OnDO PBX

Version 1.5

Administrator's Guide
(Basic)

Brekeke Software, Inc.

Version

OnDO PBX v1.5 Administrator's Guide (Basic)

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1. Introduction

OnDO PBX is an enhanced office telephone system that provides robust, high performance, and intelligent IP-PBX functionality. It is a software IP-PBX that supports the industry-standard VoIP protocol SIP. Designed for ease of use and scalability, OnDO PBX delivers all the features of a PBX.

The following is a brief list of some of the features included with OnDO PBX:

- Auto Attendant
- Call Forwarding
- Call Conference
- Call Monitoring
- Call Hunting
- Call Pickup
- Call Recording
- Call Hold
- Call Transfer
- Call Queue
- Ring Groups
- Voicemail, with Voicemail forwarding, and notification by email

This document explains the installation and basic configuration of OnDO PBX. For more advanced setting instructions, please refer to the OnDO PBX Administrator's Guide (Advanced).

2. System Requirements

OnDO PBX runs on the following environments:

OS	Microsoft Windows XP/2000, Red Hat Linux 7.x/8.x or later, Solaris 10
Java	JDK 1.4 or newer

✓ *You must install Apache Tomcat v4.1.2 or later if you use an OS other than Windows.*

3. Installation (for Windows OS)

For instructions on how to update OnDO PBX, please refer to the section called [Update] in this document.

3.1. Step 1: Installing Java2 Platform, Standard Edition (J2SE)

You must install the Java2 Platform, Standard Edition (J2SE) before installing the OnDO PBX software.

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

3.2. Step 2: Installing OnDO PBX

- Obtain the file “pbx<version_#>.exe” from Brekeke Software Inc.
- 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 did not 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 PBX]**>** [OnDO PBX Admintool].
2) You will be asked to enter OnDO PBX Product ID. Enter the 16 digit 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) To change the display language of the Admintool, please select [Users] > Click 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 next section. In this case, you will need 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 and Solaris)

For information on updating to the newest version of OnDO PBX, please refer to the "Updates" section of this document.

4.1. Step 1: Installation of J2SE SDK

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

4.2. Step 2: Installation of Tomcat

- 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.
- Set J2SDK Install directory for the environment variable JAVA_HOME.
- Install using the download file.

4.3. Step 3: Installation of OnDO PBX & OnDO SIP Server (bundled with OnDO PBX)

- Obtain files "pbx.war" and "proxy.war" from Brekeke Software.
- Copy those 2 files directly into the \webapps directory which is under the Tomcat install directory.

4.4. Step 4: Starting Tomcat

- Start Tomcat.
- 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.)
- Tomcat has started successfully if the Apache Jakarta Project page is displayed.

4.5. Step 5: Starting OnDO PBX Administration Tool

- 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.
- Enter values for User ID and Password. Then click the [Login] button.

[Default] User ID & Password (Case sensitive)

User	sa
Password	sa

- If you wish to change the display language of the Admintool, please do the following: Select [Users] **>** Click the Edit button of Admin user**>** Select desired language from [General setting]**-**[Language]
- 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. Updates

If you are using the trial version of the software, updates to newer versions are done through the website Download page. Go to Brekeke.com to find the latest software download.

If you have purchased a license and want to update to a newer version, you will need to go to the Brekeke Support Forum and download the new files from the Premium Support Forum. You will need to register as a Premium User before being able to access the files. You can register by entering your product ID into the user profile information.

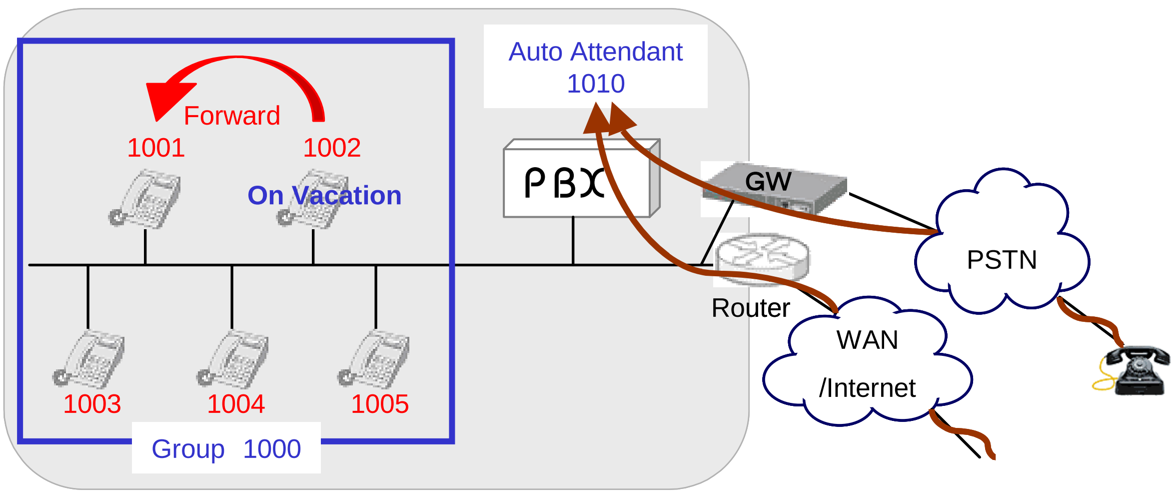
These are the instructions to follow once you have downloaded the file and are ready to update the files. Updating OnDO PBX can be done through [Update software/Service pack] without losing your current customized settings.

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. Example of OnDO PBX Set Up

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.*



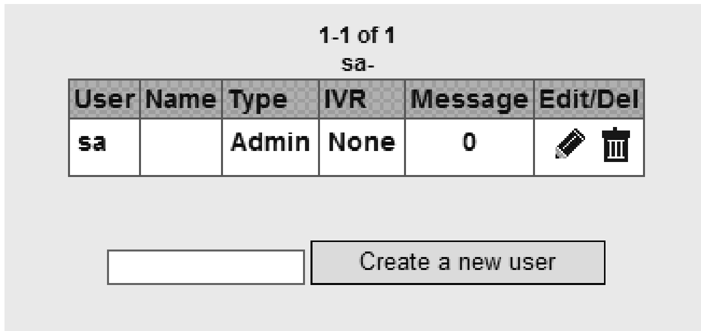
[i] A group of local extensions that ring in unison. When calls are made to the group number, any available group member can pick up the call.

6.1. Registering Users

This section explains how to set up extensions for users.

OnDO PBX Admintool > Users

1) Choose the **[Users]** menu and open the **[Users]** 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.

6.1.1. Default Values of Users

Setting item	Details of default values
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 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. Voicemail Settings

As described in section 6.1.1, user 1001's [Forwarding destination (No answer/Busy)] is set for "71001" by default. This prefix 7 in 71001 is the prefix for reaching the voicemail inbox directly. If 1001 doesn't answer for 90 seconds after your call to 1001, the call will be forwarded to voicemail. For your voicemail inbox, each user can create their own personal greeting messages and upload it from [Message files: Download/Upload] field at the User Setting.

6.3. Voicemail Notification by Email

6.3.1. Setting Email Sender

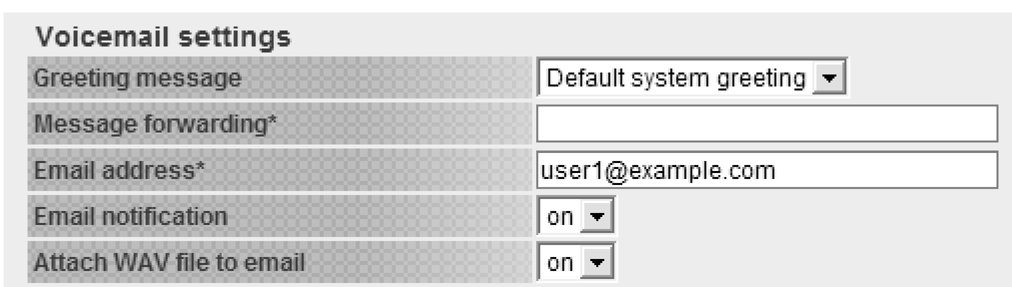
Current version of OnDO PBX supports a Mail server which does "POP before SMTP" authentication or "SMTP" authentication. Set up an Email sender as follows:

OnDO PBX Admintool > Option

6.3.2. Setting up Email Recipient

Set up an Email recipient at each user setting:

OnDO PBX Admintool > Users > Edit



6.4. Message Waiting Indicator (MWI)

OnDO PBX can send voicemail notifications to the phones that support MWI.

6.4.1. PBX Settings for MWI

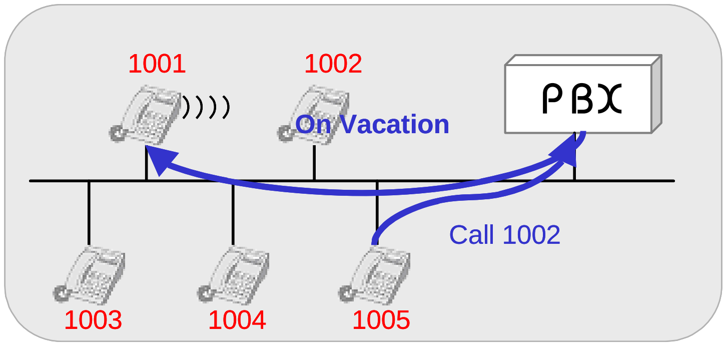
To Enable Message Waiting Indicator, set:

OnDO PBX Admintool > Options ＞ Message Waiting Indicator = on

6.4.2. Phone Settings for MWI

To activate the MWI feature on the phone, enable a “Subscribe” for the Message Waiting Indicator if such setting is available on your phone. If there is a special button to retrieve a message, assign a number to retrieve a voicemail message (Example: 81001 for the user 1001) to the button.

6.5. Setting Up Call Forwarding

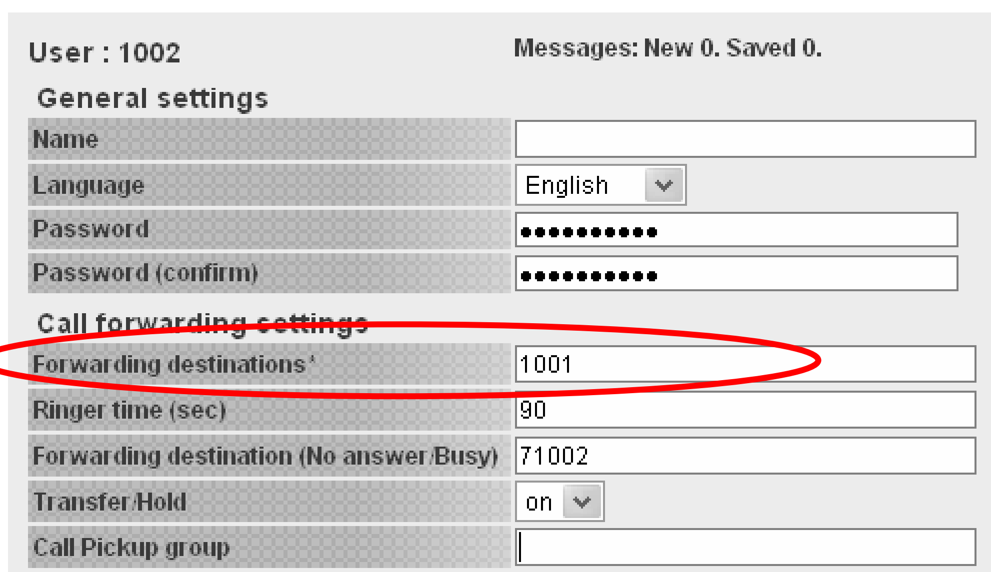


The user at extension 1002 will be on vacation for several days. Calls to extension 1002 will be forward to extension 1001.

OnDO PBX Admintool > Users > Edit

1) Click the menu **[Users]** to open the **[Users]** 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 1002 receives calls, they will be forwarded to extension 1001.



User : 1002 Messages: New 0. Saved 0.

General settings

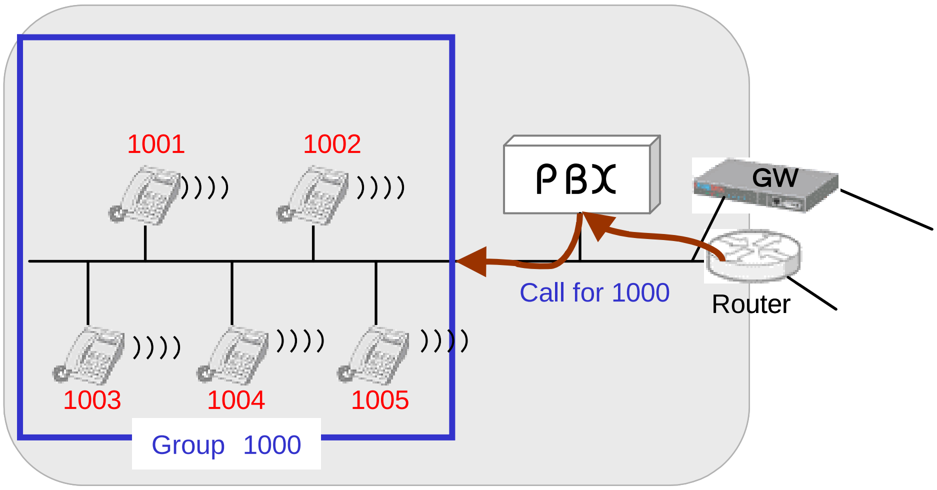
Name	
Language	English
Password	●●●●●●●●●●●
Password (confirm)	●●●●●●●●●●●

Call forwarding settings

Forwarding destinations*	1001
Ringer time (sec)	90
Forwarding destination (No answer/Busy)	71002
Transfer/Hold	on
Call Pickup group	

6.6. Setting up Ring Group

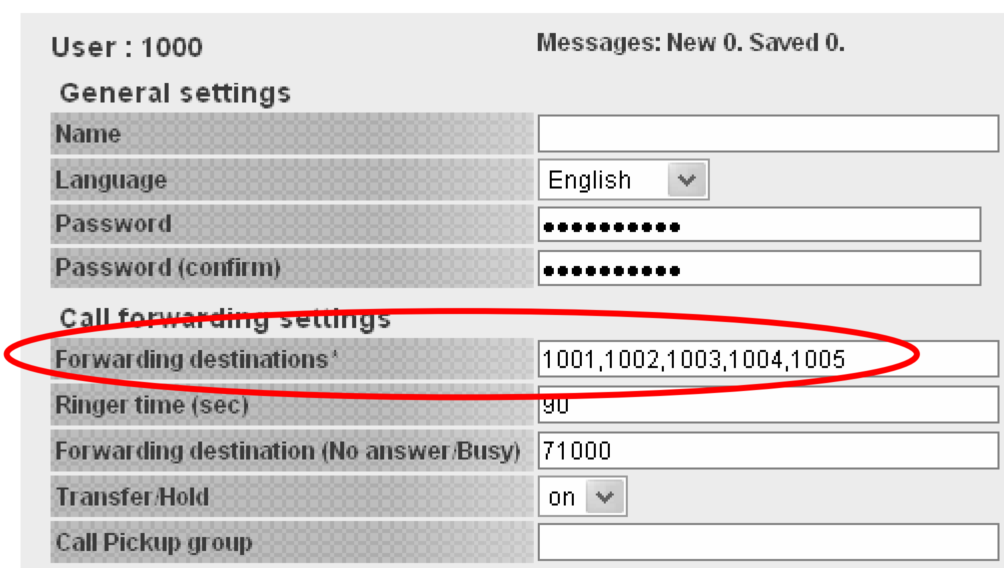
In a Ring group when any extension in the ring group is called, all the extensions in the group ring at the same time.



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

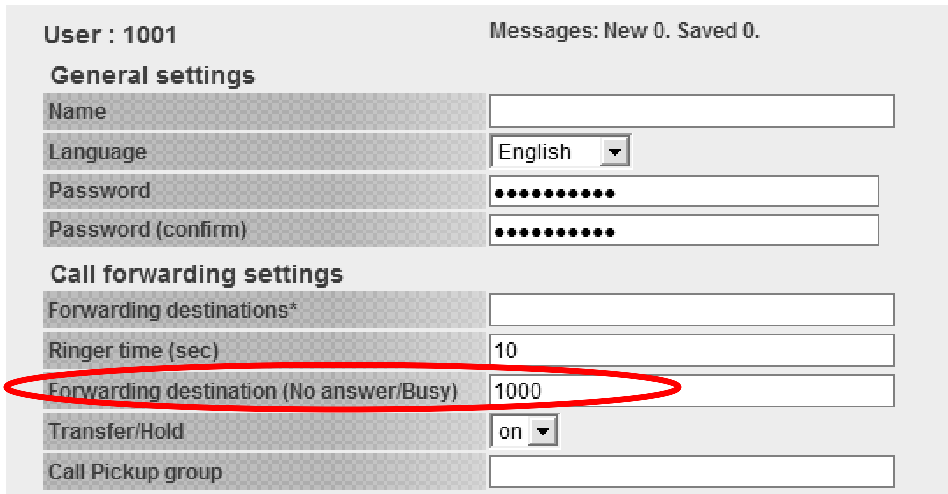
OnDO PBX Admintool > Users

1) Choose the **[Users]** 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 [Users] window.
4) Enter 1001, 1002, 1003, 1004, 1005 in the **[Forwarding destination]** field.
5) Click the **[Save]** button to save changes.



6.7. Setting Up No Answer / Busy Call Forwarding

If a user 1001 wants to forward calls to other users instead of using Voicemail answering while user 1001 is not available, set the users in **[Forwarding destination (No answer/Busy)]**. In the following example, calls will be forwarded to 1000 (Ring group) if 1001 doesn't answer for 10 seconds.



6.8. Setting Up 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 on how to set up a Ring Group, please refer to section 6.3 Ring Group Setup)

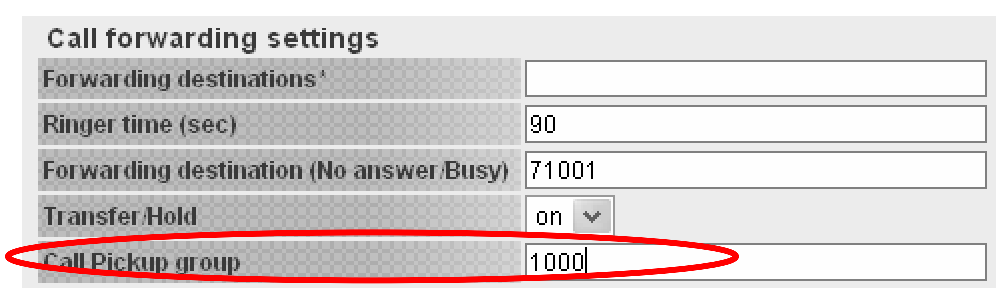
  Extension: 1000

Ring Group	1001,1002,1003,1004,1005

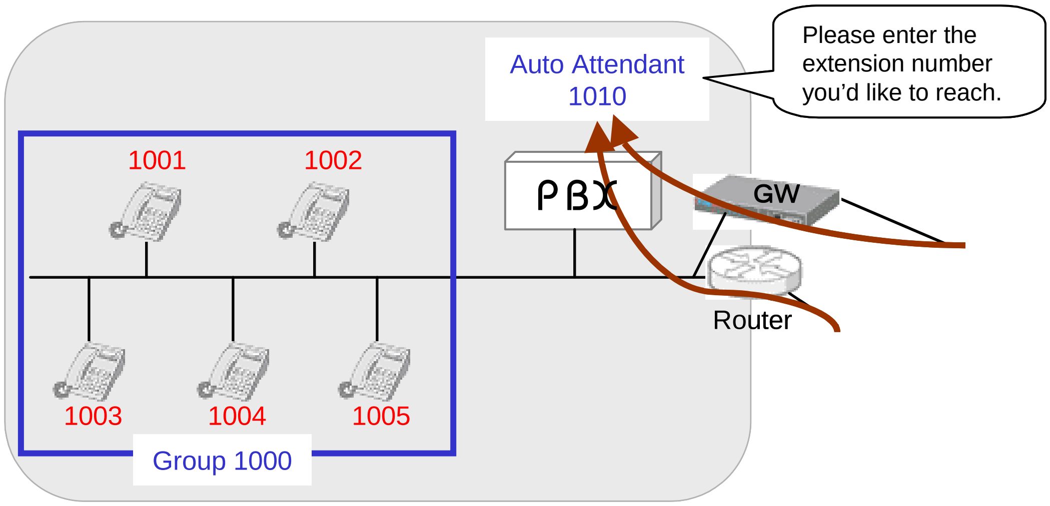
  Dialing ***1000** (Ring Group extension number) enables one to pick up any call that is directed to any of the extensions in the Ring Group. Incoming calls can still be answered by dialing * + extension for each number; however, using the group extension number does not require you to remember each extension 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).



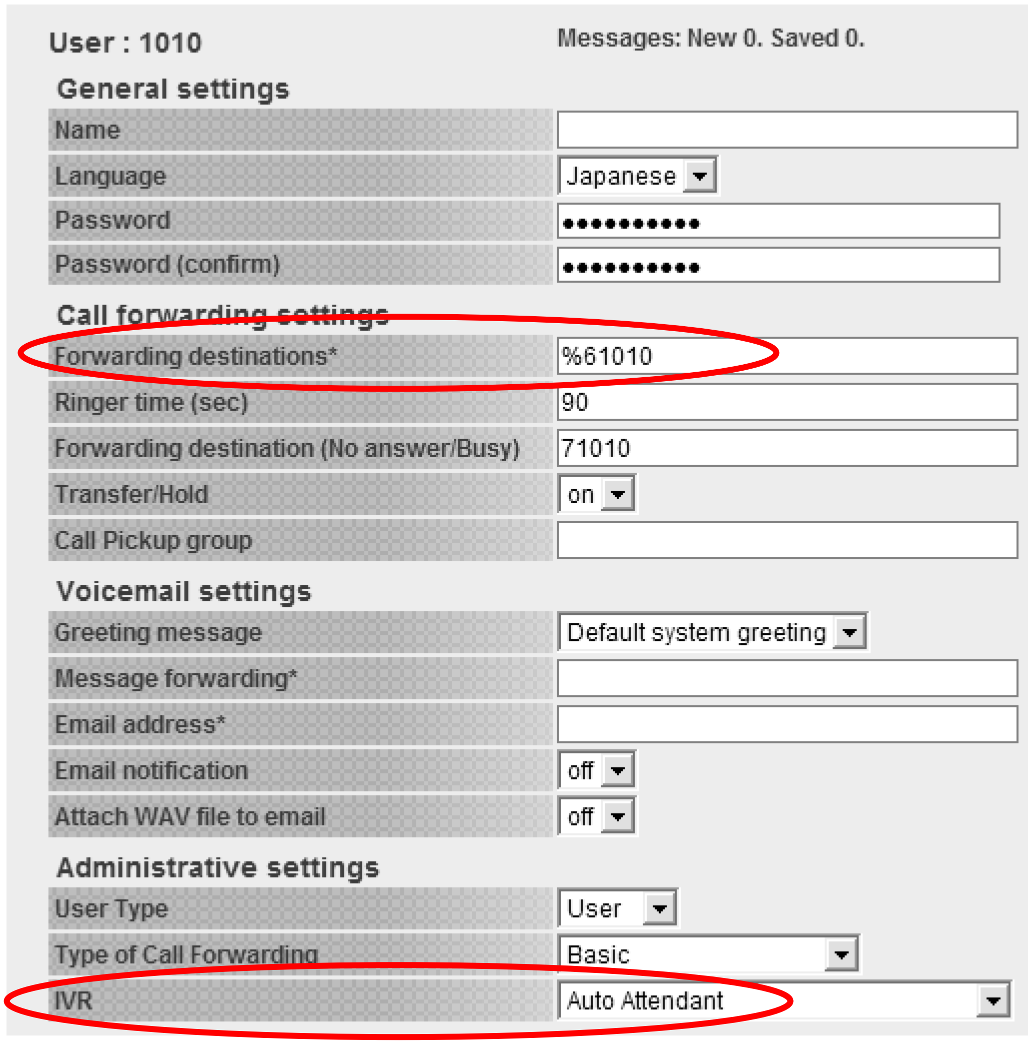
6.9. Setting Up Auto Attendant



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

OnDO PBX Admintool > Users

1) Choose the **[Users]** menu.
2) To create an Auto Attendant extension, enter 1010 in the entry field, and 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 7.4 User Setting.)
6) Click the **[Save]** button to save changes.



User : 1010 Messages: New 0. Saved 0.

General settings

Name	
Language	Japanese
Password	●●●●●●●●●●●
Password (confirm)	●●●●●●●●●●●

Call forwarding settings

Forwarding destinations*	%61010
Ringer time (sec)	90
Forwarding destination (No answer/Busy)	71010
Transfer/Hold	on
Call Pickup group	

Voicemail settings

Greeting message	Default system greeting
Message forwarding*	
Email address*	
Email notification	off
Attach WAV file to email	off

Administrative settings

User Type	User
Type of Call Forwarding	Basic
IVR	Auto Attendant

6.10. Setting Up Call Queuing

Even when all lines are busy, you can avoid missing calls by utilizing the Call Queue feature. A new call is coming to the Ring Group number 1000, but all members 1001-1005 are busy, the call can be put into a Call Queue where the caller will hold until someone is available to answer.

OnDO PBX Admintool > Users > Edit

1) Click the menu **[Users]** to open the **[Users]** window.
2) Click the **[Edit]** button for user 1000 and open the edit window for the extension.
3) Select “Round robin/Top-down” at [Administrative settings] > [Type of Call Forwarding] field and click [Save] button.
4) Set Call forwarding settings.

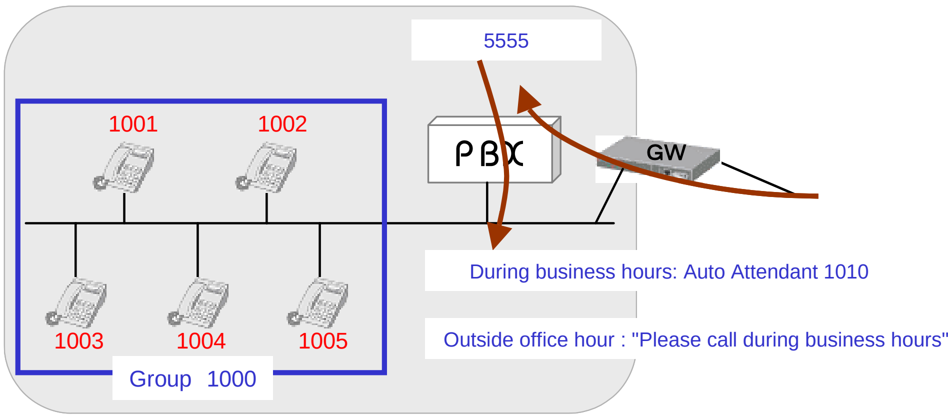
   In this example, the call is put in a queue while all of the group members 1001-1005 are busy. If any member becomes available within the time set in [Waiting time in the queue (sec)], OnDO PBX will ring the available member. If all members continue to be busy after the specified period at [Waiting time in the queue (sec)], the call will be forwarded to the destination set in [Forwarding destination (No answer/Busy)] (user 1000’s voicemail 71000 in this example).

Call forwarding settings

Forwarding destinations*	1001,1002,1003,1004,1005
Ringer time (sec)*	10
Waiting time in the queue (sec)	30
Max number of calls in the queue	10
Call interval (msec)	3000
Single attempt	no
Forwarding destination (No answer/Busy)	71000
Mode	Round robin
Transfer/Hold	on
Call Pickup group	

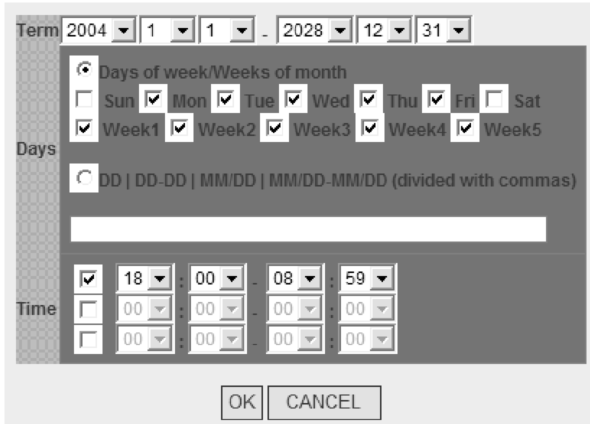
6.11. Setting Up Call Forwarding Schedule

In this example, during their business hours, PSTN callers will be directed to the user 5555 first and then directed to Auto Attendant 1010. After their regular business hours, callers will hear "To speak with a live operator, please call during regular business hours. Our regular business hours are Monday through Friday, 9 a.m. to 6 p.m." Here, the VoIP Gateway setting need to be changed to direct PSTN calls to user 5555.

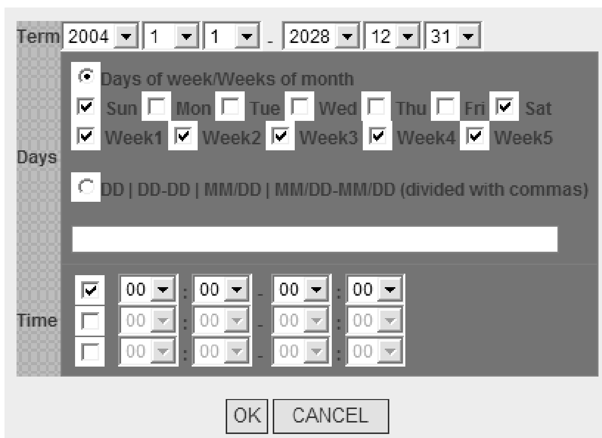


OnDO PBX Admintool > Users

1) Choose the **[Users]** menu.
2) To create an extension 5555, enter 5555 in the entry field, and then click the **[Create a New User]** button. User 5555 will be added.
3) Click the edit button for user 5555, to open the edit window for the user 5555.
4) Select "Schedule" at **[Administrative settings]** > **[Type of Call Forwarding]** field and click **[Save]** button.
5) Create a wav file for the message "Please call during business hours". Select "Voicemail personal greeting" at **[Message files: Download/Upload]** and upload the wav file.
6) Select "Personal greeting" at **[Voicemail setting]** > **[Greeting message]**.
7) Click "…" at **[Schedule setting 1]** > **[Schedule]**. A window will pop up. Select Monday through Friday and 18:00 - 8:59 there.

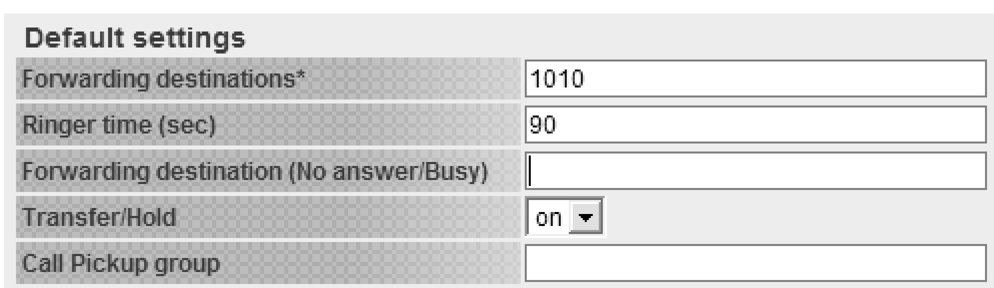


8) Click "…" at **[Schedule setting 2]** > **[Schedule]**. A window will pop up. Select Sunday, Saturday, 24 hours (00:00 – 00:00).



9) Enter 75555 (Voicemail for 5555) at both **[Schedule setting 1]**> **[Forwarding destinations]** and **[Schedule setting 2]** > **[Forwarding destinations]**

10) In **[Default settings]**, enter 1010 (Auto attendant) at **[Forwarding destinations]** and click **[Save]**



6.12. Setting Up Conference Call

Using OnDO PBX you can create conference rooms. In this example, user 2000 is used as the conference number.

6.12.1. Creating a Conference Room

OnDO PBX Admintool > Users

1) Choose the **[Users]** menu.
2) To create extension 2000, enter 2000 in the text entry field, and click the **[Create a New User]** button. User 2000 is created and added.
3) Click the edit button for user 2000, to open the edit window.
4) Select **[Conference]** at **[Administrative settings]** > **[Type of Call Forwarding]** and click the **[Save]** button.

✓ *With the above settings, any user can enter in the conference room by dialing 2000.*

6.12.2. Limiting Members Who Can Enter the Conference Room

You can limit members that join the conference by specifying members (for example 1001, 1002, and 1003) at **[Call forwarding settings]** > **[Applies to (Caller numbers)]**. Doing so, only 1001, 1002, 1003 will be allowed to join the conference. With these settings, neither users 1004 and 1005, nor any other users, will be allowed to join the conference.

6.12.3. Simultaneous Calls to All of the Conference Members

A conference member can convene all members of the conference room at once. For example, set 1001, 1002, 1003 at **[Call forwarding settings]** > **[Forwarding destinations]**. By dialing 2000, all conference members (1001, 1002, and 1003) will be invited simultaneously.

6.12.4. Starting a Conference Call (Alternate Method)

Additional methods for starting a conference call are included in the OnDO PBX User Guide.

6.13. PSTN Access Using a VoIP Gateway

Using a SIP compliant VoIP Gateway, OnDO PBX users can receive calls from Public Switched Telephone Network (PSTN) and make calls to PSTN lines.

6.13.1. Receiving PSTN calls

To receive a PSTN call at an OnDO PBX extension, you do not need any special settings on OnDO PBX. Set the following at your VoIP Gateway:

- SIP proxy address

  Specify the IP address of OnDO SIP Server
- An SIP URI which PSTN calls are directed to

  Specify an OnDO PBX user name (For the example in the section 6.9, 5555).

6.13.2. Calling PSTN numbers

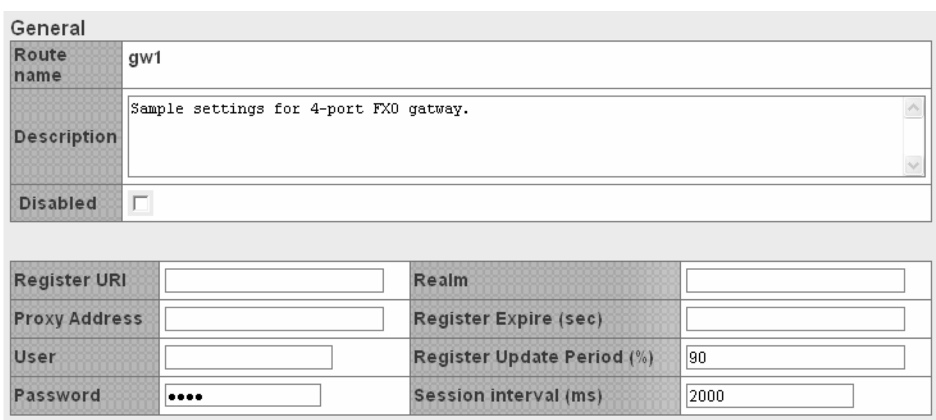
One Stage Dialing

If your VoIP Gateway supports One Stage Dialing, an OnDO PBX user can make a PSTN direct call by setting an ARS rule as follows:

Navigate to OnDO PBX Admintool > ARS > Edit

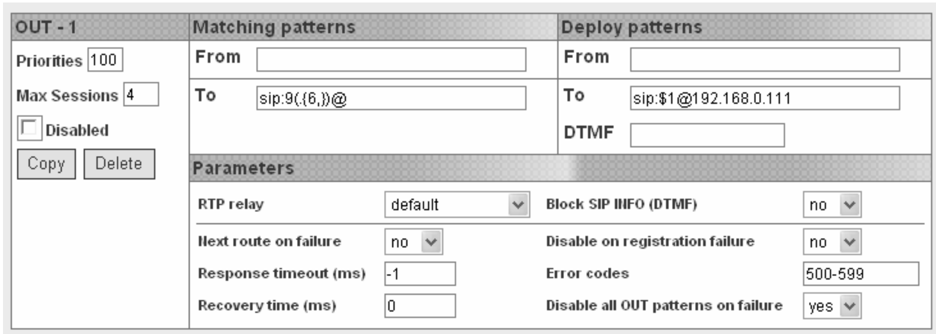
- *General*

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



- *Patterns*

  Create a pattern OUT as follows:



For this example, we will assume the VoIP Gateway's IP address is 192.168.0.111. An OnDO PBX user dials 9-prefix and a PSTN number for calling out to PSTN. For example, to make a call to 401-6636, dial "94016636".

It is useful to set priorities when there are multiple options for making calls, such as when you have multiple VoIP 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.

Two Stage Dialing

If your VoIP Gateway supports Two Stage Dialing, have the gateway's PSTN port register with OnDO SIP Server. Let's suppose the gateway's PSTN port has the SIP user name, 111. To call a PSTN number, OnDO PBX users dial 111 first and dial a PSTN number after hearing a dial tone.

6.14. Connecting with Internet Telephony Service Providers (ITSPs)

6.14.1. 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

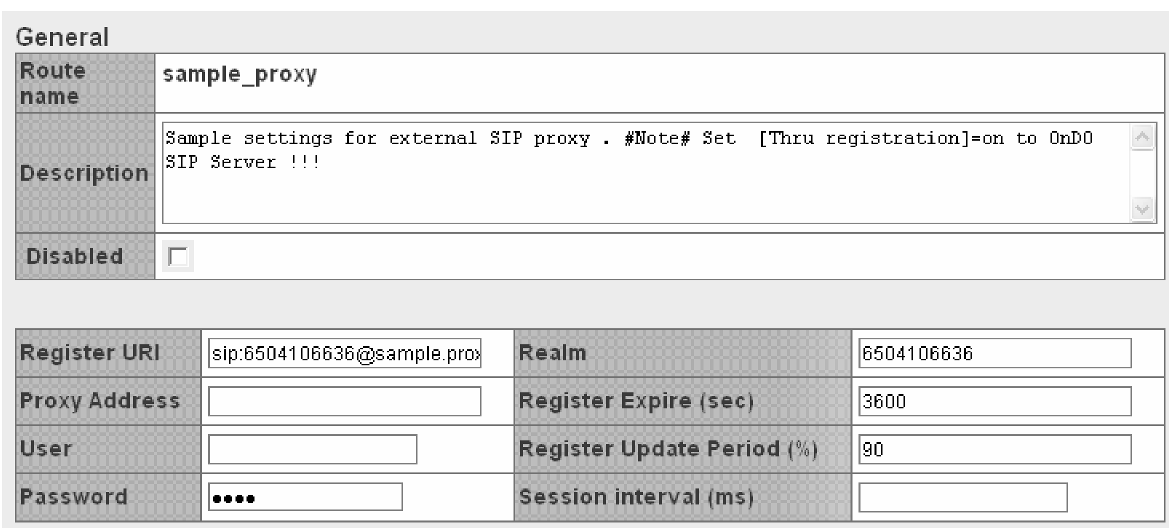
✓ *Depending upon the provider there may be restrictions for connecting to services, such as available information or equipment used to connect. Please contact your VoIP service provider for more details.*

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

6.14.2. Setting ARS for ITSP

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.



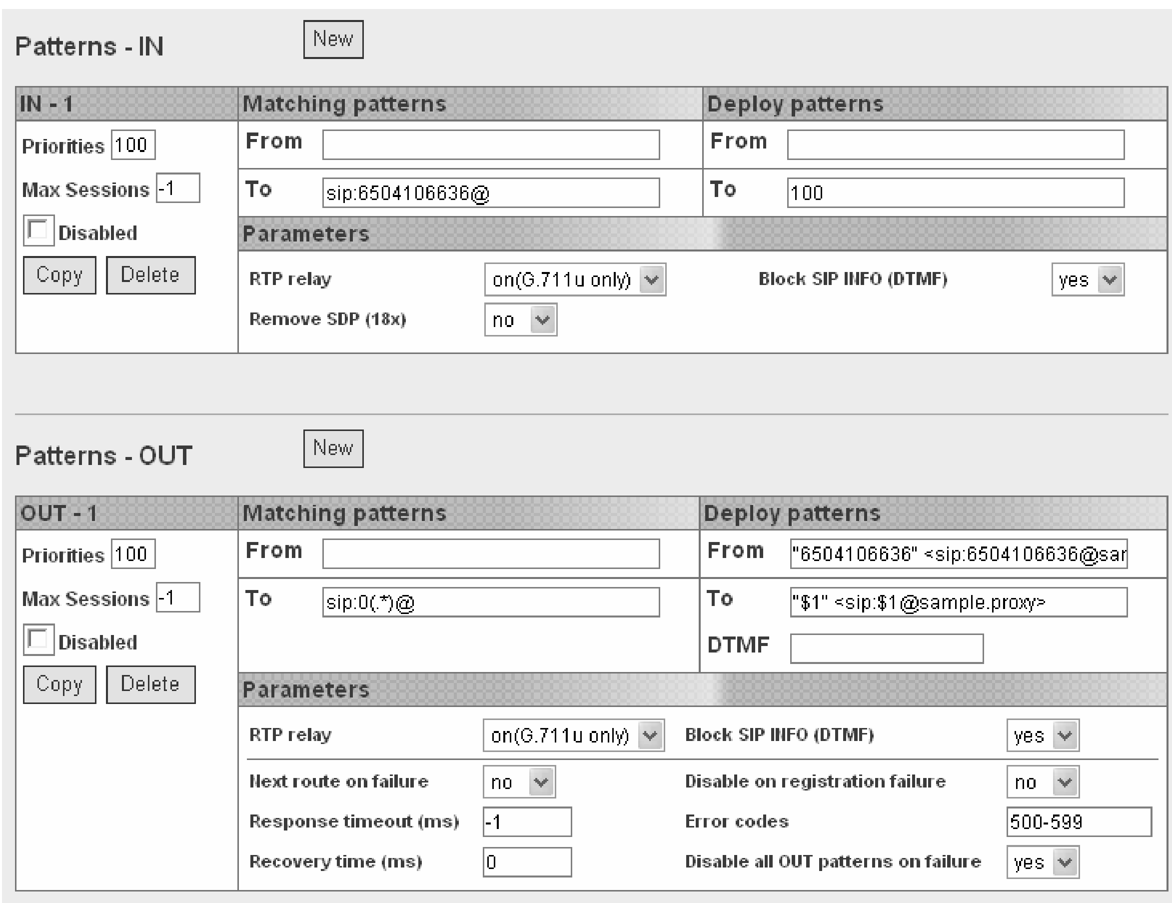
- *Patterns*

  Use these settings to define patterns for when a call is initiated and received through a 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

6.15. ARS Outbound Route Failover

This feature is currently available with the Standard Edition only.

Utilizing the Automatic Route Selection (ARS) outbound route failover feature allows users to create redundant telecommunications systems. If an outbound route is not available or usable, OnDO PBX will failover the session to an alternative route.

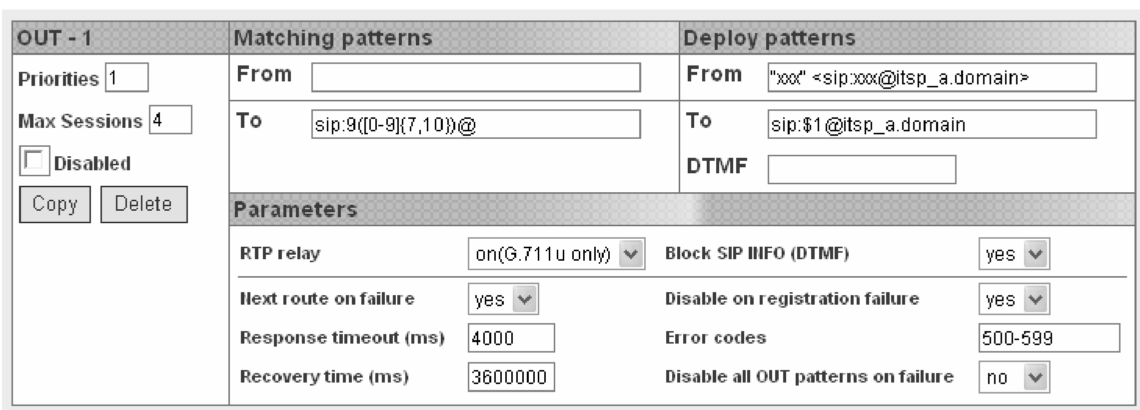
6.15.1. Usage Examples

- OnDO PBX provides automatic failover to alternative ITSP service in the event of failure at your specified ITSP service.
- OnDO PBX provides automatic failover to analog telephone session via PSTN Gateway in the event of failure at your specified ITSP service.
- Creating redundant analog telephone connections with multiple PSTN Gateways

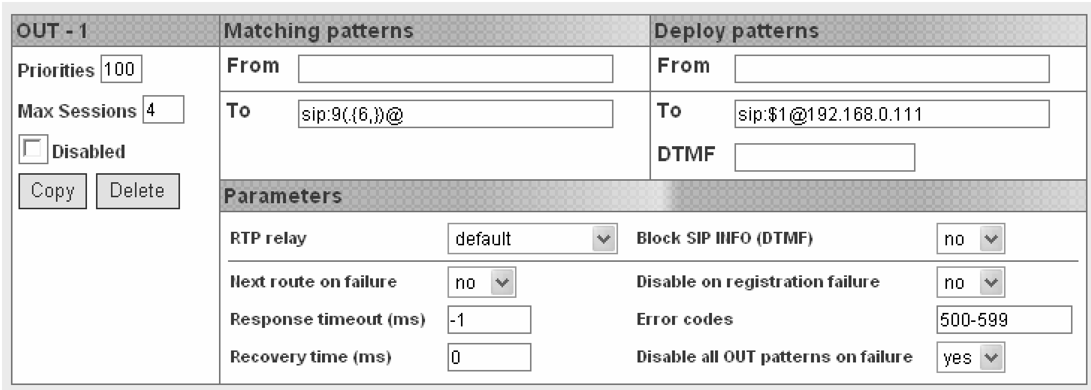
6.15.2. Setting Examples

The ITSP line is set for regular outbound sessions. When the ITSP line fails, the outbound sessions will be routed through PSTN Gateway.

Route name: ITSP_A Patterns (OUT)



Route name: MyGateway Patterns (OUT)



- *Under regular operation, the route with highest priority "ITSP_A" will be used for outbound sessions. In the event that there is no response (longer than 4 seconds) for INVITE messages or "500-599" response was received, OnDO PBX will continue route searching to meet outbound session request. And the next highest prioritized route "MyGateway" will be chosen for the alternative route for the session. Since the recovery time is set for one hour (36,000,000 ms), the session will be routed through "MyGateway" for one hour after the failover. If "ITSP_A" is back on running, the session will be routed through the highest priority route, "ITSP_A".*
- *Response Timeout (ms) should be adjusted according to your environment. For PSTN Gateways and SIP servers which locate in the local network may not require to set long Response Timeout intervals. For the route that require internet connection or delay can be expected the Response Timeout intervals should be set longer.*
- *Setting under Parameters, Disable on registration failure is set for "yes" as default setting. When registration is not working property at "ITSP-A" route, it will be disabled and "MyGateway" route will be used instead.*

7. Setup Items

7.1. Option Menu

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

7.1.1. General Settings

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

7.1.2. SIP Settings

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.

7.1.3. Phone Number Settings

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.)

7.1.4. PBX System Settings

Name	Default value	Description
Port number	15060	The port number that OnDO PBX will use. (SO)
Max concurrent sessions	12	The maximum number of concurrent sessions that OnDO PBX can handle. (SO)
Max number of user agents	40	The maximum number of user agents (SIP UAs) that PBX can handle. (SO)
Min Port	11000	Minimum port number the RTP Protocol uses for sending voice data.
Max Port	11999	Maximum port number the RTP uses for sending voice data.
RTP relay	On (G.711u only)	Specify Media server handles RTP packets or not. This setting will be in effect when there are no RTP relay specifications made in ARS or User settings. “on” – Codec is set for iLBC, G.711 ulaw, or G.711 alaw, and Media Server handles RTP packets. “on(G.711u only) – Codes is set for G.711 ulaw and Media Server handles RTP packets. off (G.711u only) – Codec is set for G.711 ulaw, and Media Server does not handle RTP packets. off – Media Server does not handle RTP packets.
Max concurrent recording sessions	10	Maximum concurrent sessions with call recording
Ringing Timeout (ms)	120000(ms)	Timeout value for awaiting an answer from the dialed party after ringing starts.
Talking Timeout (ms)	259200000 (ms)	The maximum length of time a call can last. Value 0 signifies infinite.
Max hop number	20	Maximum number of SIP Servers or OnDO PBX that a call can go through (hop number).
Days to keep call logs	90	Number of days to keep call logs

Java VM arguments	Blank	Parameters to pass to VM
Session Keep Alive (sec)	600	Interval to send keep-alive packets to UAs during a call when RTP relay = off

7.1.5. 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 concurrent session limit	12	Maximum number of concurrrent sessions for voicemail and IVR feature. (The limit cannot be modified in the SmallOffice Edition.)
Max stored 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.
Message Waiting Indicator	off	Enable (on) /Disable (off) Message Waiting Indicator (Voice mail notification to phones)
Conversation recording length (sec)	600 (sec)	Maximum recording length for each call.
Min Port	12000	Minimum port number the RTP uses for sending voice data.
Max Port	12999	Maximum port number the RTP uses for sending voice data.
Ringing Timeout	120000(ms)	Timeout value for awaiting an answer from the

(ms)		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.
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.
Java VM Arguments	Blank	Parameters to pass to VM

✓ *ms = 0.001 second*

7.1.6. Email Settings

Name	Default value	Description
SMTP server	Blank	The SMTP Server Address for sending email notifications when the user receives a new voicemail message.
SMTP authentication	on	Enable (on) /Disable (off) SMTP authentication setting
POP3 server	Blank	The address of the POP3 server. (for POP-before-SMTP authentication)
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.

7.1.7. Multiline Settings

Name	Default value	Description
External Line (ARS)*	Blank	A special setting for SAXA phones

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

7.2. Call Status

The list of ongoing calls will be displayed under the **[Call Status]** menu. By specifying search criteria, the search result will be displayed on the screen. You can view detailed information for the selected search result. By default, the detailed information for the result on the top will be displayed on the screen.

7.2.1. Status

Name	Description
ID	Call ID
Status	Call Status
Call Park	The number that has been parked
Conference	Conference number
Start	Time the call begin

7.2.2. User Agents

Name	Description
User	User Name
ARS	Used ARS route
URI	SIP URI
Connected	Time the call begin
Disconnect	Disconnect the call. (If the user does not have rights to disconnect, this option will not be displayed.

7.3. 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 VoIP Gateways or PBXs.

To add a new route:

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

To view currently used ARS route:

- Choose [ARS] menu
- Select [Running Status]

7.3.1. 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
Register Expire (sec)	3600	The length of time till REGISTER expires
User	Blank	User ID for authentication account. This field is optional field when authentication is not being used.

Register Update Period (%)		The period of interval till RE-REGISTER occurs
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.

7.3.2. Pattern - IN

Name		Default value	Description
Priorities		100	Lower numbers hold a higher priority.
Max Sessions		-1	Specify the number of sessions (including RINGING and BYE sessions) that are allocated to the route.
Disabled		Off	Disable this pattern
Matching patterns	From	Blank	Specify a matching rule for FROM header using regular expressions. When the field is left blank, all calls will be considered as matched.
	To	Blank	Specify a matching rule for TO header using regular expressions. When the field is left blank, all calls will be considered as matched calls.
Deploy patterns	From	Blank	Specify replace patterns for FROM header using regular expressions.
	To	Blank	Specify replace patterns for TO header using regular expressions.
	DTMF	Blank	For when DTMF needs to be issued after calling gateway (2 stage calling), you can specify the DTMF string using some part of [TO] Matching Pattern.

Parameters	RTP relay	Default	Select RTP relay ON/OFF. If “Default” is selected, [Option] menu > [RTP relay] setting will be used. “on” – Codec is set for iLBC, G.711 ulaw, or G.711 alaw, and Media Server handles RTP packets. “on(G.711u only) – Codes is set for G.711 ulaw and Media Server handles RTP packets. off (G.711u only) – Codec is set for G.711 ulaw, and Media Server does not handle RTP packets. off – Media Server does not handle RTP packets.
	Block SIP INFO (DTMF)	n	Stop or not for passing the DTMF from a user to the other party when OnDO PBX received DTMF
	Remove SDP (18x)	no	Remove SDP or not when 180 Ringing or 183 Session Progress from called party contains SDP.

7.3.3. Patterns – OUT

Name		Default value	Description
Priorities		100	Lower numbers hold a higher priority.
Max Sessions		-1	Specify the number of sessions (including RINGING and BYE sessions) that are allocated to the priority.
Disabled		Off	Disable this pattern
Matching patterns	From	Blank	Specify a matching rule for FROM header using regular expressions. When the field is left blank, all calls will be considered as matched.
	To	Blank	Specify a matching rule for TO header using regular expressions. When the field is left blank, all calls will be considered as matched calls.

Deploy patterns	From	Blank	Specify replace patterns for FROM header using regular expressions.
	To	Blank	Specify replace patterns for TO header using regular expressions.
	DTMF	Blank	For when DTMF needs to be issued after calling gateway (2 stage calling), you can specify the DTMF string using some part of [TO] Matching Pattern.
Parameters	RTP relay	Default	Select RTP relay ON/OFF. If “Default” is selected, [Option] menu > [RTP relay] setting will be used. “on” – Codec is set for iLBC, G.711 ulaw, or G.711 alaw, and Media Server handles RTP packets. “on(G.711u only) – Codes is set for G.711 ulaw and Media Server handles RTP packets. off (G.711u only) – Codec is set for G.711 ulaw, and Media Server does not handle RTP packets. off – Media Server does not handle RTP packets.
	Block SIP INFO (DTMF)	No	Stop or not for passing the DTMF from a user to the other party when OnDO PBX received DTMF
	Remove SDP (18x)	No	Remove SDP or not when 180 Ringing or 183 Session Progress from called party contains SDP.
	Next route on failiure	No	Set failover for outbound sessions or not
	Disable on registration failure	no	Enable (yes)/Disable (no) this Pattern when registration failed
	Response timeout (ms)	-1	The period of time before timeout is activated when response has not been received

	Error codes	500-599	Failover will be activated when specified error codes are received for INVITE requests
	Recovery time (ms)	0	The period of time till this pattern will be reactivated
	Disable all OUT patterns on failure	yes	Disable all OUT patterns when the route using this OUT pattern failed.

7.4. 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 **[Users] > [Edit] > [Call log]**.

7.5. User Settings

OnDO PBX Admintool > Users > Edit

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

7.5.1. 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.

7.5.2. General Settings

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

[iii] These options will be set as the Administrator who created this user.
[iv] These options will be set as the Administrator who created this user.
[v] These options will be set as the Administrator who created this user.

7.5.3. 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.
Pattern Setting	1	Multiple calling patterns can be set by dividing [Forwarding destinations], [Ringer time], [Forwarding destination (No answer/Busy)] settings by slash(es) "/". The pattern you would like to use will be set by specifying the numerical order of pattern from the left.

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

7.5.4. 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.

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

Ringer time (sec)*	20	Ringer 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 not been answered. 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.
Pattern Setting	1	Multiple calling patterns can be set by dividing [Forwarding destinations], [Ringer time], [Forwarding destination (No answer/Busy)] settings by slash(es) "/". The pattern you would like to use will be set by specifying the numerical order of pattern from the left.
Call Pickup group	Blank	Enables one touch Call Pickup by assigning the user to a preset Call Pickup group.

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

7.5.5. Call Forwarding Settings

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

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.

Pattern Setting	1	Multiple calling patterns can be set by dividing [Forwarding destinations], [Ringer time], [Forwarding destination (No answer/Busy)] settings by slash(es) "/". The pattern you would like to use will be set by specifying the numerical order of pattern from the left.

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

7.5.6. Call Forwarding Settings

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

Name	Default value	Description
Forwarding destinations*	Blank	By specifying phone number(s) or SIP URI(s), a user can invite multiple users to this conference by dialing this extension.
Applies to (Caller numbers) *	*	Phone number(s) that can join this conference. An asterisk (*) and a questions mark (?) can be used for matching metacharacters. An asterisk (*) means zero (0) or more characters and a question mark (?) means one character.
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.
Pattern Setting	1	Multiple calling patterns can be set by dividing [Forwarding destinations], [Ringer time], [Forwarding destination (No answer/Busy)] settings by slash(es) "/". The pattern you would like to use will be set by specifying the numerical order of pattern from the left.

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

7.5.7. Voicemail Settings

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).*

7.5.8. Administrative Settings (SA)

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, Conference
IVR	None	Setting for the usage of the Interactive Voice Response (IVR) system. The following options are available: **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 **Add/Remove Forwarding Destinations**: By calling this extension, caller will be added/removed to/from the Forwarding Destinations of the users set in [Add/Remove forwarding destinations] > [Target users]. **Switch Patterns**: Caller's [Pattern Setting] will be changed **None**: No IVR service

- ✓ *(SA)This menu is only available to the system administrator*

7.5.9. PBX Settings (SA)

Name	Default value	Description
RTP relay	default	Specify Media server handles RTP packets or not. “on”: this user can use iLBC or G.711 ulaw or G.711 alaw. Media server will relay RTP packets. “on (G.711u only)”: Only G.711 ulaw can be used by this user. Media server will relay RTP packets. “off (G.711u only)”: Only G.711 ulaw can be used by this user. Media server will not relay RTP. “off”: this user can use Any codec. Media server will not relay RTP relay. “default”: RTP relay setting in ARS will be valid.
Call Recording	off	“on”: records all calls of this user. “off”: doesn’t record all calls of this user. You can stop or start recording when initiating a call or during a call. Please refer to OnDO PBX User’s guide.
Max sessions	unlimited	Specify the maximum received session numbers for the user.
Join other user’s conversation	yes	Allow (yes) or not (no) this user to join other user’s conversation.
Accept other users to join my conversation	yes	Allow other users join this user’s conversation.

- ✓ *(SA)This menu is only available to the system administrators*

7.5.10. Auto Attendant Settings (SA) – [Administrative settings]-[IVR]

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 has not 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.
Transfer to unregistered users	disable	Enables/disables call transfers to an unregistered user. Options: disable/enable

- ✓ ** This category allows multiple entries (divided with commas).*
- ✓ *(SA)This menu is only available to the system administrators.*

7.5.11. Add/Remove Forwarding Destinations (SA) - [Administrative settings]-[IVR]

Button Name		Description
Target users*	Blank	By calling this extension, caller's extension number will be added/deleted from [Forwarding destinations] of the users set in this [Target users] field.

- ✓ ** This category allows multiple entries (divided with commas).*
- ✓ *(SA)This menu is only available to the system administrators.*

7.5.12. Switch Patterns - [Administrative settings]-[IVR]

Button Name		Description
Pattern Number	2	By calling this extension, caller's [Pattern Setting] value will be changed to the value set here

7.5.13. 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]**.

7.5.14. 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. (For example, 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.*

7.6. Notes for Sound Files

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

8. Uninstall (Windows)

This topic will assist you with uninstalling the OnDO PBX software from your computer with a Windows operating system.

Navigate to **[Start]>[Program]>[Brekeke]>[OnDO PBX]>[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.*

9. Uninstall (Red Hat Linux and Solaris)

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