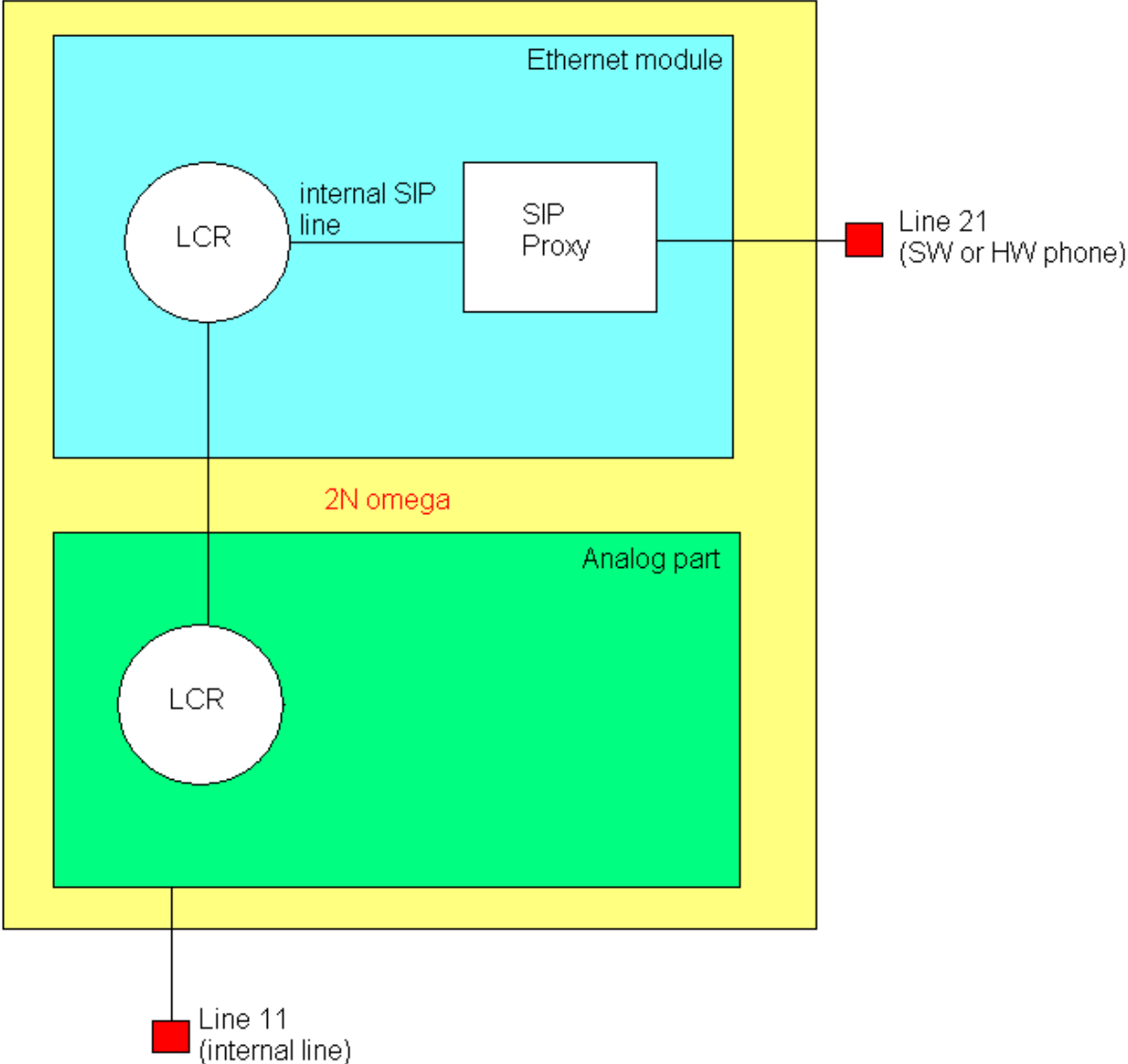


Guide through setup of VoIP in Omega PBX

Setup procedure :

- 1) User create
- 2) SIP phone registration
- 3) Ars setup
- 4) LCR setup
- 5) SIP proxy setup



Before we start with configuration it's necessarily to discover IP address of the VoIP module. To do it, simply connect to module by serial line and run Hyperterminal.

```
PBX VoIP module Lite V2.3.0beta1Network Configuration Menu VoIP
modul
Option Value Description
1 - Address [ 192.168.50.60 ] - Internet address
2 - Network Mask [ 255.255.255.0 ] - Internet subnet mask
3 - Gateway [ 0.0.0.0 ] - Internet default gateway
4 - Routing [ menu ] - IP routing table configuration
5 - Dns1 [ 0.0.0.0 ] - DNS server 1
6 - Dns2 [ 0.0.0.0 ] - DNS server 2
7 - Host name [ "VoIP_modul" ] - Host name
8 - Domain [ "" ] - Domain name
9 - Dhcp [ off ] - Use DHCP on startup
01 - HW Addr [ 00:50:C2:81:F0:A2 ] - Ethernet HW Address
Enter an option number, <ESC> previous menu
>_
```

1)Creating user:

Now you can Connect to VoIP module by using internet browser. As an address fill in an IP **address of VoIP module**.

Now create new user. Fill in **name, password and number of line**. These parameters must be the **same as fill in connected IP phone**.

User name:	<input type="text"/>
New password:	<input type="text"/>
Confirm new password:	<input type="text"/>
Group:	Administrators <input type="button" value="v"/>
Language:	Slovensky <input type="button" value="v"/>
Default application:	User management <input type="button" value="v"/>
Rights:	<input checked="" type="checkbox"/> USERS+LINES+LCR
Rights denied:	<input checked="" type="checkbox"/> USERS+LINES+LCR
Line number:	<input type="text"/>

(USER COUNT THAT IS ABLE TO CREATE IS RESTRICTED BY LICENSE!!!!)

2)IP phone setup

We chose SJphone as IP phone for demonstration.

Create new profile, fill in name and chose **Call through SIP proxy**.

Profile name:	<input type="text"/>
File name:	<input type="text"/>
Profile type:	Calls through SIP Proxy <input type="button" value="v"/>

Mark all caller ID boxes as shown in picture.

SIP Registration	Advanced	DTMF	STUN
Profile Options	Initialization	SIP Proxy	
User data:	Inquired	Saved	Required
Account:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
Password:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
Caller ID:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
Full Address of Record:	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Fill in **address of SIP proxy** into Domain/Realm box. It's an **address of VoIP module** you discovered earlier.

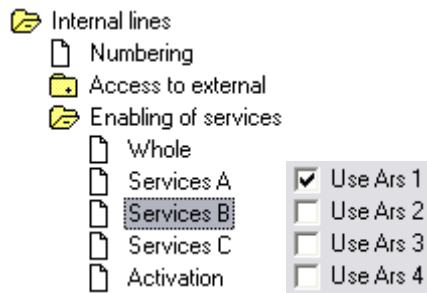
SIP Registration	Advanced	DTMF	STUN
Profile Options	Initialization	SIP Proxy	
Domain/Realm:	<input type="text" value="192.168.50.60"/>		

After confirm fill in account name password and caller ID. This values should be the same.

Account:	<input type="text" value="22"/>
Password:	<input type="password" value="••"/>
Caller ID:	<input type="text" value="22"/>

3) Ars setup:

Setup Ars for used lines, as shown on figure.

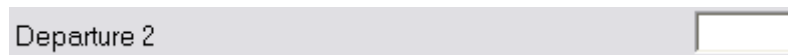


(You have to do these settings for all lines, that you want to make calls to VoIP)

Now set up prefix used in Ars as shown. In our example **prefix is 2 to direction to VoIP**.



It's **necessarily to delete DDI to departure 2** to prevent collisions with Ars settings.



Set up rights to Access outgoing trunks and mark choice to use saving automat.

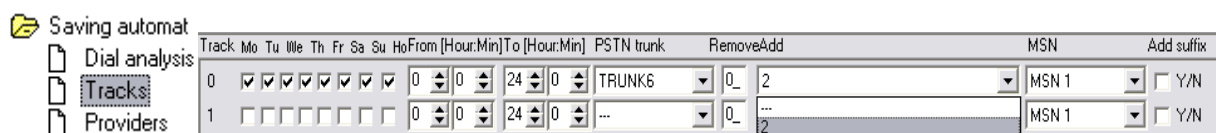


4) LCR setup:

Create new provider with number **2**.



Creating of new provider enables you to add number **2** in following card as shown on figure below.



Set up ARS to be the same as in previous steps.



Fill in a **question mark** in box as shown – that means ANY number. As you can see by using provider number **2** will be addend instead deleted prefix **2** in Ars.

Now you have to set up LCR in VoIP module. Default values are prefix **1** to direction to Omega and prefix **2** to direction to VoIP.



Destination		Prefix	Enabled	Routes
TOOMEGA	<input type="checkbox"/>	1	<input checked="" type="checkbox"/>	OMEGA
TOVOIP	<input type="checkbox"/>	2	<input checked="" type="checkbox"/>	VOIP

5) SIP proxy setup:

Add way at SIP proxy and set up **connection to LCR**, when prefix is **1**

If prefix:	<input type="text" value="sip:1"/>
Strip:	<input type="text" value="0"/>
Add:	<input type="text"/>
Do action:	<input type="text" value="connect to LCR"/> ▼
With parameter:	<input type="text" value="SIP - Moje SIP linka"/> ▼

6)Creating SIP line

By pass through previous point you created a SIP line as shown on figure that connects LCR and SIP proxy parts at the VoIP module.

If prefix	Strip	Add	Do action	With parameter			
sip:1	0		connect to LCR	SIP - Moje SIP linka	