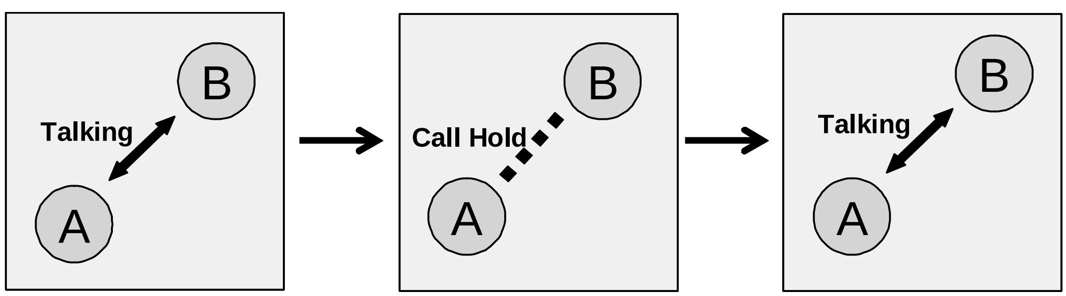


Figure 1 Figure 2 Figure 3

16.2. Initiating Call Transfer

After putting a call on hold, OnDO PBX users can transfer a call to the other users.

16.2.1. Initiating Call Transfer 1 – Calling the 3rd party

User A puts the call with User B on hold (Figure 4), and initiates a new session with User C. (Figure 5) When User C answers the call, the session between User A and User C will be connected.

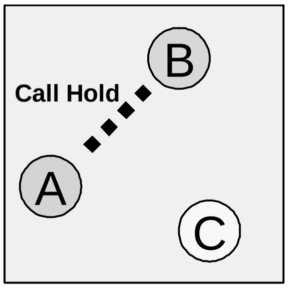


Figure 4

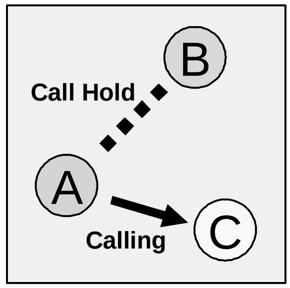


Figure 5

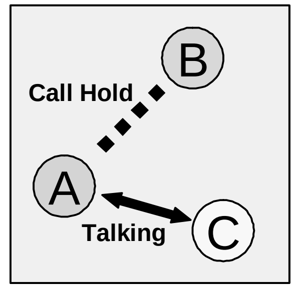


Figure 6

16.2.2. Initiating Call Transfer 2 – Calling multiple Users

If User C had set [Forwarding destinations] for his/her incoming calls, the call will be transferred to the specified destinations. With this example, User C has set User C and User D for the destinations. (Figure 7, 8) The first user who answers the phone will be connected with User A.

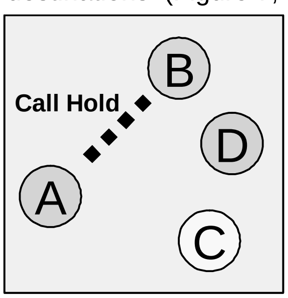


Figure 7

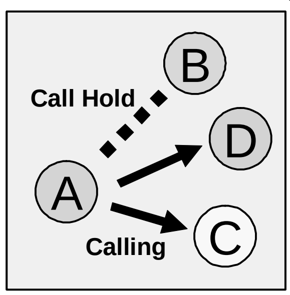


Figure 8

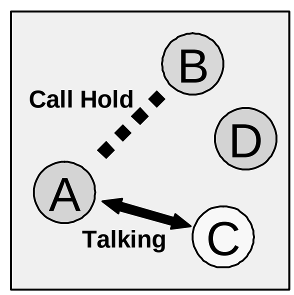


Figure 9

16.3. Processing Call Transfer

After talking with the transfer target user, you will have the opportunity to go back to the original session with User B.

16.3.1. Processing Call Transfer 1 – Attended Transfer

After User A initiated Call Transfer, the call between Users A and B was put on hold. When User A hangs up the phone while Users A and B's session is connected (Figure 10, similar situation with Figure 6 and 9), the call with Users A and B will be transferred to User C. Now Users B and C are connected.

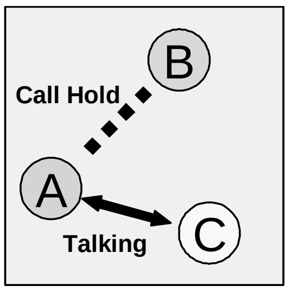


Figure 10

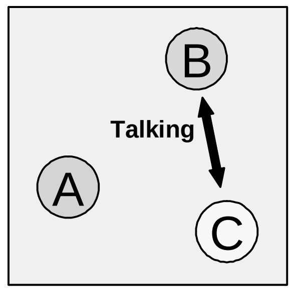


Figure 11

16.3.2. Processing Call Transfer 2 – Unattended Transfer

While the session between Users A and B are on hold, User A tries to connect with User C. (Figure 12, similar situation with Figure 6, 9, and 10) User A hangs up the phone before User C answers the call, then the call will be transferred to User C and User B wait for User C to answer. (Figure 13)

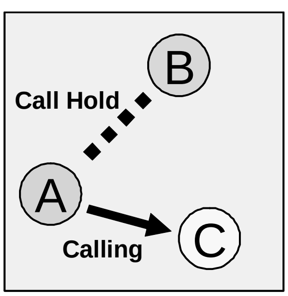


Figure 12

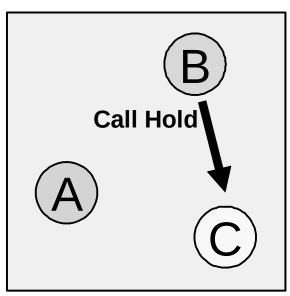


Figure 13

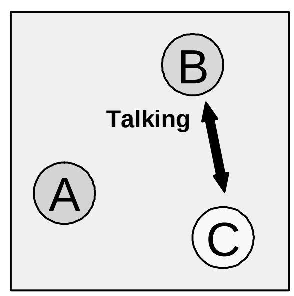


Figure 14

✓ *Using Unattended Transfer User A cannot retrieve the call with User B even when User C didn't answer the phone.*

16.4. Canceling Call Transfer

You may cancel Call Transfer in the middle of the process, and either go back to talk with the original caller or transfer the call to the different destination.

16.4.1. Canceling Call Transfer 1 – Canceling Transfer while talking with the 3rd party

There are 2 ways to cancel Call Transfer when you are talking with the 3^{rd} party

1) User A executes the cancel Call Transfer command
2) User C hangs up call with User A

When the Call Transfer process was canceled while User B was on hold (Figure 16, similar situation with Figure 6, 9, and 10), the call will be back to the hold state with User A and B. (Figure 17)

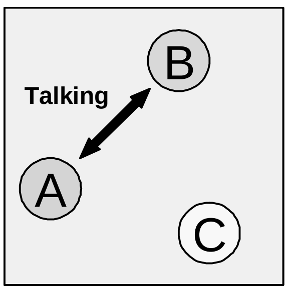


Figure 15

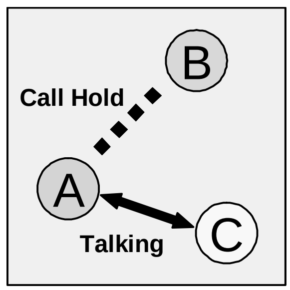


Figure 16

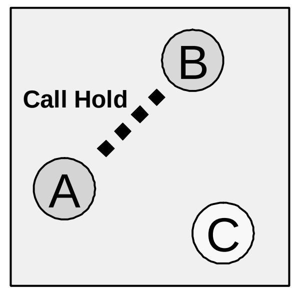


Figure 17

16.4.2. Canceling Call Transfer 2 – Cancel Transfer while ringing the 3rd party

There are 2 ways to cancel Call Transfer when session is ringing for the 3^{rd} party

1) User A executes the cancel Call Transfer command
2) When a call to User C cannot be completed (User C's phone is not connected, Call blocking, etc.)

You can cancel Call Transfer while the phone is ringing at the destination, User C. (Figure 19, similar situation at Figure 5 and 8) The call status goes back to the hold status between Users A and B.

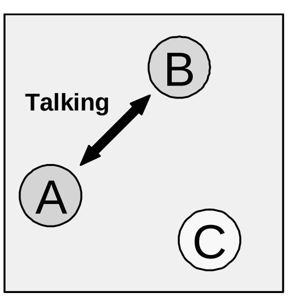


Figure 18

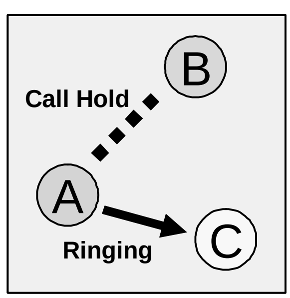


Figure 19

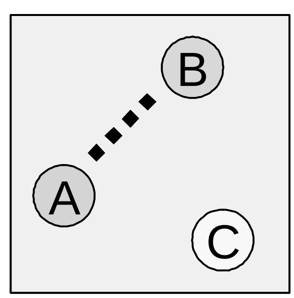


Figure 20

16.5. Cautions for Call Transfer

16.5.1. Extending Call Transfer

In the state of Figure 21 with regular settings, User C cannot initiate Call Transfer with the call transferred to him/her. For User C to transfer the transferred call, either User A or B need to hang up the call first. Then User C can transfer call to User D.

(If you really need to transfer the call to the 4th party, you need to write settings in Dial Plan of OnDO SIP Server to redirect the call from User A to C to OnDO PBX.)

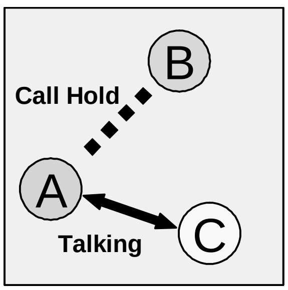


Figure 21

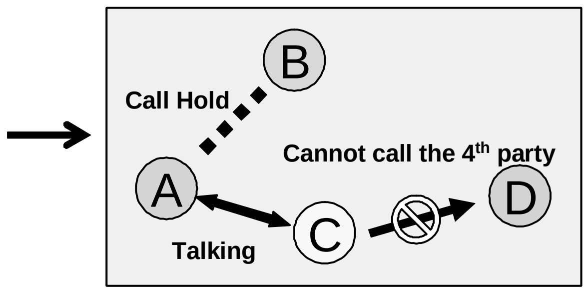


Figure 22

17. Auto Attendant

Auto Attendant is an automated receptionist system. Calls made to a specific extension are directed to a greeting system, which plays a recorded message and asks for the extension of the party the caller wishes to reach. Calls are then forwarded to the requested extension.

1.1. Setting up Auto Attendant

Set up for Auto Attendant is done by the System Administrator through the OnDO PBX Admintool. For detailed set up information, please refer to Section 6.4 Setting the Auto Attendant.

17.1.1. Accessing Auto Attendant

User: 001

Choose **[User Setting] > [Administrative setting]**

IVR	Auto Attendant

Calling “6001” (IVR prefix+ User extension) connects you to the Auto Attendant.

✓ *Usually the Auto Attendant is set up to receive calls from outside the OnDO PBX system. In that case, you will need to set your PSTN Gateway or OnDO SIP Server to direct call sessions from the PSTN Gateways to the Auto Attendant extension number.*

17.2. Cautions for Auto Attendant

17.2.1. Call Transfer to Auto Attendant

When a call is transferred (Attended Transfer) to the Auto Attendant (Figure 23), Auto Attendant cannot transfer the call. (Figure 24) Refer to "14.5.1 Extending Call Transfer."

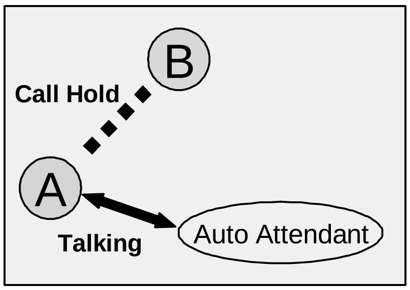


Figure 23

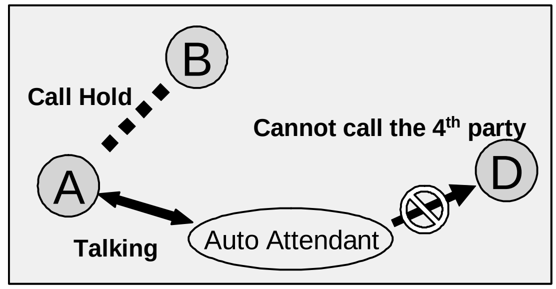


Figure 24

However, Call Transfer to the 4th party is possible when the Call Transfer is initiated from another Auto Attendant. Since Auto Attendant operates automatically and the first Call Transfer is done instantly, 2nd Auto Attendant can transfer the call to the User D. (Figure 25, 26)

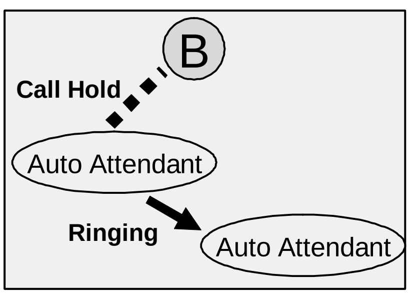


Figure 25

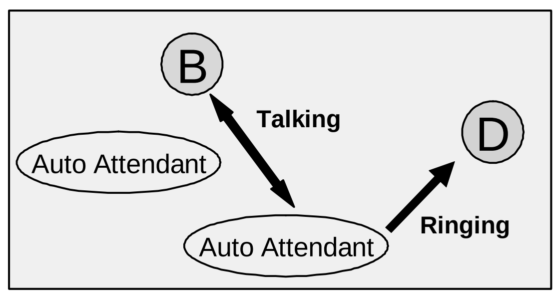


Figure 26

17.2.2. Call Transfer from Auto Attendant to the non-OnDO PBX Users

Even when **[Transfer to unregistered users]** is disabled under **[User Setting]**, Call Transfers from Auto Attendant can be made if the non-OnDO PBX User's number or SIP URI are set at **[Default operator]** field or **[Speed dial]** field in the **[Auto Attendant]** section of the **[User Setting]** menu.

17.3. Auto Attendant Setting Example

User: 001

Auto Attendant Setting

Default operator	001
Speed dial*	6=001, 7=101

User: 001

Forwarding destinations*	001, 002

It is handy to set up **[Speed dial]** for using the Auto Attendant feature.
Your Auto Attendant guidance could say:

"If you know your party's extension, please enter it now. For sales, press 6. For support, press 7."

If 6 or 7 are pressed, the call will be immediately transferred to the appropriate destinations. User 001 has defined multiple destinations for its **[Forwarding destinations]** setting as Ring Group setting. If there is no entry from the caller, the call will be transferred to the destination that is set in the **[Default operator]**.

- ✓ *With this setting, you won't be able to use the numbers that start with 6 or 7 for extension numbers.*

18. ARS

Most of the call management rules for outgoing and incoming calls through outside lines are set at Automatic Route Selection (ARS). Common outside lines are connected to;

- *The third party SIP Servers*

  Any SIP server that is not specified as the SIP server for OnDO PBX (defined at **[option] > [SIP Setting] > [SIP Proxy Address]**) is considered to be third party SIP server. Common third party SIP servers are those that are located at VoIP service providers, or other office locations.

- *Gateways*

  The gateways referred to here are SIP PSTN Gateways that connect digital and analog PSTN lines or software gateways that convert between different protocols.

Defining rules enables ARS to make optimal use of outside lines, and your gateways. In the **[ARS]** menu, you can define extension numbers that will receive incoming calls, and which lines to use for outgoing calls.

Here are some examples of uses for ARS feature:

Select the least expensive calling options

For example, you can send all international calls through a specific call long distance carrier, or to connect to the branch office located in Taipei. With ARS feature, the least expensive calling option can be utilized.

Effective port use of PSTN Gateways

When there are multiple PSTN Gateways are installed in the office, some ports may be used frequently than others. The ARS feature can select the most effective PSTN Gateway port automatically.

Effective use of outside lines

Similar to the above example, when you subscribe to multiple VoIP service providers, some phone numbers may be used frequently than others. The ARS feature can select the most effective service provider automatically.

Network traffic management and Load balancing

When many sessions are in progress and network bandwidth is insufficient, some calls can be directed through PSTN gateways

18.1. Route search process

Two sessions are used for one call in OnDO PBX.

For a simple call between two users, two route searches are done. The first search checks for possible route matches, and the second search selects a route to use. The first route search rules are defined in the IN pattern settings, and the second route search rules are defined in the OUT pattern settings.

Route Search is done by searching for Matching Patterns that apply to a call. A Route Search starts from the highest priority (lowest number) as defined by the ARS rules. If multiple patterns share a priority level, a pattern is selected randomly. If a rule has a **[Priorities No]** value that is lower than the number defined in the **[Ignore Priorities smaller than No]** field, that pattern will be ignored. In the case when a rule has multiple priorities specified, the priorities will be handled based on the value set in the **[priorities]** field, not in the order they were listed.

When a matching pattern is found, the process will check if the Max Sessions value has been reached. If the pattern has sessions available, the process will select the pattern and convert the FROM and TO header fields according to the settings defined in **[Deploy Patterns]**.

Routes of sessions incoming to OnDO PBX cannot be changed. It is possible to define conditions for changing routes of incoming calls using priority, but to do so have no effect. For incoming sessions, ARS is only used for checking for route matches, and setting a Dial-in number. You can choose to forward incoming calls to multiple extensions or to the Auto Attendant in the **[User Setting]** menu. You also need to take care to ensure that calls between extension users don't match ARS rules, because those calls need not be routed through your gateway and third party SIP provider.

For outgoing calls, it is necessary to set priorities for effective routing depending on the usage of lines.

18.2. Matching patterns

Using regular expressions enter conditions of TO or FROM headers for matching process. When the field is left blank, all calls will be considered to match the condition. This field can be set in the same way as the Dial Plan Matching pattern in OnDO SIP Server Admintool.

18.3. Deploy patterns

This is similar to OnDO SIP Server Dial Plan's Deploy patterns; however, there are several differences:

- *Numbers follow the "$" sign in deploy patterns (OSS dial plan uses %)*
- *Conversion of TO header that matches Matching Pattern is done as described in Deploy Pattern for TO header. Conversion of From header is done as defined in Deploy Pattern for FROM header and in DTMF field.(Multiple lines can be defined in Matching Pattern and Deploy Pattern in an OSS Dial Plan rule and it is process at the same time)When FROM and TO field are blank, no conversion is performed. Also when DTMF field is blank, DTMF will not be sent.*

18.4. Priorities

18.4.1. Example: 1 Priority

Suppose Matching Patterns for outgoing calls in Route A and Route B are the same. Both Route A and Route B can have 4 simultaneous sessions (as in the case of a 4 port PSTN gateway). For incoming sessions, both rules are set at [No.1] to accept any sessions that match with [IN] Matching Pattern.

Route A

No.	Direction	Max Sessions	Priority
1	IN	-1	1
2	OUT	2	100
3	OUT	4	120

Route B

No.	Direction	Max Sessions	Priority
1	IN	-1	1
2	OUT	2	101
3	OUT	4	121

For next outgoing call, route search will be done in the following order

1) If there are less than 2 sessions in use on route A, use route A.
2) If there are less than 2 sessions in use on route B, use route B.
3) If 4 sessions are not being used in route A, use route A.
4) If 4 sessions are not being used in route B, use route B.

19. Voicemail

Each OnDO PBX User can have his/her own voicemail inbox.

1.1. Voicemail inbox number

The OnDO PBX system treats a Voicemail inbox number the same as a User ID or extension number.

19.1. Accessing Voicemail

Users can access their voicemail inbox by pressing the voicemail prefix number plus their extension number. For details, refer to Section 12.2 Prefix.

Example: Extension (or User ID) --- 001

- Dialing 7001 → access his/her voicemail inbox directly to leave messages
- Dialing 8001 → retrieve his/her messages, change settings for voicemail inbox
- Pressing "*" (Star) while listening to the voicemail guidance → same as dialing 8001

By default, each user's voicemail inbox is set for the destination of **[Call forward (no answer)]**. For details about **[Forwarding destination (no answer)]**, refer to **Section 12. Call forwarding / Call forwarding (no answer).**

When a voicemail inbox receives a call, specified guidance will be played for the caller. You can choose voicemail guidance from 2 personalized greetings and the default greeting.

19.2. Greeting message

19.2.1. Recording / Uploading Greeting messages

Users can record their personalized Greeting messages through **[Review / Option]** (Refer to "Voicemail Navigation Map") or upload them through OnDO PBX Web Admintool.

19.2.2. Selecting Greeting message

Users can have two personalized greeting messages, "Personal Greeting" and "Alternative Greeting." Common use of the "Alternative Greeting" is for special occasions such as long holidays, sick days, business trips, etc.

Selecting Greeting messages can also be done through calling the voicemail inbox and accessing **[Review / Option]** or through the OnDO PBX Admintool.

19.3. Managing messages

Maximum length of the voicemail messages that callers can leave is set in the Administrator's **[Option]** menu. The maximum number of voicemail messages that users can save in their inboxes is also set in the **[Option]** menu. Users can also set to forward the voicemail messages to specific extensions or destinations. For details, please refer to the section "16.9 Voicemail messages: Forwarding and Email notification."

19.4. Retrieving voicemail messages

There are a few ways to retrieve voicemail messages:

- *By downloading sound file (WAV file) from the **[User setting]** in OnDO PBX Admintool*
- *By calling your voicemail inbox and accessing **[Review / Option]** menu*
- *By using Email notification setting, and opening sound file (WAV file) attached to the email*

19.5. Voicemail Message Status

There are 2 states for your voicemail messages

- *New message → a message that you haven't saved or erased including newly arrived messages*
- *Saved message → a message that you have chosen to save on the system*

19.5.1. New message status

This status is for the newly arrived voicemail messages, including forwarded messages from other users. This message will remain in the inbox up to the number of days set in the **[Days to keep unsaved messages]** in the **[Option]** menu.

19.5.2. Saved message status

Users can save Voicemail messages when they are in message review. Saved messages will **not** be deleted even after the days set in the **[Days to keep unsaved messages]**.

19.6. Retrieving Voicemail messages

Users can listen to their voicemail messages by accessing **[Review / Option]** menu.

19.7. Email Notification

When a new message arrives at the user's voicemail inbox, an email notification will be sent to the specified email address.

19.7.1. Set up Email Notification

Enabling or Disabling the email notification features can also be done in the **[Review / Option]** menu. Other settings can only be done through the OnDO PBX Admintool. In the OnDO PBX Admintool, you can set up your email address to receive notifications and choose whether or not to receive voicemail message as an attachment with the notification or not. (For details, please refer to section 16.9 Voicemail messages: Forwarding and Email notification)

19.8. Voicemail messages: Forwarding and Email notification

19.8.1. Forwarding Voicemail message

If the User has set up voicemail messages forwarding, (**[User Setting]** menu > Voicemail setting **[Message forwarding*]**), all Voicemail messages will be forwarded to the specified destinations. You can assign multiple destinations to receive the voicemail messages. If you want to leave a copy of voicemail message as well as forwarding to destinations, you will need to specify your voicemail inbox number in the field as well.

You can also forward voicemail messages to forward to a group of destinations by using the forwarding settings.

19.8.2. Email notifications

You may choose receive email notifications when new voicemail messages arrive in your inbox. You can receive email notifications for messages that have been forwarded to other users as well.

Example:

- User 001 sets messages to forward to User 002 → Only User 002 will receive voicemail message that came in to User 001
- User 001 enters both email addresses of User 001 and 002 → Both User 001 and 002 will receive email notifications when User 001 receives a voicemail message

19.8.3. Max messages

OnDO PBX System Administrators can set the limit of the number of messages that users can have in their voicemail inboxes. **[Option] > [Media Server system setting] > [Max messages]** When the number of voicemail messages in the inbox reaches the maximum number, callers cannot leave messages in your voicemail inbox. Callers will hear the guidance, “The person you are trying to reach is unavailable. To leave a message, please wait for the tone.”

A message forwarded to a user who has reached the maximum number of messages in their inbox will not be accepted. If you are forwarding your voicemail messages to multiple destinations, the message will also be forwarded to inboxes that haven’t reached the maximum number of messages.

19.9. Voicemail inbox access limitation

Each voicemail inbox will only allow a single user access at a time. When you are accessing voicemail inboxes using a phone, you cannot modify your settings through the OnDO PBX Admintool. When some other person is using your voicemail inbox, your voicemail will be locked for other accesses. Sometimes the lock-outs may persist for half a minute or so after a user exits from his/her voicemail inbox.

19.10. Creating / Deleting Voicemail inboxes

This feature is only available to the system administrators. Creating or deleting a voicemail inbox is the same as creating or deleting an OnDO PBX User. You may create/delete voicemail inboxes as follows.

Using the phone to create / delete voicemail inboxes:

1) In the OnDO PBX Admintool choose **[User Setting] –>[IVR] > [Setup]**
2) Call the number (IVR prefix) + (User number) (ex. 6001 for User 001)

Follow the guidance to create/delete the voicemail inboxes (For details, please refer to “Voicemail Navigation Map.”)