Connecting with *SIPphone.com* service

SIPphone.com (http://www.sipphone.com/) is a VoIP service provider that is operated by SIPphone, Inc. in California, U.S.A. This document will explain how to utilize OnDO PBX with *SIPphone.com*.

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1. Setting up *SIPphone.com* account

Please obtain an account from *SIPphone* website

***SIPphone.com* Telephone number**	1747#######
Password	******

2. ARS Setting

To define ARS settings for your *SIPphone.com* account:

1) Choose **[ARS]** in the OnDO PBX Admintool.
2) Type a route name in entry field (we suggest *SIPphone.com*)
3) Click **[Create a new route]**. Your new route will appear in the list above.
4) Click the **[Edit]** button next to your new route to define conditions.

For more information on ARS settings, please refer to the OnDO PBX Administrator's Guide.

Please see the following sections of this document for ARS settings relevant to *SIPphone.com* service.

2.1. General Settings

Register URI	sip:1747#######@proxy01.sipphone.com
Realm	
Proxy Address	
User	1747#######
Password	******

In the **[Register URI]** field, enter your *SIPphone.com* phone number in the format shown above. Enter your User No and password as assigned by *SIPphone.com* in the appropriate fields.

2.1.1. Incoming Call Pattern

Direction	IN
Matching patterns: From	
Matching patterns: To	sip:1747#######@
Deploy patterns: From	
Deploy patterns: To	OnDO PBX Extension
Ignore Priorities smaller than No.	1
Deploy patterns: DTMF	
RTP relay	on(G.711u only)

To direct incoming calls, enter your *SIPphone.com* number in the **[Matching patterns: To]** field, in the format shown above. Call received by your *SIPphone.com* number will be directed to the extension set in the **[Deploy patterns: To]** field.

2.1.2. Outgoing Call Patterns

Direction	OUT
Matching patterns: From	
Matching patterns: To	sip:1747(.+)@
Deploy patterns: From	"1747#######"<sip:1747#######@proxy01.sipphone.com>
Deploy patterns: To	sip:1747$1@proxy01.sipphone.com
Ignore Priorities small than No.	1
Deploy patterns: DTMF	
RTP relay	on

This rule defines "1747" as the prefix for making calls through your *SIPphone.com* account. Outgoing calls dialed with the prefix "1747" will be routed through your *SIPphone.com* account. Note that this number may be different depending on your *SIPphone.com* number.

2.2. Priorities

Direction	IN/OUT
Max sessions	-1
Priorities	100

Here, all priority settings have been left at their respective default values. To give this rule a higher priority, set its Priority to a number with a lesser value. In the **[Max sessions]** field, -1 indicates an unlimited amount.

3. OnDO SIP Server Dial Plan Settings

Dial Plan can accommodate the unique requirements of each VoIP service provider.

Receiving calls to a *SIPphone.com* number

Matching Patterns	Deploy Patterns
`$addr= 198\.65\.166\.131` `$request=^INVITE`	`$target=localhost:15060` `To=%1`

to=(.*)	$auth=off

✓ *This Dial Plan setting must be placed above "to PBX" (default dial plan).*

4. OnDO SIP Server Config Settings

Under the Configuration settings choose System and in the Network section put the Internet side address of your Broadband router. This settings is necessary when OnDO SIP Server is behind a router

Interface address 1	197.166.8.99 (example)

- *If your OnDO SIP Server is behind a router please set the port forwarding as described below:*

First, you will need to set port forwarding at the router located between the global network and your local network to forward packets with specific destination port numbers to the OnDO SIP Server. The ports you need to forward can be as follows:

Config > SIP [(General)] > SIP exchanger – Local Port [UDP]
Default value: 5060
Config > RTP > RTP exchanger – From Minimum Port to Maximum Port [UDP]
Default value: 10000-10999

5. Notes for Dial Plan setting

♦ Here we did not specify any priority settings. Please set your priorities according to your needs, remembering that lower number hold a higher priority.

♦ This Dial Plan allows calls from the *sipphone.com* line to pass through without authentication from OnDO SIP Server. When INVITE authentication is tuned off at [Config] menu in OnDO SIP Server, this Dial Plan setting can be omitted.

♦ The Dial Plan for receiving calls includes the *SIPphone.com*'s SIP server's IP address. when *SIPphone.com* changes their SIP server IP address, you will need to you update your Dial Plan.

♦ In this tutorial guide, [RTP relay] is set for "on" in ARS deploy pattern setting. In some cases, setting [RTP relay] = "off" may work; however, we have not tested all of the applicable environments.