How to set up calls from OmegaLite VoIP module to GSM/PSTN

Presumptions in this tutorial:

- 1. You want to make a call to GSM/PSTN number over the GSM module or CO line in the basic part of OmegaLite/Omega48.
- 2. GSM numbers are in form of 6xxxxxxx or 7xxxxxxx. PSTN numbers could have any other prefix.
- 3. If you want to dial to GSM/PSTN, you must dial **additional prefix 0** (selection of the external call).

ATTENTION: You have to replace prefixes 6 and 7 in the following examples with GSM prefixes appropriate to your country. Prefix 0, introduced in this example, is stripped in the Normalization – not sent to the network.

If you are not familiar with Normalization, read first the following explanation!!!

Normalization means changing the prefix of the called or calling (caller's) number. This figure shows how it works.



It could be done before the LCR (incoming point) or after the LCR (outgoing point). If it is done before LCR, number will be changed i.e. normalized and after that sent to the LCR table. If we do the normalization after the LCR, number is sent to the LCR table in original form, appropriate route is found and change is done at the exit (outgoing) point.

You can select one of four options, according to the number you want to change and point you want to do it:

- 1. Called outgoing (the most often used)
- 2. Called incoming
- 3. Caller outgoing
- 4. Caller incoming

If you select **Any** in the *Line* field, defined normalization will be applied to any line if specified prefix is matched. If you specify a certain line, normalization will be done only if the call is incoming/outgoing from/to the specified line.

Attention!!!

Digit(s) that you put in the *Prefix* field will be automatically stripped (removed). Number in the *Remove count* field means: how many digits AFTER THE PREFIX will be removed too. Digits in the *Add number* field will be added to the number, after removing prefix (and additional digits), from the LEFT i.e. at the head of the number.

Example:

You call from 111 to 0603555666 and you want to sent it over the route ToPBX (contains only one line named PBX). You want to change the called number to 55603555666 in the case of prefix 06. This change would apply only if the call is routed to the line PBX. At the same time you need leading 0 to select ToPBX route in the LCR table. Yoy have to set:

| Line: PBX | valid for calls routed to line PBX only (see the Type setting too) |
|------------------------|--------------------------------------------------------------------|
| Prefix: 06 | digits 0 and 6 will be removed |
| Remove count: 0 | no digits after the leading 06 will be removed |
| Add number: 556 | add 55 and "returns" 6 |
| | |

Type: Called outgoing

Setup procedure:

VoIP module setup:

- 1) SIP proxy setup
- 2) LCR setup
- 3) Normalization setup

PBX setup:

- 4) Virtual line setup
- 5) Ringing tab DDI setup
- 6) Transit call identification

VoIP module setup:

1) Sip proxy settings

Set up routing to LCR if prefix is **0** at the SIP proxy table.

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

Strip 0 means: no digit is stripped from the original number.

2) LCR setup

Now set prefixes **06** and **07** for GSM network and **0** for Access to PSTN at LCR settings. Prefixes are not removed by this checking.

| Destination | Prefix | Enabled | Routes |
|-------------|--------|----------|--------|
| TOOMEGA | 1 | | OMEGA |
| TOVOIP | 2 | | VOIP |
| To Omega | 06; 07 | Z | OMEGA |
| Do Omegy | 0 | V | OMEGA |

Prefixes in this table are checked in order from the top of the table to the bottom. Therefore it's necessary to have prefix **0** on the very bottom of the table. If not, prefixes **06** and **07** will never be routed as they should be.

3) Normalization setup

Using normalization replace dialed prefixes **06**, **07** and **0** by auxiliary prefixes that will be used for routing in the basic part of OmegaLite.

| Line | Prefix | Remove count | Add number | Туре |
|------|--------|--------------|------------|-----------------|
| | 06 | | 996 | Called outgoing |
| | 07 | | 997 | Called outgoing |
| | 0 | | 98 | Called outgoing |

As seen on the picture, prefixes 06 and 07 are replaced by prefixes 996 and 997 (it's necessarily to ad 996 or 997 instead of just 99, because prefixes 06 and 07 will be removed automatically). 6 and 7 belongs to the "usefull part of number". That's why we have to add them back.

PBX setup:

4) Creating virtual lines

First of all, create virtual lines, one for VoIP to GSM calls and one for VoIP to PSTN calls.



Now, setup at least one authorization rule for those lines.



Do not mark choice to use saving automat in this scenario!

Now assign trunk that will be used to leave PBX.

First check assignment of lines to trunks in the **Trunks** setting of External calls menu In this example, **TRUNK4** contains GSM modules and **TRUNK1** contains CO lines.

| GSM call volume levels | | | | Trun | (S | | | |
|------------------------------|---|--------|--------|--------|--------|--------|--------|---|
| AUX Groups | | TRUNK1 | TRUNK2 | TRUNK3 | TRUNK4 | TRUNK5 | TRUNK6 | * |
| System lines | 1 | CO1 | | | GSM1 | | VoIP1 | Ξ |
| Virtual lines External lines | 2 | CO2 | | | GSM2 | | VoIP2 | - |
| Line types | 3 | | | | GSM3 | | | |
| Types of digital lines | 4 | | | | | | | |
| | 5 | | | | | | | |
| Ringing | 6 | | | | | | | |
| Ringing tables | 7 | | | | | | | _ |

We'll define that TRUNK4 is the default trunk for line 51 and TRUNK1 is the default trunk for line 52.

| 🗁 Virtual lines | | |
|-------------------------------|------|------------|
| 🗋 Numbering | | |
| May direct dial to trunks | Line | Trunk PSTN |
| | | |
| | 51 | TRUNK4 💌 |
| Assigning of trunks and prior | | |
| Mask for dial analysis | 52 | TRUNK1 - |
| Service and private MSN | 1 02 | J |

5) DDI to ringing tables:

Set up DDI to ringing tables. Prefixes **99** and **98** that we introduced in Normalization will be used to select the appropriate Ringing table. After that they will be **automatically** removed. Create New ringing tables Tab.3 and Tab.4.

| Թ Numberina | - | DDI | 🔆 Day | 💽 Night |
|-------------------|---|-----|---------|---------|
| D Internal lines | 1 | 99 | Tab.3 💌 | Tab.3 💌 |
| DDI to ring table | 2 | 98 | Tab.4 💌 | Tab.4 💌 |

For the basic part of OmegaLite both VoIP module and GSM/CO are external lines. So, this is the call between two external lines – transit call. In ringing tables we use action **Transit DDI** to make this type of call. Tables **3** and **4** contains transit DDI to virtual lines. It means that the call with prefix 99, will activate actions defined in Tab.3. This call is routed to the default trunk of the virtual line 51, i.e. to the TRUNK4. The same logic used for the call with prefix 98.

| | Action | Par. |
|---|-------------|------|
| 1 | Transit DDI | 51 |
| 2 | Hang up | |

| | Action | Par. |
|---|-------------|------|
| 1 | Transit DDI | 52 |
| 2 | Hang up | |

6) Transit call identification:

In the basic part of OmegaLite, all outgoing calls must be assigned to a certain extension. That data is used in the CDR of that call (data.acc file). It is clear when call is made from analogue extension, but when we have transit call it couldn't be assigned to the other external line. To resolve this situation use **TransitCall indentification** setting in the External lines menu.

In VoIP field will be put the number of (virtual) extension that will be "responsible" for calls COMMING FROM VoIP lines. Here Virt.1 is the virtual line on the first position i.e. 51.



The same logic is applied for the other external lines.