Connecting with *IPKall*

IPKall (http://www.ipkall.com/) offers telephone service using the VoIP standard SIP. The service is a unique VoIP service which provides users with PSTN phone numbers in Washington State in United States. This document will explain how to utilize OnDO PBX with *IPKall* service. (*IPKall* requires an active SIP proxy address and SIP number to receive a phone number.) With our example in this document, we used Free World Dialup (FWD) information as the proxy and user ID. Please visit *IPKall* website for more information).

1. Setting up an *IPKall* account

Please obtain an account from *IPKall* website.

PSTN telephone number	1 (360) xxx-xxxx
Server Name	fwd.pulver.com
User ID/number	123456 (example)
Password	******

✓ *IPKall needs a valid email address for account information to receive a phone number, also IPKall # that has been inactive (0 usage) for 30 days will be recycled back to the available pool.*

2. ARS Configuration for *IPKall*

To define ARS settings for your *IPKall* account:

1) Choose **[ARS]** in the OnDO PBX Admintool.
2) Type a route name in the entry field (we suggest *IPKall*)
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 *IPKall* service.

2.1. General Settings

Register URI	sip:123456@fwd.pulver.com
Realm	
Proxy Address	fwd.pulver.com
User ID/Number	123456
Password	******

In the **[Register URI]** field, enter your *IPKall* 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 *IPKall* in the appropriate fields.

2.2. Pattern Settings

2.2.1. **Incoming call pattern**

Direction	IN
Matching patterns: From	
Matching patterns: To	sip:123456@
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 *IPKall* 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.

2.3. 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 an *IPKall* number

Matching Patterns	Deploy Patterns

`$addr=^69\.90\.155\.70`
`$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)

- *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 settings*

- *This Dial Plan allows calls from the* IPKall *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* FWD's *SIP Server IP address.* FWD *may change their Server's IP address at any time.*
- *In this tutorial guide, [RTP relay] is set for "on(G.711 u 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.*