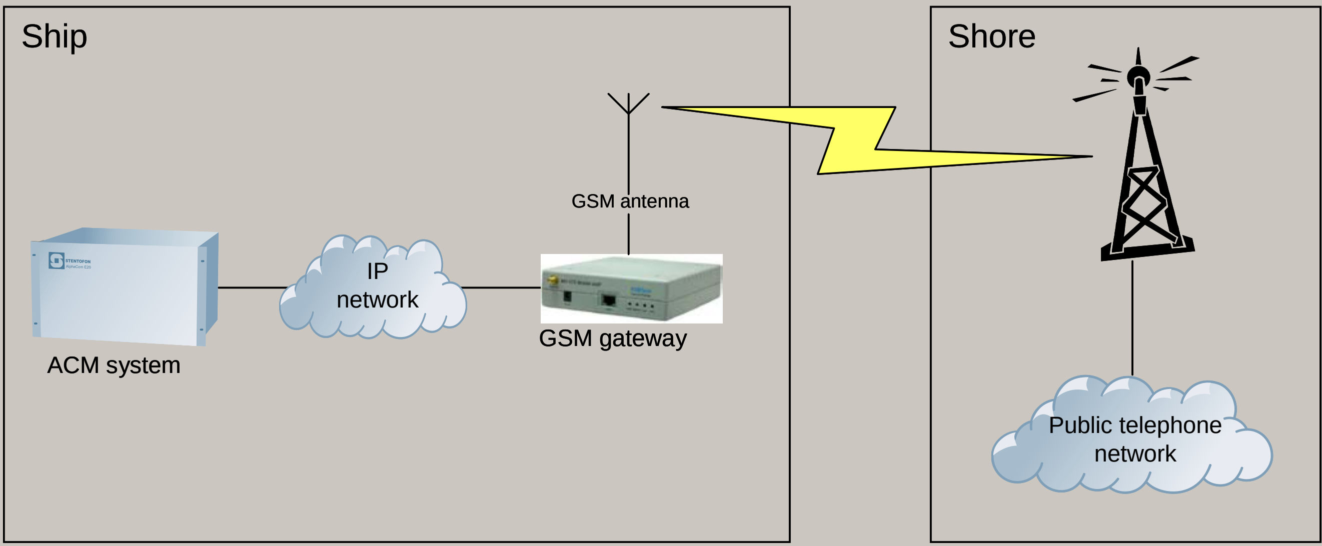


TECHNICAL MANUAL

AlphaCom SIP GSM Gateway

MV-370

Document Contents

1 Basic information

1.1 REVISION HISTORY

Document no. A100K10390
Last revised by: Sen
Revision: v.2.0
Date: 18.07.2017

1.2 SCOPE

This document is a configuration guide describing the setup of the AlphaCom XE system and the MV-370 SIP GSM Gateway. The document covers the most common features used in an AlphaCom XE <-> MV-370 interconnection.
This manual is intended to give relevant information on the system features, available equipment, typical configurations, simplified wiring, programming, and technical data for the concept.
This document is aimed at:

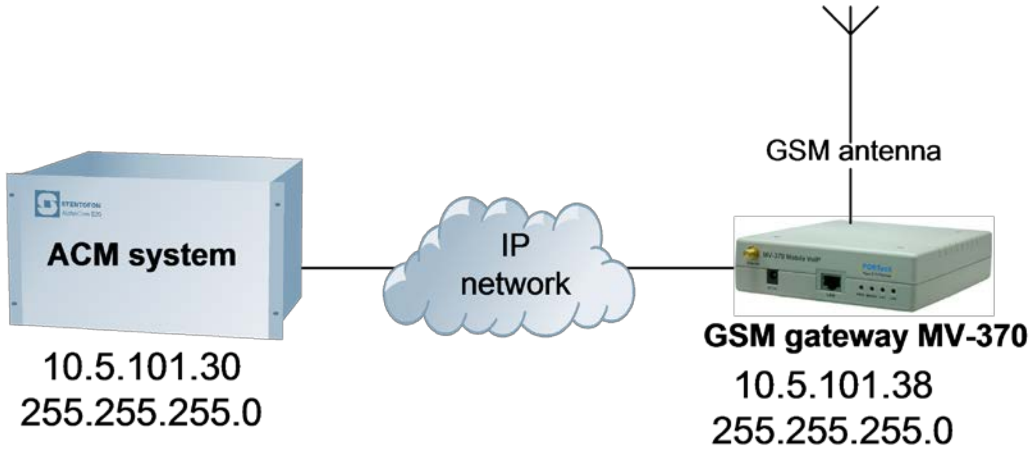
- Sales and marketing personnel
- Consultants
- Installers
- End users

1.3 RELATED DOCUMENTATION

For further information, refer to the following documentation

Doc.no.	Documentation
A100K10805	AlphaCom XE Installation, Configuration & Operation Manual
	MV-370 VoIP GSM Gateway User Manual

2 ALPHACOM CONFIGURATION



The SIP GSM Gateway must be equipped with a SIM card and registered to the GSM network as a regular mobile phone subscriber. Via the gateway, calls can be made from the AlphaCom XE to the GSM network, as well as from the GSM network in to the AlphaCom XE.

The AlphaCom must be equipped with license for SIP Trunk.

The configuration of AlphaCom XE with SIP GSM gateway includes the following steps

2.1.1 Using AlphaWeb

- Assign IP address to the AlphaCom Ethernet port
- Insert SIP Trunk licenses
- Firewall (filter) settings

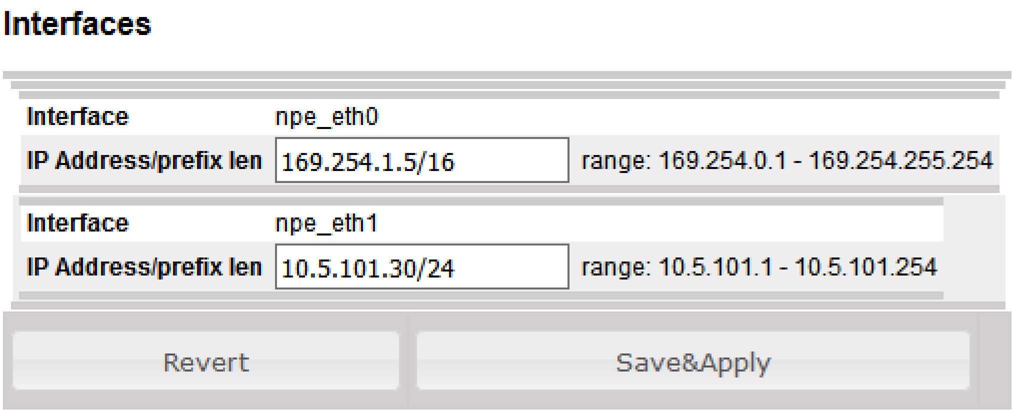
2.1.2 Using AlphaPro

- Create a SIP Trunk Node
- Define the AlphaCom / SIP routing
- Create prefix number
- Update the exchange

2.2 ALPHAWEB CONFIGURATION

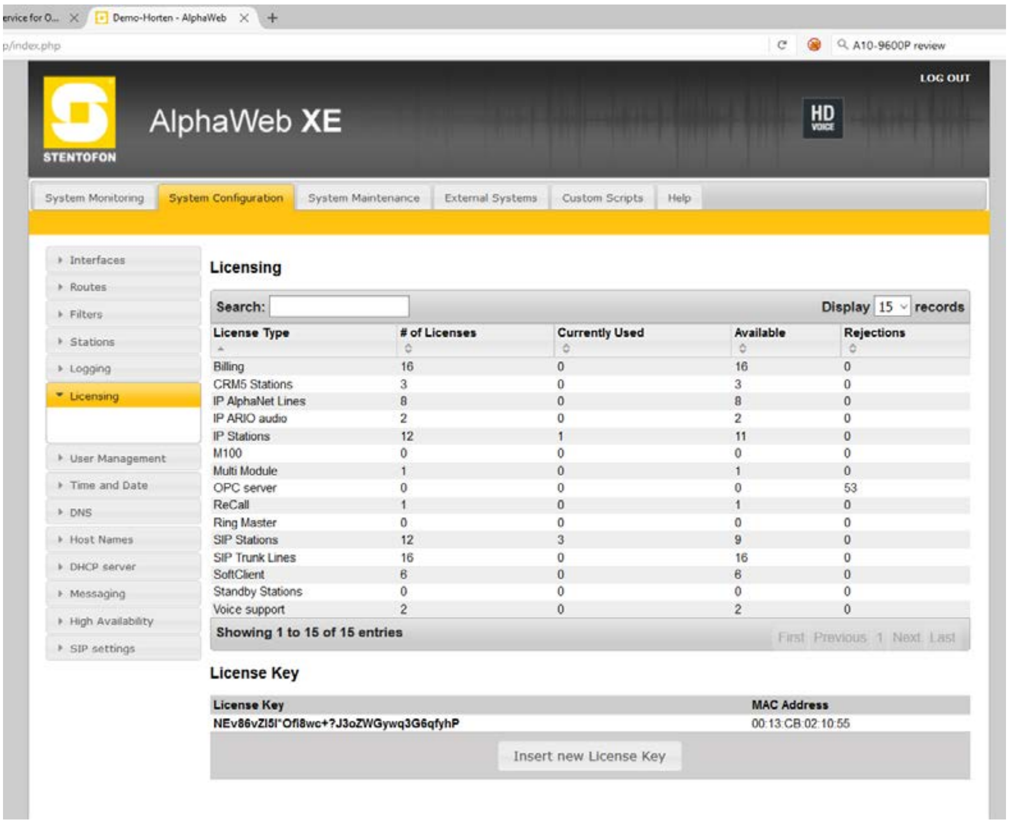
2.2.1 Assign IP address to the AlphaCom E Ethernet port(s)

Log on to AlphaWeb and enter a valid IP address on the Ethernet port.
In the example below, Ethernet port 1 is used. Consult your network administrator to obtain the IP address.



2.2.2 Insert SIP Trunk licenses

Log on to Alpha Web and install the SIP Trunk license.



2.2.3 Firewall (filter) settings

Enable the UDP ports for SIP protocol and VoIP Audio on the desired Ethernet port (default enabled for Ethernet port1).

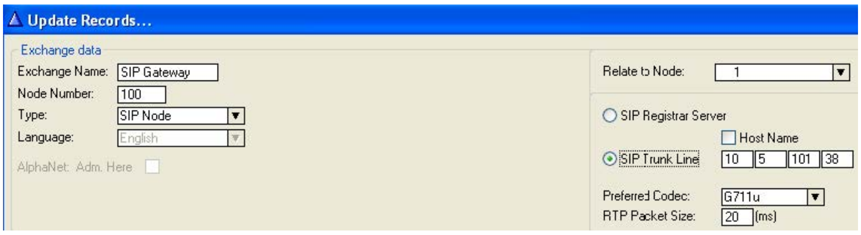
SIP	5060	☑	☑	Delete
VoIP Audio	61000:61150	☑	☑	Delete

2.3 APHAPRO CONFIGURATION

2.3.1 Create a SIP Trunk Node

From the AlphaPro main menu, use the '**+**' button next to the '*Select Exchange*' dropdown list to create a new exchange.

Set the parameters as follows:



The SIP Trunk IP address must be identical to the IP address of the SIP GSM Gateway.
Note:
If the AlphaCom is configured with a SIP Registar node in addition to the SIP Trunk node, the SIP Register node must have a lower node number than the SIP Trunk node.

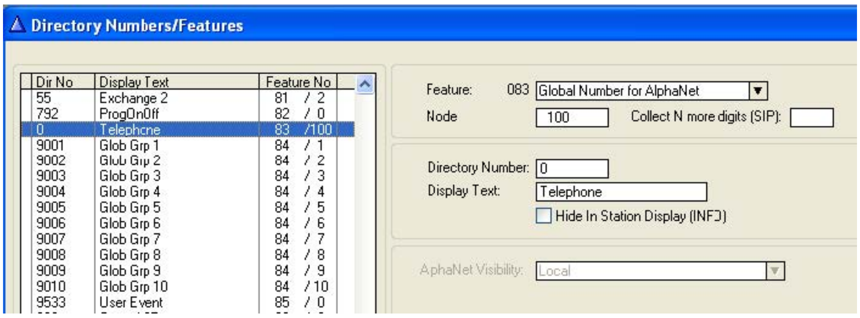
2.3.2 Define the AlphaCom / SIP routing

In Exchange & System > Net Routing use the Insert button to create a route between the AlphaCom and SIP Gateway.

- Set ACDP Link = SIP Link
- Set Preferred codec to G711u and RTP Packet Size to 20 ms.x

2.3.3 Create Prefix number

The directory number (prefix) used to access the GSM network must be programmed in the AlphaCom directory table with feature 83 and Node = SIP Trunk node number (100 in this example). In the example below the default directory number 0 has been modified to be used as a prefix.

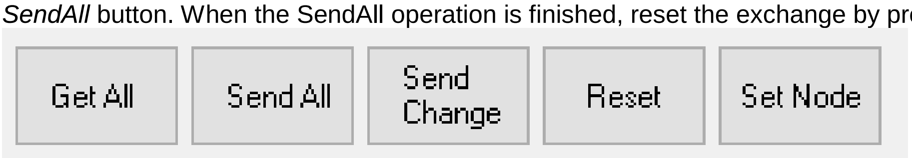


2.3.4 Set SIP Dial Delay parameter

In order to call to preprogrammed phone numbers from a station DAK key or substation call button, set "*SIP dial delay*" = 20 (2 sec.) in in Exchange & System -> System -> VoIP

2.3.5 Update the exchange

Log on to the exchange and update the exchange by pressing the *SendAll* button. When the SendAll operation is finished, reset the exchange by pressing the *Reset* button.



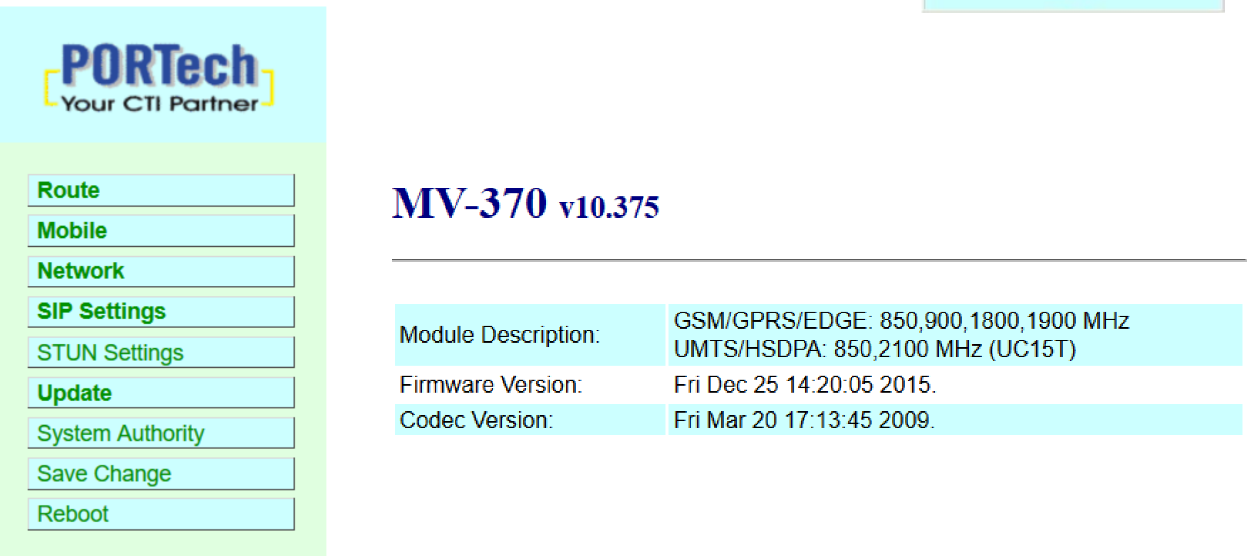
3 Mv-370 CONFIGURATION

3.1 ACCESS TO WEB PAGE

The MV-370 SIP GSM Gateway comes with default network parameters (factory default parameters).

Before you can set up the gateway in the network, you have to change the default IP address to a fixed IP address matching your network environment. The unit is configured from a web browser, e.g. Internet Explorer or Navigator. Consult the network administrator to get the correct IP address. Follow these steps:

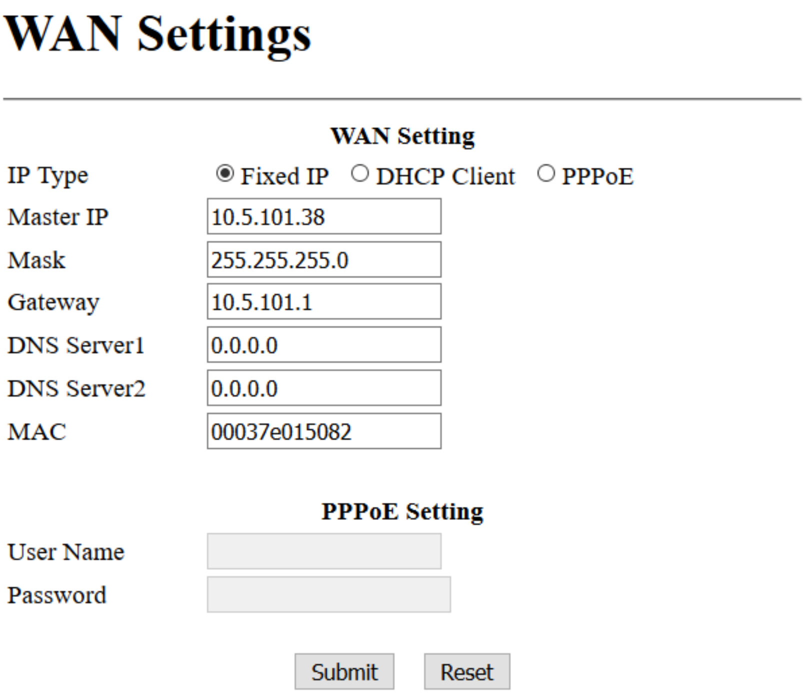
- The default IP address of the gateway is 192.168.0.100, subnet mask 255.255.255.0. The default IP address is set by pressing the reset button (located next to the SIM card) for 3 seconds. Other settings will be kept. Before accessing the web page, the IP address of the configuration PC must be on the same subnet, e.g. 192.168.0.x.
- Change the IP address of your PC to 192.168.0.101, sub mask 255.255.255.0.
- Connect the LAN port of the PC to the Ethernet port of the Gateway. Use a crossed cable or connect the PC and the VoIP Gateway to a common switch using straight cables.
- Start your Web Browser and type http://192.168.0.100 in the URL field.
- Type in username voip and password 1234. (Case-sensitive!)
  The Web page opens:



3.2 NETWORK SETTINGS

Change the network settings according to the network environment. Select Network > WAN Settings:

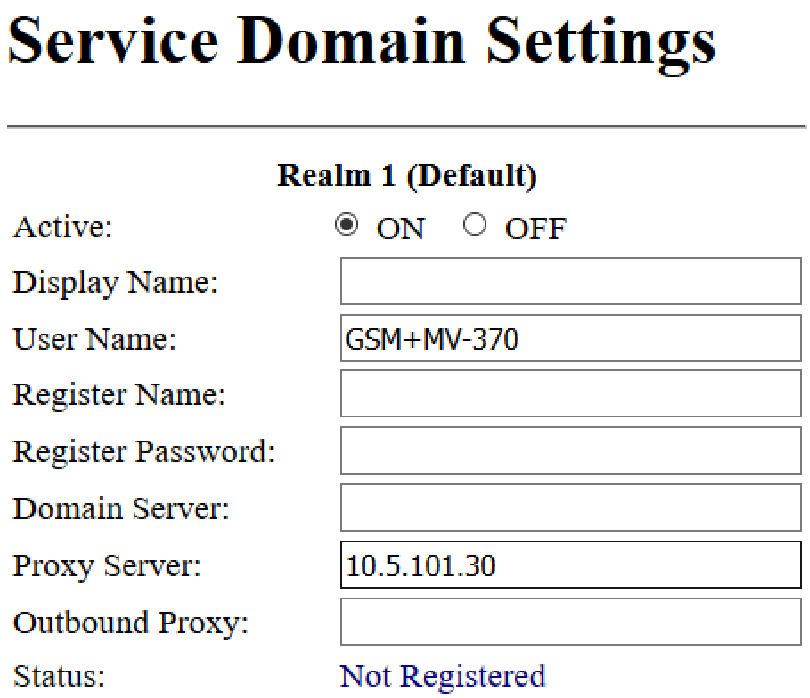
- **IP Type** = Enable *Fixed IP*
- **IP** = IP address of the SIP GSM Gateway
- **Mask** = Network mask
- Gateway = IP address of the network gateway (if any, else leave it blank).



The IP address of the SIP GSM Gateway must be identical to the IP address of the SIP Trunk Node created in AlphaPro.

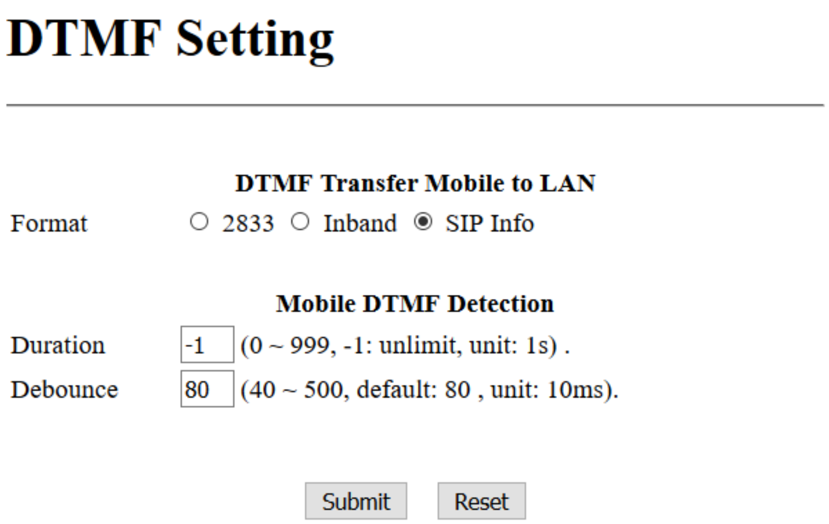
3.3 SIP SETTINGS

In the menu SIP Settings > Service Domain, enter information for"Realm 1":



- Active = ON
- User Name = any text, used for Caller ID. This text will be shown in
  the display on incoming calls from the GSM network, together withthe telephone number
- Proxy Server = IP address of the AlphaCom
- Status will show Not Registered.
- Enable DTMF signaling by SIP INFO method:

- SIP Settings > DTMF Setting: Enable Send DTMF SIP Info



3.4 ROUTE

Select how incoming calls from the GSM network should be handled in Route Mobile to LAN Settings

3.4.1 Alternative 1 (Second dial-tone):

Item	CID	URL
0	*	*
1		

- CID = *
- URL = *

With this setting there will be a second Dial-tone presented on incoming calls from the GSM network. The user must dial the intercom number. The dialing can be terminated by '#', or alternatively one can wait for 5 seconds and the call will be established.

Calls from the GSM network can be forced to be in Private Ringing mode by setting a flag in AlphaPro: Exchange & System -> System -> VoIP: Incoming calls from SIP in private ringing mode

3.4.2 Alternative 2 (Call a station):

Item	CID	URL
0	*	101
1		

- CID = *
- URL = *

Incoming calls will automatically be connected to station 101 (or any other station number you enter).

Calls from the GSM network can be forced to be in Private Ringing mode by setting a flag in AlphaPro: Exchange & System -> System -> VoIP: Incoming calls from SIP in private ringing mode

3.4.3 Alternative 3 (Ringing group):

Item	CID	URL
0	*	6701
1		

- CID = *
- URL = *

Incoming calls are forwarded to a group of stations. There are by default ten directory numbers for ringing groups: 6701 – 6710, linked to groups 51 to 60. The members of the ringing group are defined in the Group menu in AlphaPro. When a call is received a special ringing tone is heard in the stations, which will distinguish an external call from internal calls.

3.5 MOBILE SETTINGS

3.5.1 Mobile > Settings > SIP from:

- Select User/Tel (Not Reg)

Mobile Setting

VoIP Tx Gain	9 (0~12)	VoIP Rx Gain	11 (0~15)
LAN Dialtone Vol	9 (0~12)		

Mobile	○ ON ● OFF		
Routing Range	0 ~ 49		
CODEC Tx Gain	6 (0~7)	CODEC Rx Gain	6 (0~7)
SIP From:	User/Tel (Not Reg)	Answer delay	0 (0~15)
Hide Caller ID	○ ON ◉ OFF	Restart dial fails	1 (0~15)
PIN Code	On ☐ Code:	Confirm:	
Dial Prefix		LAN Answer Mode	Answered
Init AT Cmd			

Submit Reset

3.5.2 Mobile > Settings > Mobile PIN Code:

- If the SIM card needs to be unlocked by a pin code you must enable ON, and enter the pin code, and confirm the pin code

3.5.3 Mobile > Settings > Band Type:

- Choose the GSM frequency

3.5.4 Mobile > Status:

Mobile Status

2005-01-07 07:13

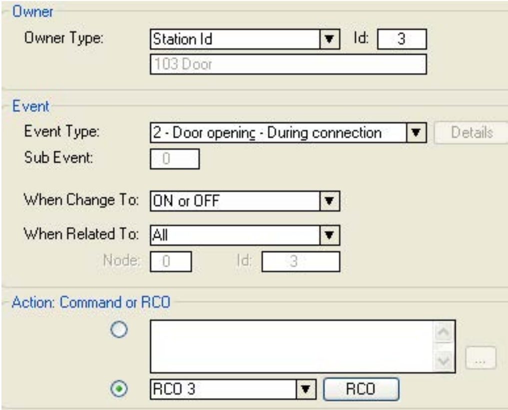
Operator:	24201: N Telenor
SIM Card ID:	242013112602416
Signal Quality:	18
Registration State:	0,1
GSM S/N:	863835023449924

- Shows that the SIM card is in place, and that the Mobil VoIP unit is registered to the GSM network.the GSM frequency

Note! Signal Quality should be 10 or more. If the GSM reception is good, you should get around 18 or 19. Maximum is 31.

4 MISCELLANEOUS FEATURES

4.1 DOOR OPENING FEATURE



During a conversation between a door station and a telephone, the telephone operator can activate the Door Opening feature in the AlphaCom by pressing digit 6.

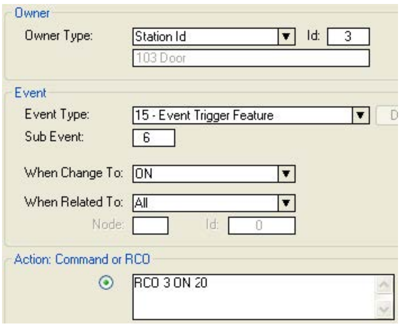
The Door Opening feature is programmed in the Event Handler. There are two separate events for the door opening feature, depending on who is the calling side:

- Calling from the mobile side to the AlphaCom
- Calling from the AlphaCom to the mobile side

4.1.1 Calling from the mobile side to the AlphaCom:

The standard door-opening event is used. i.e. Event type “2 – Door Opening – During conversation”.

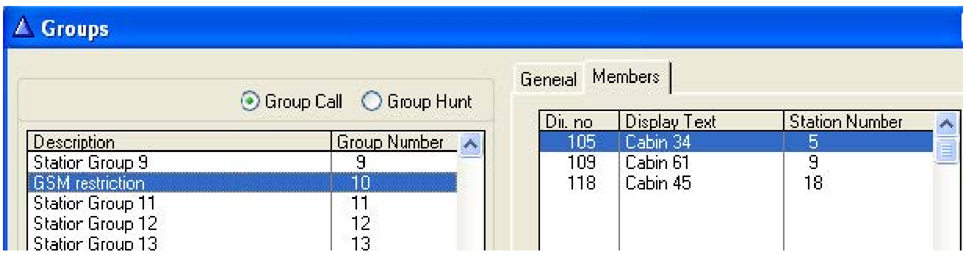
4.1.2 Calling from the AlphaCom to the mobile side:



When the phone presses digit 6, the event type Event Trigger Feature (15) is reported, with the digit 6 as sub event. The calling AlphaCom station is Event Owner, and called SIP phone number and node number is Related To. The RCO pulse time is specified in the RCO action string, i.e. RCO 3 ON 20 means pulse RCO 3 for 2 seconds.

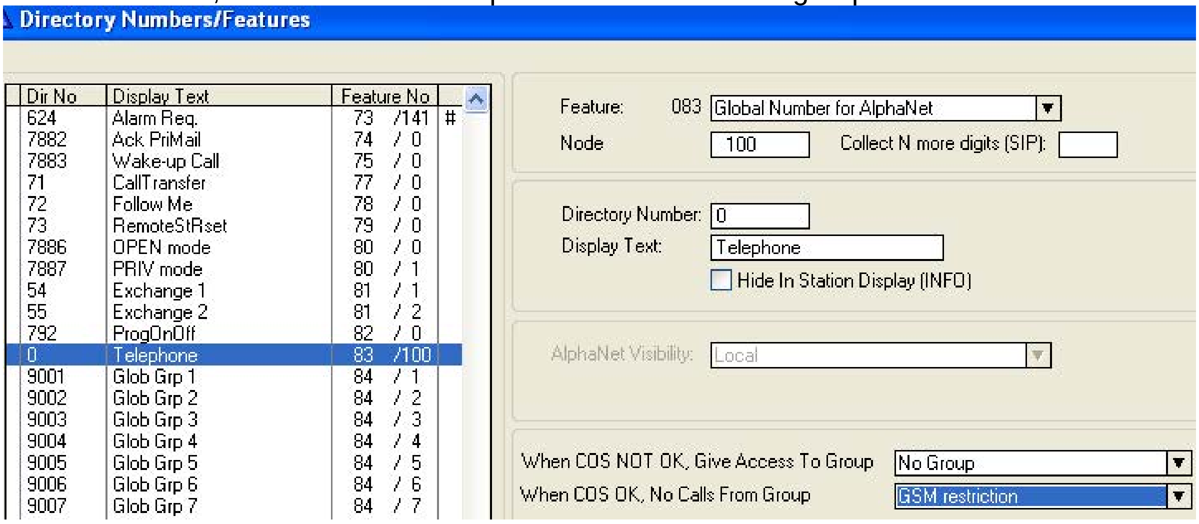
4.2 ACCESS RESTRICTIONS

Selected intercom stations can be denied access to the GSM network.



- In the Groups menu in AlphaPro define a “restriction group” by selecting an unused group and include stations, which are not allowed to call to the mobile network.

- In the Directory & Features menu, select the prefix code for access to the gateway, and in "When COS OK, No Calls From Group" select the restriction group.

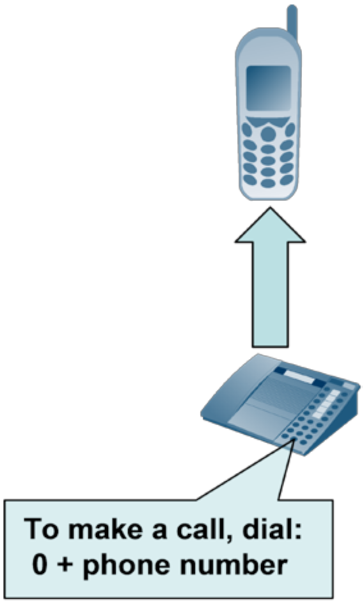


5 FEATURE GUIDE

5.1 MAKE A CALL

- Dial prefix – wait for the dial tone – dial phone number
- When pressing digits during connection, DTMF digits are sent (Call center etc.)
- The DTMF signals ‘*’ and ‘#’ will be transmitted when pressing DAK 1 (*) and DAK 2 (#)
- A complete phone number can be stored under a DAK key or a substation call button.

  Program DAK key from station:
  784 + <prefix> + <phone number> + M + DAK key
  Example: 784 + 0 + 40002500 + M + DAK key
  Program DAK key from AlphaPro:
  I <prefix> P <phone number> Example: I 0 P 40002500



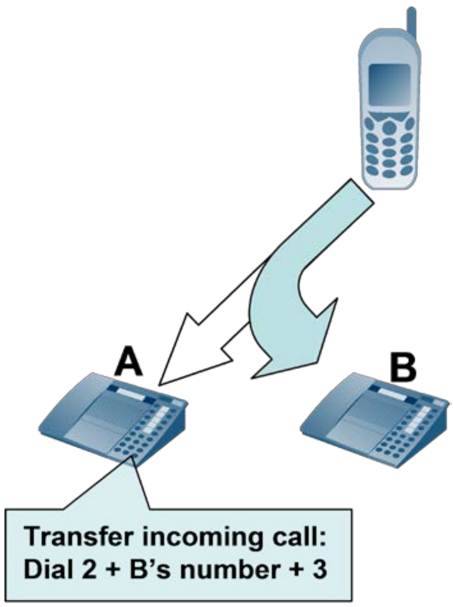
5.2 CALL TRANSFER

Incoming calls from the GSM network can be put on hold and transferred to another intercom station.

- Press 2 to put the current call OnHold
- Dial the intercom number
- Press 3 to transfer the call

Outgoing calls to the GSM line can be transferred to another intercom station.

- Press DAK 9
- Press 2 to put the call OnHold
- Dial the intercom number
- Press 3 to transfer the call

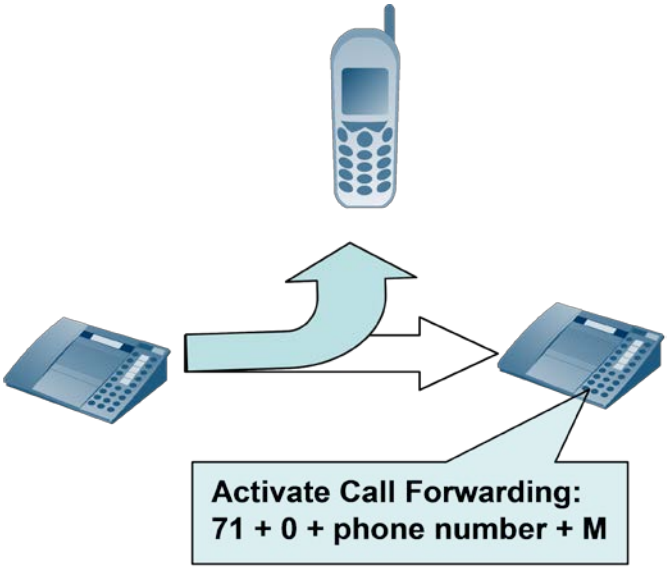


5.3 CALL FORWARDING

Calls to an intercom station can be forwarded to a telephone via the GSM gateway. To activate “Call Forwarding” on a station

- From keypad:
  71 + 0 + <phone number> + M

- A DAK key can be preprogrammed from AlphaPro to activate the call forwarding by a single key press:
  DAK string: I 71 I 0 P 400025000



5.4 M-KEY CONTROL FROM TELEPHONE NETWORK

The ‘*’ and ‘#’ buttons on the telephone can be used to control M-key function (simplex audio) ON or OFF:• From keypad:

- Press the ‘*’-key briefly and the M-key is turned ON

- Press the ‘#’-key briefly and the M-key is turned OFF

Note! This can be useful for group call announcement from the telephone.

5.5 TRANSMIT ‘*’ AND ‘#’ FROM ALPHACOM

The DTMF signals ‘*’ and ‘#’ can be transmitted to the line by pressing the two first DAK keys during the conversation.

- DAK 1 transmits '*'

- DAK 2 transmits #

Note! No configuration is required for this function.

www.zenitel.com