

Connecting with *Free IP Call*

Free IP Call (<http://www.freeipcall.com/>) offers telephone service using the VoIP standard SIP. The service allow users making/receiving VoIP calls to/from VoIP telephone numbers and making VoIP calls to PSTN telephone numbers. This document will explain how to utilize OnDO PBX with *Free IP Call*.

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1. Setting up a *Free IP Call* account

Please obtain an account from *Free IP Call* website.

PSTN telephone number	#####
Server Name	sip.freeipcall.com
User ID/number	1311112222 (example)
Password	*****

2. ARS Configuration for *Free IP Call*

To define ARS settings for your *Free IP Call* account:

- 1) Choose **[ARS]** in the OnDO PBX Admintool.
- 2) Type a route name in the entry field (we suggest *Free IP Call*)
- 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 *Free IP Call*.

2.1. General Settings

Register URI	sip:1311112222@sip.freeipcall.com
Realm	
Proxy Address	freeipcall.com
User ID/Number	1311112222
Password	*****

In the **[Register URI]** field, enter your *Free IP Call* user ID/number and the server name in the format shown above. Enter the server/proxy address, your User ID/number and password as assigned by *Free IP Call* in the appropriate fields.

2.2. Pattern Settings

2.2.1. Incoming call pattern

Direction	IN
Matching patterns: From	
Matching patterns: To	sip:1311112222@
Deploy patterns: From	
Deploy patterns: To	OnDO PBX extension number
Ignore Priorities smaller than No.	1
Deploy patterns: DTMF	
RTP relay	on(G.711u only)

To direct incoming calls from PSTN callers or VoIP callers, enter your user ID from *Free IP Call* in the **[Matching patterns: To]** field, in the format shown above. All calls terminated on your user ID/number will be directed to the extension set in the **[Deploy patterns: To]** field.

2.2.2. Outgoing call pattern

Direction	OUT
Matching patterns: From	
Matching patterns: To	sip:9(.+)
Deploy patterns: From	"1311112222"<sip:1311112222@sip.freeipcall.com>
Deploy patterns: To	sip:\$1@sip.freeipcall.com
Ignore Priorities small than No.	1
Deploy patterns: DTMF	
RTP relay	on(G.711u only)

This rule handles calls made by OnDO PBX users via *Free IP Call*: dialing 9 first to call places outside of the PBX. In the **[Matching patterns: To]** field, "9" represents the dialing prefix, and (.+) represents any number of dialing digits. The **[Deploy patterns: To]** will automatically remove the prefix, 9, from all outgoing calls.

2.3. 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 *Free IP Call* number

Matching Patterns	Deploy Patterns
\$addr=^82\.97\.10\.20 \$request=^INVITE to=(.*)	\$target=localhost:15060 To=%1 \$auth=off

✓ *This Dial Plan needs to be placed before "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)
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- *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
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5. Notes for Dial Plan settings

- ◆ This Dial Plan allows calls from the *Free IP Call* line to pass through without authentication from OnDO SIP Server. When INVITE authentication is turned off at [Config] menu in OnDO SIP Server, this Dial Plan setting can be omitted.
- ◆ Note that you may need to update Dial Plan rules that include *Free IP Call's* SIP Server IP address. *Free IP Call* may change their Server's IP address at any time.
- ◆ In this tutorial guide, [RTP relay] is set for "on(G.711u only)" in ARS deploy pattern setting. In some cases, setting [RTP relay] = "off" may work; however, we have not tested all of the applicable environments.