

Connecting with IP-Com

IP-Com (<http://www.ip-com.co.uk/>) offers telephone service using the VoIP standard SIP. This service allows users to make or receive VoIP calls to or from VoIP telephone numbers, and to make VoIP calls to PSTN telephone numbers. This document explains how to use OnDO PBX with IP-Com.

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1. Setting Up an IP-Com Account

After signing up for an IP-Com account, you will have your own telephone number, user id, password, and SIP proxy server information of the account you have just set up. Throughout this document, the following demo account will be used to guide you through setting up your IP-Com ITSP account to work with OnDO PBX and SIP server (OSS).

Telephone number	123456
UserID	123456
Password	*****
SIP Proxy server	sip.ip-com.co.uk

2. ARS Configuration

Most of the call management rules for outgoing and incoming calls through outside lines are set at ARS (Automatic Route Selection). Defining rules enables ARS to make optimal use of outside lines and your PSTN gateways.

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

To define ARS settings for your account:

- 1) Choose **[ARS]** in the PBX Admintool.
- 2) Type a route name in the entry field.
- 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.

Please see the following sections of this document for ARS settings relevant to IP-Com's ITSP service.

ARS Settings
Brekeke
Current user: sa

General

Route name	IP-Com		
Description	IP-Com		
Disabled	<input type="checkbox"/>		

Register URI	sip:123456@sip.ip-com.co.uk	User	123456
Realm		Password	xxxx
Proxy Address	sip.ip-com.co.uk	Session interval (ms)	

Patterns new

	Matching patterns	Deploy patterns
	From <input type="text"/>	From <input type="text"/>
	To sip:123456@	To 2000
IN <input type="button" value="Delete"/>	Ignore Priorities smaller than llo. 1	DTMF <input type="text"/>
		RTP relay <input type="text" value="default"/> Block SDP <input type="text" value="no"/> Block dtmf-relay <input type="text" value="no"/>

	Matching patterns	Deploy patterns
	From <input type="text"/>	From "123456" <sip:123456@sip.ip-com.co.uk>
	To sip:9(.7.)@	To sip:\$1@sip.ip-com.co.uk
OUT <input type="button" value="Delete"/>	Ignore Priorities smaller than llo. 1	DTMF <input type="text"/>
		RTP relay <input type="text" value="default"/> Block SDP <input type="text" value="no"/> Block dtmf-relay <input type="text" value="no"/>

Priorities new

No.	Direction	Max Sessions (-1=unlimited)	Priorities	
1	IN/OUT	-1	100	<input type="button" value="Delete"/>

Your changes will be in effect after restart.

2.1. General Settings

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

Register URI	sip:123456@sip.ip-com.co.uk
Realm	
Proxy Address	sip.ip-com.co.uk
User ID/Number	123456
Password	*****

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	2000
Ignore Priorities smaller than No.	1
Deploy patterns: DTMF	
RTP relay	on(G.711 u only)

To direct incoming calls from PSTN callers or VoIP callers, enter your user ID 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(.{7,})@
Deploy patterns: From	"123456" <sip:123456@sip.ip-com.co.uk>
Deploy patterns: To	sip:\$1@sip.ip-com.co.uk
Ignore Priorities small than No.	1
Deploy patterns: DTMF	
RTP relay	on(G.711 u only)

This rule handles calls made by OnDO PBX users via IP-Com's ITSP service: dialing 9 first to calls placed 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

In this example, 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.

For outbound calls:

Matching Patterns	Deploy Patterns
<pre>\$request=^INVITE to=(sip\.ip-com\.co\.uk) \$port=15062</pre>	<pre>\$auth=off</pre>

✓ *This Dial Plan needs to be placed before "to PBX1" (default dial plan).*

For inbound calls:

Matching Patterns	Deploy Patterns
<pre>\$addr=^212\.112\.146\.147 \$request=^INVITE to=(.*)</pre>	<pre>\$target=localhost:15060 To=%1 \$auth=off</pre>

✓ *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
```

5. Notes for Dial Plan Settings

- ◆ This Dial Plan allows calls from the IP-Com 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 IP-Com's SIP Server IP address. IP-Com 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.