Connecting with *VoIP USER*

VoIP USER (http://www.voipuser.org/) offers a community funded telephone service using the VoIP standard SIP. Additionally, community members can request free UK phone numbers to be terminated both to PSTN in most countries over the world or to any SIP/IAX destination. Revenue from these UK number allocations, together with discretionary contributions by members, funds the community outbound service via VoIP User's PSTN gateways.

1. SETTING UP A *VOIP USER* ACCOUNT 2

2. ARS CONFIGURATION FOR *VOIP USER* 2

2.1. General Settings 3

2.2. Pattern Settings 3

2.2.1. Incoming call pattern 3

2.2.2. Outgoing call pattern 4

2.3. Priorities 4

3. ONDO SIP SERVER DIAL PLAN SETTINGS 5

4. ONDO SIP SERVER CONFIG SETTINGS 5

5. NOTES FOR DIAL PLAN SETTINGS 6

1. Setting up a *VoIP USER* account

Please obtain an account from *VoIP USER* website.

PSTN telephone number	######
Server Name	VoIPUSER.org
User ID/number	JohnDoe (example)
Password	******

2. ARS Configuration for *VoIP USER*

1) To define ARS settings for your *VoIP USER* account: Choose **[ARS]** in the OnDO PBX Admintool.
2) Type a route name in the entry field (we suggest *VoIP USER*)
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 *VoIP USER* .

2.1. General Settings

Register URI	sip:JohnDoe@sip.voipuser.org
Realm	
Proxy Address	sip.voipuser.org
User ID/Number	JohnDoe
Password	******

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

2.2. Pattern Settings

2.2.1. Incoming call pattern

Direction	IN
Matching patterns: From	
Matching patterns: To	sip:JohnDoe@
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 *VoIP USER* 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	"JohnDoe"<sip:JohnDoe@sip.voipuser.org>
Deploy patterns: To	sip:$1@sip.voipuser.org
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 *VoIP USER*: 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 *VoIP USER* number

Matching Patterns	Deploy Patterns

`$addr=^216\.127\.66\.119`
`$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 *VoIP USER* 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 *VoIP USER's* SIP Server IP address. *VoIP USER* 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.