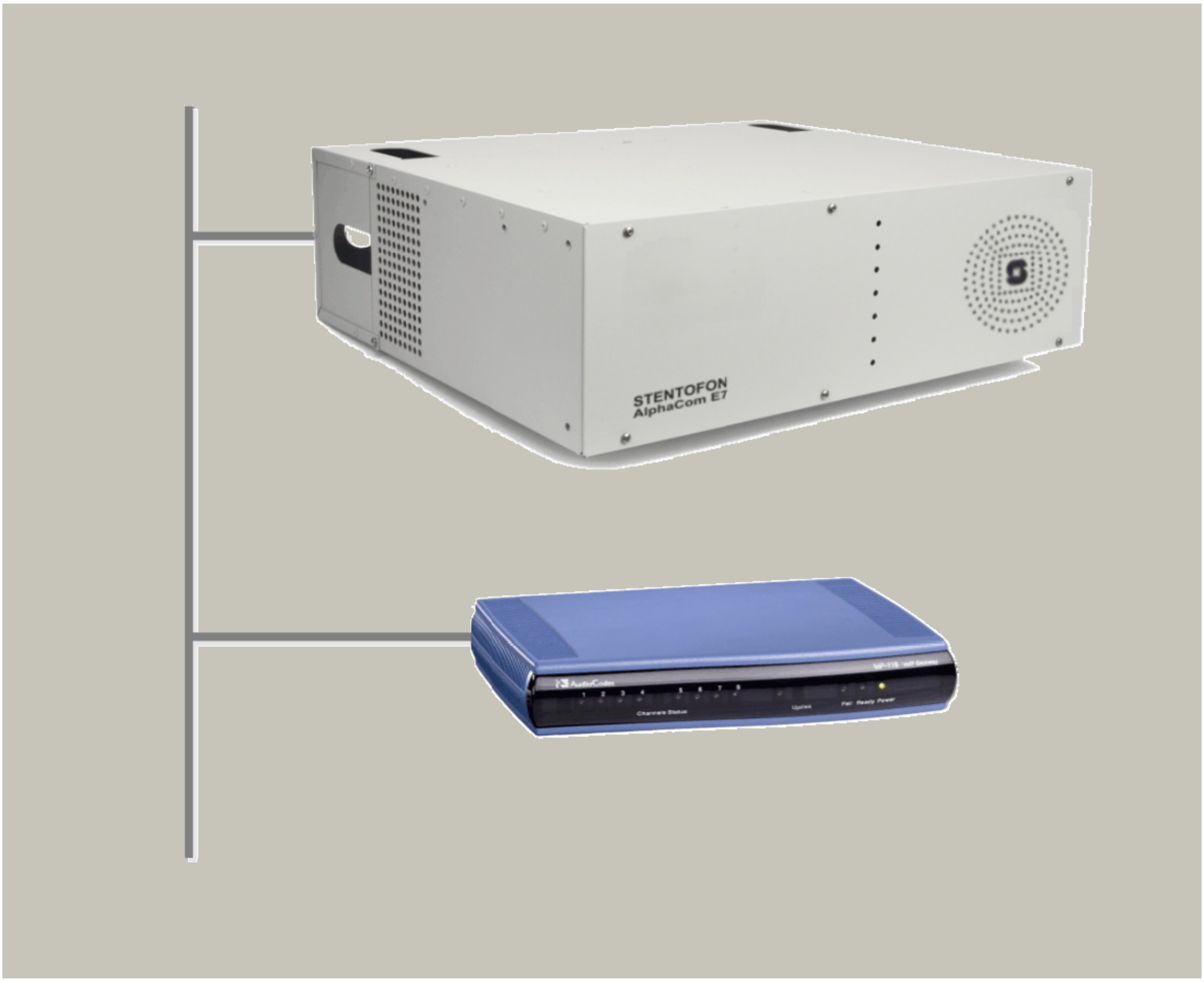
SIP Gateway

AudioCodes MP-114/118

STENTOFON



CONFIGURATION GUIDE

A100K10333 v.5.5

2009-02-04

Table of Contents

1 GENERAL INFORMATION

1.1 About this Document

This document is a configuration guide describing the setup of the AlphaCom E system and the AudioCodes MP-114/118 SIP Gateway. The document covers the most common features used in an AlphaCom E <-> AudioCodes interconnection.

This manual is intended to give relevant information on the system features, available equipment, typical configurations, simplified wiring and programming and technical data for the concept.

This document is aimed at

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

1.2 Related Documents

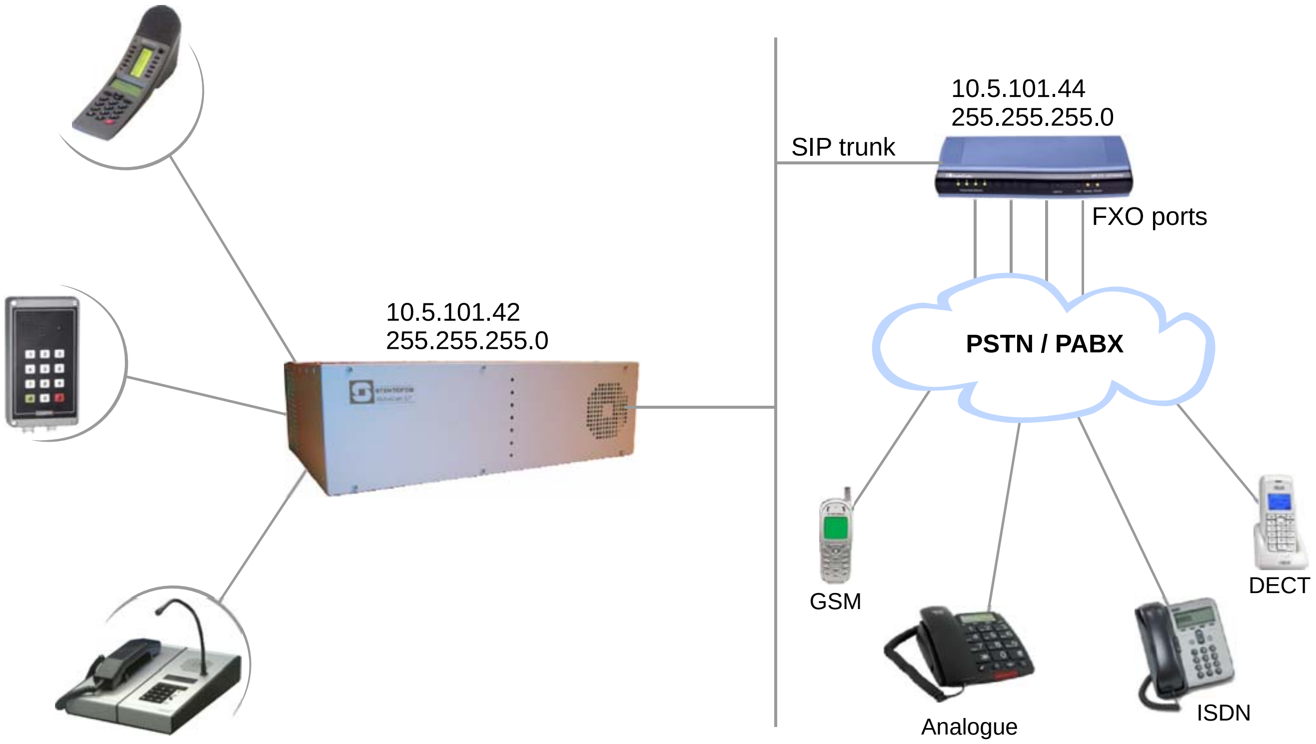
For detailed information on the AudioCodes product please see User Manual 'MP-11x_and_MP-124_SIP_User's_Manual_Ver_5.4.pdf' found on the CD packed with the product.

For detailed information on the AlphaCom E please see the System Management and Operation manual A100K 10318. This manual can be found at www.zenitel.com/stentofon/support.

1.3 Publication log

Version	Date	Modifications
1.0	2007.09.18	First issue, complete.
5.4	2009.02.04	Update according to new SW. Message log added.
5.5	2009.04.16	New front & back, some layout changes. No change in text.

2 ALPHACOM CONFIGURATION



Configuration example

The configuration of AlphaCom E with SIP gateway includes the following steps:

Using AlphaWeb

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

Using AlphaPro

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

2.1 AlphaWeb Configuration

2.1.1 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.

Interface 0	Name: npe_eth0
IP Address:	169 . 254 . 1 . 5
Subnet Mask:	255 . 255 . 0 . 0
Interface 1	**Name: npe_eth1**
IP Address:	10 . 5 . 101 . 42
Subnet Mask:	255 . 255 . 255 . 0

2.1.2 Insert SIP Trunk licenses

☑ Log on to AlphaWeb and install the SIP Trunk license.

License Key

License key	MAC Address
lHj7cKqCbTehZLOO7QY@euzx	00:13:CB:00:02:A5

New License Key

Input License Key	
Insert new license key	

2.1.3 Firewall (filter) settings

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

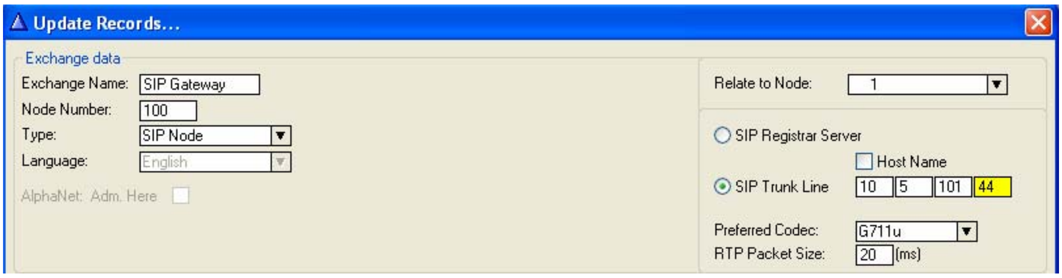
Protocol (UDP)	Port (Lo:Hi)	Eth0	Eth1
sip	5060	☐	☑
VoIP Audio	61000:61150	☐	☑

2.2 AlphaPro Configuration

2.2.1 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.
The exchange type must be set to 'SIP Node'.

☑ Set the parameters as follows:

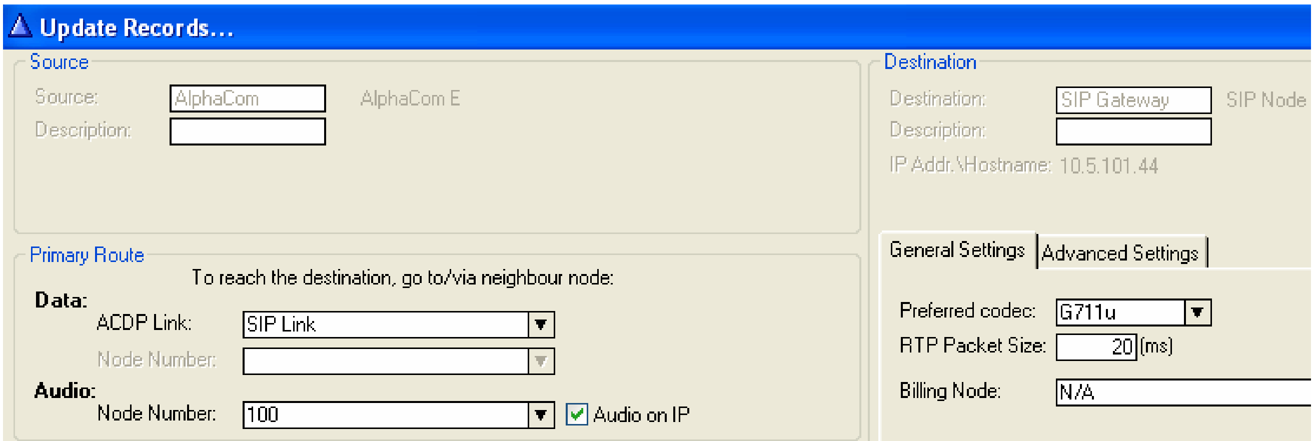


The SIP Trunk IP address must be identical to the IP address of the SIP Gateway.

Note: *If the AlphaCom is configured with a SIP Registrar node in addition to the SIP Trunk node, the SIP Registrar node must have a lower node number than the SIP Trunk node.*

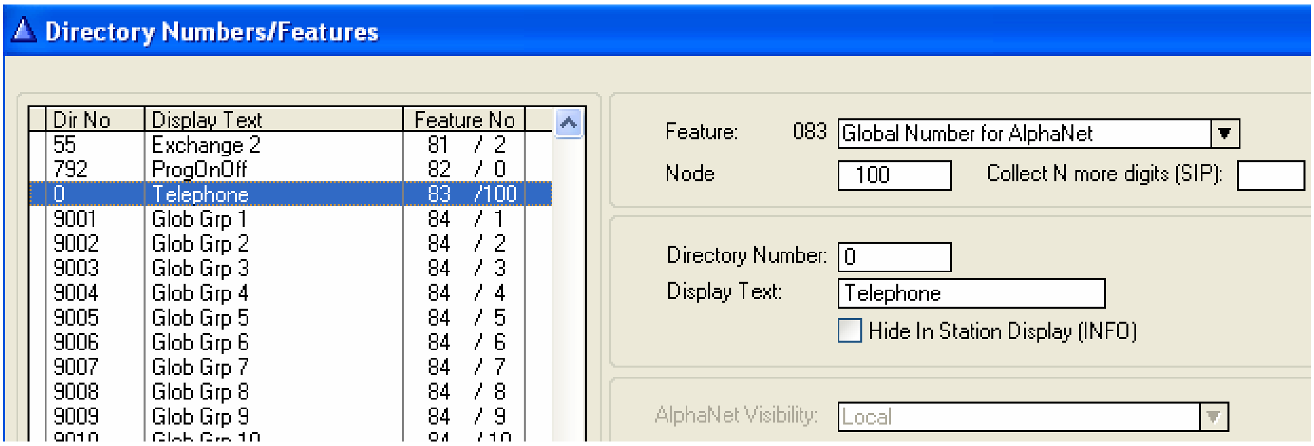
2.2.2 Define the AlphaCom / SIP routing

☑ In **Exchange & System > Net Routing** use the *Insert* button to create a route between the AlphaCom and SIP Gateway. Set **Preferred codec** to *G711u* and **RTP Packet Size** to *20 ms.*



2.2.3 Create Prefix number

☑ The directory number (prefix) used to access the telephone line 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.



2.2.4 Update the exchange

☑ Log on to the exchange and update the exchange by pressing the *SendAll* button. Reset the exchange after the send operation is finished.

3 AUDIOCODES MP-114/118 CONFIGURATION

3.1 Configure Network Parameters

The AudioCodes MP-114/118 VoIP 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 in 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:

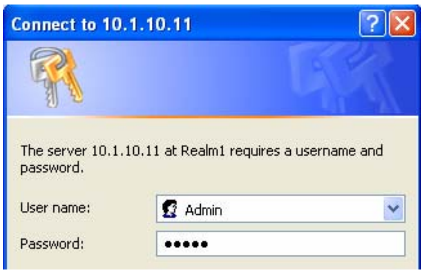
☑ Load factory network parameters and reset the username and password to its default settings (username: **Admin**, password: **Admin**) by following these three steps:

1. Disconnect the Ethernet cable from the device.

2. With a paper clip or any other similar pointed object, press and hold down the Reset button (located on the rear panel) for about six seconds; the Fail LED turns red and the device restores to factory default settings.

3. When the Fail LED turns off, reconnect the Ethernet cable to the device.

The VoIP Gateway will now get the IP address 10.1.10.11, subnet mask 255.255.0.0.



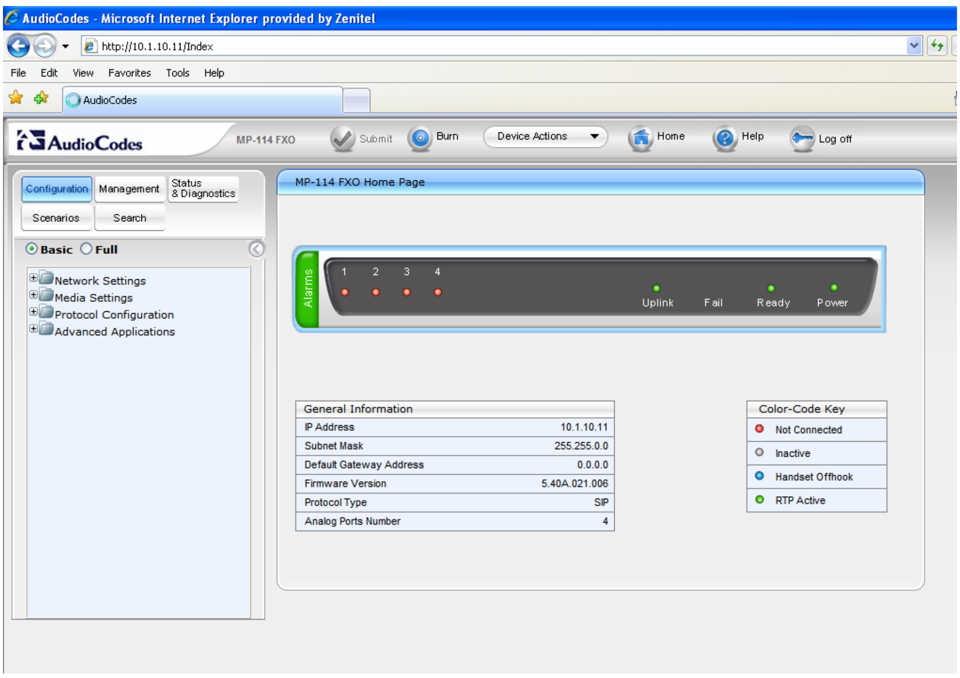
☑ Change the IP address of your PC to 10.1.10.12, subnet mask 255.255.0.0.

☑ 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 **10.1.10.11** in the URL field.

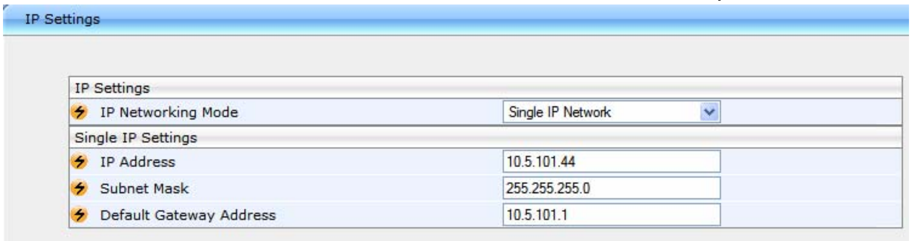
☑ Type in user name *Admin* and password *Admin*. (Case-sensitive!)

The Home page of the Web Interface:



3.2 IP Configuration

☑ In the 'IP Settings' page (**Configuration** tab > **Network Settings** menu > **IP Settings** page item) enter the **IP Address**, **Subnet Mask** and optionally the **Default Gateway Address** of the AudioCodes Gateway.
This IP address must be identical to the IP address of the SIP Trunk Node created in AlphaPro.



☑ Click **Submit** to apply the changes.

Note: *The IP address is immediately changed when pressing Submit, but it is not permanently stored. Without resetting or powering off the device, you need to log on to the Gateway using its new IP address in order to Burn the new IP address to flash:*

☑ Disconnect the PC from the Gateway.

☑ Reconnect the Gateway and PC to the LAN.
The PC and Gateway must be on the same sub-net.

☑ Restore the PC’s IP address and subnet mask to what they originally were, and re-access the Gateway using the new assigned IP address.

☑ Click **Burn** to permanently apply the changes.

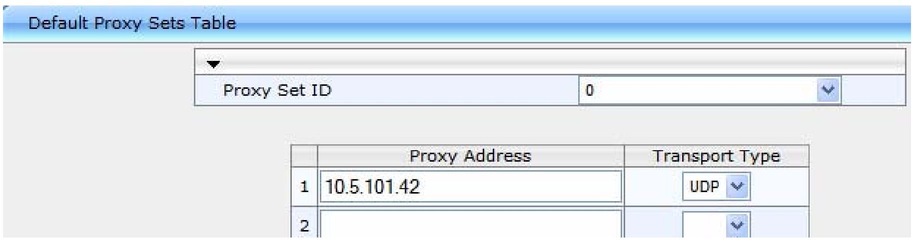
3.3 SIP Parameters

☑ In the 'Proxy & Registration' page (**Configuration** tab > **Protocol Configuration** menu > **Protocol Definition** submenu > **Proxy & Registration** page item) set the ‘Use Default Proxy’ field to ‘Yes’.

☑ Click the **Proxy Set Table** button. In the ‘Proxy Address’ field enter the IP address of the AlphaCom.

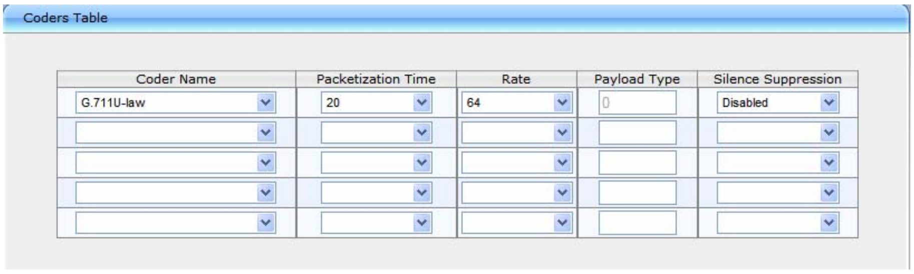
☑ Set 'Transport Type' to 'UDP'.

☑ Press **Submit** to save changes



3.4 Audio Codec

☑ Select the voice coder in 'Coders Table' page (**Configuration** tab > **Protocol Configuration** menu > **Protocol Definition** submenu > **Coders** page item). Choose *G.711U-law* Coder Name, *20 ms* Packetization Time and Silence Suppression *Disabled*.



☑ Press **Submit** to save changes

3.5 About Saving Changes

The **Submit** button will save the data to the running volatile memory. The changes take effect on-the-fly. The changes ***will not*** survive hardware reset or power off.

To permanently save the configuration data, store the data to flash memory by selecting **Burn** from the Tool Bar.

Note: *Parameters proceeded by a yellow lightning symbol is not changeable on-the-fly and require that the device is reset.*

3.6 Backup and Restore

The configuration of the AudioCodes Gateway can be stored to a file on your PC.

From the Tool Bar select **Device Actions** -> **Save Configuration File**. Select **Save INI File** to save the configuration to the PC, and select **Load INI File** to upload a configuration file to the SIP Gateway.

3.7 AlphaCom to Telephone Network

There are two ways of selecting a FXO line from the AlphaCom.

- **Group Hunt**, where a prefix is dialed and you are connected to one out of several lines
- **Direct FXO line selection**, where there is one prefix assigned for each of the FXO lines.

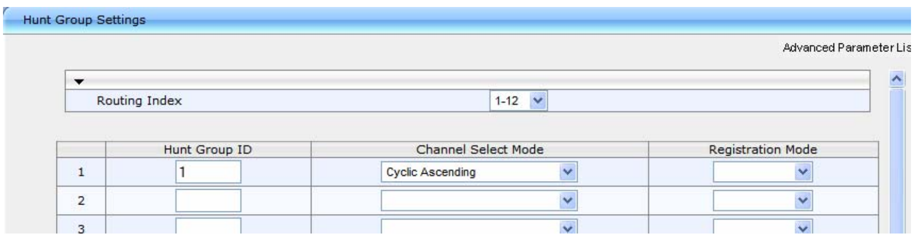
The two methods can be combined.

3.7.1 Group Hunt

Dial a prefix and get connected to a free FXO line.

Hunt Group Settings

☑ In the 'Hunt Group Settings' page (**Configuration** tab > **Protocol Configuration** menu > **Hunt/IP Group** submenu > **Hunt Group Settings** page item) set the 'Hunt Group ID' field to '1' and 'Channel Select Mode' to 'Cyclic Ascending'.



IP to Hunt Group Routing

When dialing the prefix from AlphaCom, the call needs to be routed to the appropriate Hunt Group ID associated with the FXO ports.

In the example below the call is routed to group hunt ID 1. (**Configuration** tab > **Protocol Configuration** menu > **Routing Tables** submenu > **IP to Trunk Group Routing** page item)

IP To Hunt Group Routing Table

Advanced Parameter List

Routing Index: 1-12
IP To Tel Routing Mode: Route calls before manipulation

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID	IP Profile ID
1	*	*	*		1	0
2						

Endpoint Phone Number

In the 'Endpoint Phone Number Table' page the FXO lines are linked to the prefix in AlphaCom and to the hunt group ID.

(**Configuration** tab > **Protocol Configuration** menu > **Endpoint Number** submenu > **EndPoint Phone Number** page item).

In the example below all four FXO lines belong to Hunt Group ID 1. When dialing 0 on an intercom station the first available line will be granted. Directory number 0 must be programmed in the **AlphaCom directory table** with *feature 83/<node>.* See paragraph 2.2.3.

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	0	1	0
2	2	0	1	0
3	3	0	1	0
4	4	0	1	0

If there are unused lines, leave the fields for that line blank.

3.7.2 FXO Line Select

In installations with different types of equipment connected to the various FXO lines the user must be able to select which FXO port to use. On a ship, for instance, there could be a mix of shore lines, GSM interface and Satellite lines.

Line selection is achieved by assigning each port a unique **Phone Number** in the **Endpoint Phone Number Table**. These directory numbers must be programmed in the **AlphaCom directory table** with *feature 83/<node>*. See paragraph 2.2.3.

Replace "0" with four lines, "41" to "44".

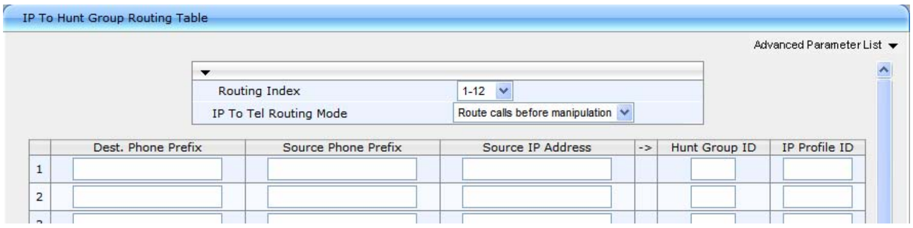
Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	41		0
2	2	42		0
3	3	43		0
4	4	44		0

In this table the four FXO lines are selected by dialing 41 – 44

If there are unused lines, leave all fields for that line blank.

Group hunt is not used in this call mode, and the **IP to Hunt Routing Table** must be empty.



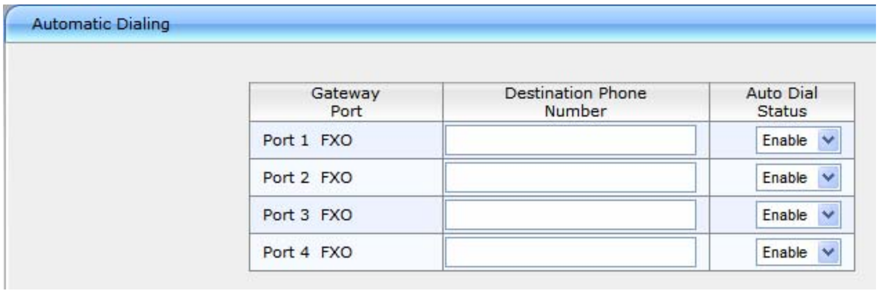
3.8 Telephone Network to AlphaCom

You can choose between three different ways of handling an incoming call from the telephone line:

- Selective Dialing
- Automatic Dialing
- Delayed Automatic Dialing

3.8.1 Selective Dialing

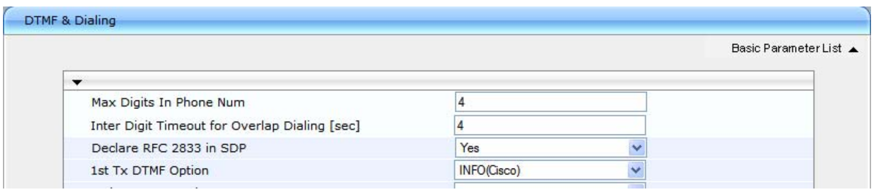
A second dial tone will be presented when calling in, and the user can dial the desired intercom number. The fields in the 'Automatic Dialing' page item must be left blank (Default setting).
(**Configuration** tab > **Protocol Configuration** menu > **Endpoint Settings** submenu > **Automatic Dialing** page item).



In this mode the gateway collects digits from the line, and sets up the call towards the AlphaCom when a predefined number of digits are collected and no more digits are received within a preset time (default 4 seconds), or when the '#' key is dialed.

In the 'DTMF & Dialing' page (**Configuration** tab > **Protocol Configuration** menu > **Protocol Definition** submenu > **DTMF & Dialing** page item) select 'Advanced Parameter List' in order to view all parameters, and set 'Max Digits In Phone Num' equal to the number of digits used on the AlphaCom stations, normally 3 or 4.

The parameter 'Inter Digit Timeout for Overlap Dialing' specifies the waiting time for more digits before setting up the call.



3.8.2 Automatic Dialing (Call to Switchboard)

When calling in, the call will automatically be connected to a predefined intercom number.

☑ Enter the intercom number in the 'Destination Phone Number' field in the 'Automatic Dialing' page item. Set 'Auto Dial Status' to 'Enable'. (**Configuration** tab > **Protocol Configuration** menu > **Endpoint Settings** submenu > **Automatic Dialing** page item).

Automatic Dialing

Gateway Port	Destination Phone Number	Auto Dial Status
Port 1 FXO	103	Enable
Port 2 FXO	101	Enable
Port 3 FXO	6701	Enable
Port 4 FXO	6701	Enable

In the example above, incoming calls on line 1 are routed to station 103, calls on line 2 are routed to station 101, and calls on line 3 and 4 are routed to RingingGroup 6701.

3.8.3 Delayed Automatic Dialing

If '**Auto Dial Status'** is set to '*Hotline',* a second dial tone will be presented when calling in, allowing the user to dial a number. But if no digits are pressed within the '**Hotline Dial Tone Duration'** time, the number in the **Destination Phone Number** is automatically dialed.

Automatic Dialing

Gateway Port	Destination Phone Number	Auto Dial Status
Port 1 FXO	103	Hotline

The '**Hotline Dial Tone Duration'** can be changed from the 'DTMF & Dialing' page item (**Configuration** tab > **Protocol Configuration** menu > **Protocol Definition** submenu > **DTMF & Dialing** page item). Select the 'Advanced Parameter List'. The default value is 16 seconds.

3.8.4 Caller ID

Use the 'Caller Display Information' page to send display information to the intercom station that receives the call. (**Configuration** tab > **Protocol Configuration** menu > **Endpoint Settings** submenu > **Caller Display Information** page item).

Caller Display Information

Gateway Port	Caller ID/Name	Presentation
Port 1 FXO	Shore line 1	Allowed
Port 2 FXO	Shore line 2	Allowed
Port 3 FXO	GSM line	Allowed
Port 4 FXO	SatCom line	Allowed

The prefix code entered in the **End Point Phone Number Table** will be shown together with the text in **Caller ID/Name**.

If Caller ID name is detected on the FXO line, this will be used instead of the Caller ID name in the table above. Display the Navigation Tree in **Full** View. Caller ID from FXO line must be enabled in **Configuration** tab > **Protocol Configuration** menu > **SIP Advanced Parameters** submenu > **Supplementary Services** page item.

☑ Set '**Enable Caller ID'** to '*Enable'* and choose the 'Caller ID Type' as used by the PSTN supplier. Check with the local telephone company to find the 'Caller ID Type' used.

Supplementary Services

Basic Parameter List ▲

Enable Caller ID	Enable
Hook-Flash Code	
Caller ID Type	Standard ETSI

4 FAR END DISCONNECT (FED)

Far End Disconnect refers to methods for detecting that a remote party has hung up. The far end disconnect signal is not mandatory and this could create problems. If the Far End Disconnect signal is not sent to or properly detected by the SIP Gateway, the connection will not be released by the unit, thus freezing the FXO line in the off hook state.

4.1 Call Termination options in the SIP Gateway

The following methods for call termination are supported by the AudioCodes MP-114/118. Note that the used disconnection methods must be supported by the CO (Central Office) or to PBX (Private Branch Exchange).

- Detection of polarity reversal / current disconnect
- Detection of Busy / Dial tones
- Detection of silence
- Timeout of Conversation

4.1.1 Detection of polarity reversal / current disconnect

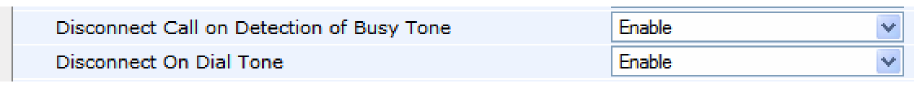
This is the recommended method. The call is immediately disconnected after polarity reversal or current disconnect is detected on the Tel side (assuming the PBX / CO produces this signal).

☑ Display the Navigation Tree in **Full** View. Enable the relevant detection method in **Configuration** tab > **Protocol Configuration** menu > **SIP Advanced Parameters** submenu > **Advanced Parameters** page item.

4.1.2 Detection of Busy / Dial tones

The call is immediately disconnected after Busy or Dial tone is detected on the Tel side (assuming the PBX / CO produces this tone). This method requires the correct tone frequencies and cadence to be defined in the Call Progress Tones (CPT) file of the SIP Gateway. If these frequencies are not known, define them in the CPT file (the tone produced by the PBX / CO must be recorded and its frequencies analyzed). This method is slightly less reliable than the previous one.

☑ Open the 'FXO Settings' page item in **Configuration** tab > **Advanced Applications** menu > **FXO Settings** page item and enable the relevant detection method.



<u>Call Progress Tones (CPT)</u>

The Detection of Busy / Dial tones method requires the correct tone frequencies and cadence (on/off sequence) to be defined in the Call Progress Tones (CPT) file of the SIP gateway. These tones are region specific and telephone exchange dependent.

The Call Progress Tones (CPT) configuration file is a binary file (with the extension *.dat*).

Users can either use one of the supplied configuration (*dat*) files found on the CD provided with the gateway, or construct their own file.

A file with the most common tone patterns can also be downloaded from www.zenitel.com/stentofon/support

To construct your own file, either:

- Modify the supplied *usa_tone.ini* file (in any standard text editor) to suit the specific requirements, and convert the modified *ini* file into binary format using the **TrunkPack Downloadable Conversion Utility**.

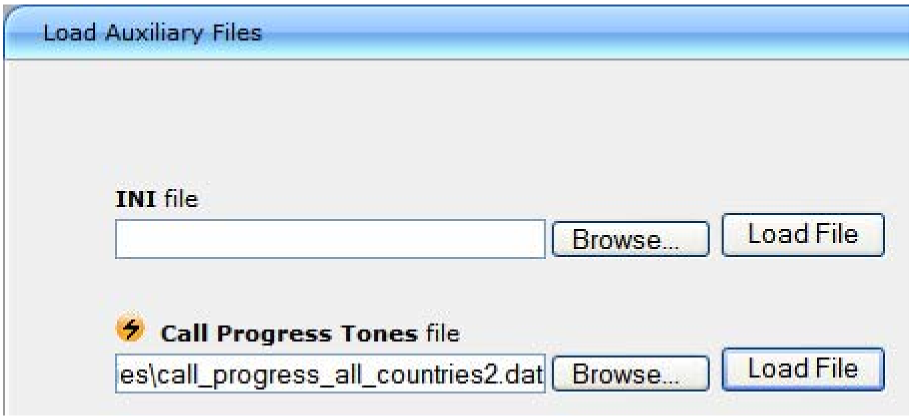
Or:

- Use the Call Progress Tones Wizard.
  The **Call Progress Tones Wizard** (*CPTWizard*) is an application designed to detect the Call Progress Tones generated by your PBX (or telephone exchange) to create a basic Call Progress Tones *ini* file containing definitions for all relevant Call Progress Tones. This provides a good starting point when configuring the SIP gateway. This *ini* file can then be converted to a *dat* file using the **TrunkPack Downloadable Conversion utility**.

Both the **TrunkPack Downloadable Conversion Utility** (*DConvert.exe*) and the **Call Progress Tones Wizard** (*CPTWizard.exe*) can be found on the AudioCodes CD.

☑ Load a Call Progress Tones (*dat*) file to the SIP gateway:

- Select Management tab > Software Update menu > Load Auxiliary Files page item.



- Click the *Browse* button and navigate to the folder that contains the file you want to load
- Click the file and click the *Open* button; the name and path of the file appear in the field beside the *Browse* button
- Click the *Load File* button
- *Burn* the configuration so the file can be available after a power failure
- *Reset* the SIP Gateway for the changes to take effect

For more detailed information regarding Call Progress Tones please refer to the AudioCodes User's Manual found on the CD supplied with the unit.

4.1.3 Detection of Silence

The call is disconnected after silence is detected on both call directions for a specific (configurable) amount of time. This method should only be used as a backup.

☑ Display the Navigation Tree in **Full** View. Enable the relevant detection method in **Configuration** tab > **Protocol Configuration** menu > **SIP Advanced Parameters** submenu > **Advanced Parameters** page item.

4.1.4 Timeout of Conversation

As an additional safety to prevent lines from accidentally locking up, it is recommended to enable a timeout of conversation.

The ‘Max Call Duration’ defines the maximum call duration in minutes. If this time expires, both sides of the call are released (IP and Tel). The valid range is 0 to 120. The default is 0 (no limitation).

☑ Display the Navigation Tree in **Full** View. Enable the ‘Max Call Duration’ in **Configuration** tab > **Protocol Configuration** menu > **SIP Advanced Parameters** submenu > **Advanced Parameters** page item.

5 MESSAGE LOG

The 'Message Log' page displays Syslog debug messages sent by the device. You can select the Syslog messages in this page, and then copy and paste them into a text editor such as Notepad. This text file (txt) can then be sent to Technical Support for diagnosis and troubleshooting.

Note: *It's not recommended to keep a Message Log session open for a prolonged period. This may cause the device to overload. For prolonged (and detailed) debugging, use an external Syslog server*.

To activate the Message Log, take these 3 steps:

☑ In the 'Advanced Parameters' (**Configuration** tab > **Protocol Configuration** menu > **SIP Advanced Parameters** submenu > **Advanced Parameters** page item), set the parameter 'Debug Level' to 6. This parameter determines the Syslog logging level in the range 0 to 6, where 6 is the highest level.

☑ Open the 'Message Log' page (**Status & Diagnostics** tab > **Status & Diagnostics** menu > **Message Log** page item). Now the 'Message Log' page is displayed and the log is activated.

```
Log is Activated

1d:14h:6m:20s (      lgr_flow)(481       )  ---- Incoming SIP Message from 10.5.101.30:5060 ----
1d:14h:6m:20s INVITE sip:0@10.5.101.44 SIP/2.0
Via: SIP/2.0/UDP 10.5.101.30:5060;branch=z9hG4bK769344622
From: "Stentofon Statio" <sip:9547@10.5.101.30>;tag=1339896590
To: "" <sip:0@10.5.101.44>
Call-ID: 1738965555@10.5.101.30
CSeq: 8136 INVITE
Contact: <sip:9547@10.5.101.30:5060>
Max-Forwards: 70
User-Agent: AlphaSip gateway / 0116
Subject: Forwarding AlphaCom call
Expires: 120
Allow: INVITE, REGISTER, ACK, BYE, CANCEL, INFO
Content-Type: application/sdp
Content-Length:   142

v=0
o=StentofonStatio 20000001 20000001 IN IP4 10.5.101.30
s=-
c=IN IP4 10.5.101.30
t=0 0
m=audio 61002 RTP/AVP 0
a=rtpmap:0 PCMU/8000
1d:14h:6m:20s (      lgr_flow)(483       )  |       | new GetNewSIPCall created - #8
```

The displayed logged messages are color coded as follows:

- Yellow - fatal error message
- Blue - recoverable error message (i.e., non-fatal error)
- Black - notice message

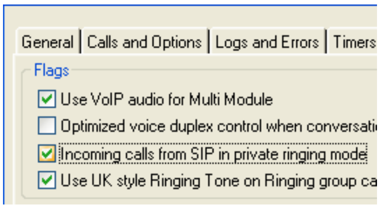
☑ To clear the page of Syslog messages, in the Navigation tree, click the page item Message Log again; the page is cleared and new messages begin appearing.

To stop the Message Log, take this step:

☑ Close the page by accessing any another page in the Web interface.

6 MISCELLANEOUS FEATURES

6.1 Incoming Calls in Private



Incoming calls from the telephone line can be forced to be in private ringing mode, independent of the private/open switch of the intercom station.

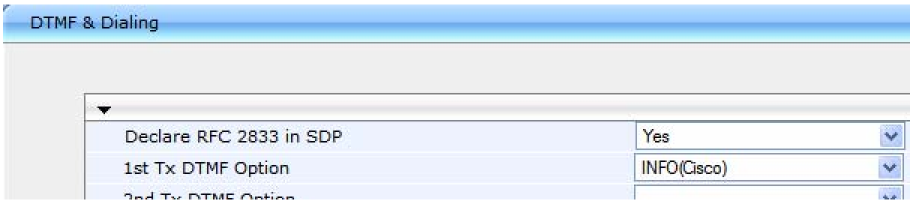
Check the flag **Incoming calls from SIP in private ringing mode** in AlphaPro, (**Exchange & System > System > VoIP**).

6.2 Door Opening Feature

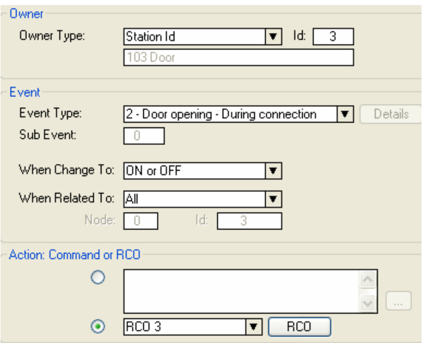
During a conversation between a door station and a telephone, the telephone operator can activate the *Door Opening* feature in the AlphaCom by pressing digit 6.

6.2.1 SIP Gateway configuration

To enable digit actions from the telephone line during conversation, set '**1st Tx DTMF Option'** to '*INFO(Cisco)'* in the 'DTMF & Dialing' page. (**Configuration** tab > **Protocol Configuration** menu > **Protocol Definition** submenu > **DTMF & Dialing** page item).



6.2.2 AlphaCom configuration

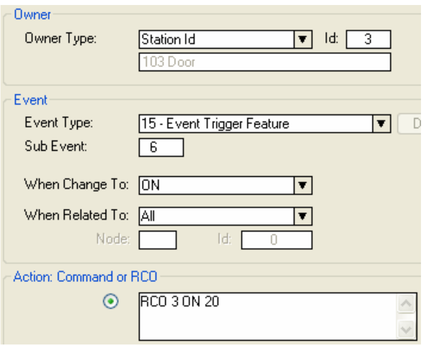


The Door Opening feature is programmed in the Event Handler. There are two separate events for the door opening feature, depending on who is the calling side:

- calling from the telephone to the door
- calling from the door to the telephone

<u>Calling from the telephone to the door:</u>

The **Standard** door opening event is used.



<u>Calling from the door to the telephone:</u>

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 as an additional parameter in the RCO action string, i.e. *RCO 3 ON 20* means pulse RCO 3 for 2 seconds.

6.3 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:

- Press the '∗'-key briefly and the M-key is turned ON
- Press the '#'-key briefly and the M-key is turned OFF

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

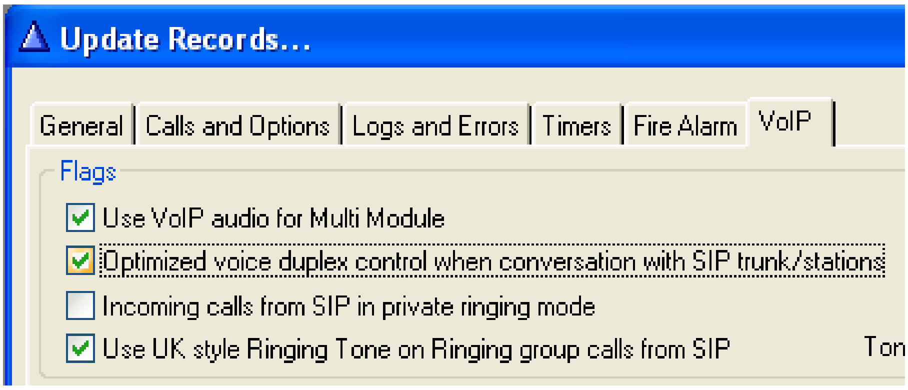
The feature is enabled by setting **1st Tx DTMF Option** to *INFO(Cisco)* in **Protocol Management > Protocol Definition > DTMF & Dialing**.

6.4 Transmit '∗' and '#' from AlphaCom

The DTMF signals '∗' and '#' will be transmitted to the line when DAK 0 (∗) and DAK 1 (#) is pressed during a telephone conversation. No programming is required.

6.5 Voice Switching in Noisy Environment

If the intercom station is located in a noisy environment, it might be difficult to switch the voice direction from the telephone towards the intercom station. However, there is a setting in the AlphaCom to overcome this problem. In AlphaPro, **Exchange & System** > **System** > **VoIP**, set the parameter **Optimized voice duplex control when conversation with SIP trunk/stations**



When this flag is set, the initial voice direction is forced to be from the intercom towards the telephone. When the phone operator starts to speak, the voice direction will switch towards the intercom station, regardless of the level of the audio signal from the intercom station. As soon as the phone operator stops speaking, the voice direction will switch back to the initial direction.

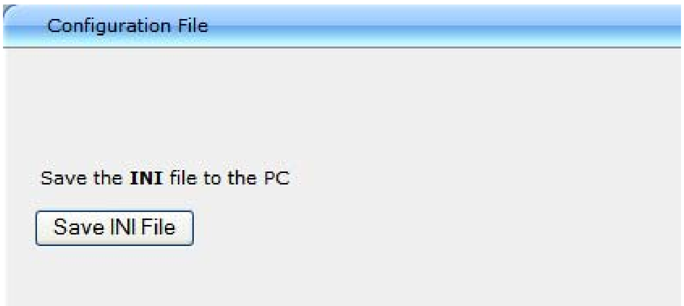
Make sure that the *Echo Canceller* is enabled in the SIP Gateway. (**Configuration** tab > **Media Settings** menu > **Voice Settings** page item > **Echo Canceller** = Enabled).

6.6 Country Settings

The Line Characteristics (AC impedance matching, hybrid balance, Tx & Rx frequency response, Tx & Rx Gains, ring detection threshold, DC characteristics) should be set according to country of origin.

Some of the SIP Gateway parameters are configurable through the *ini* configuration file only (and not via the Web). The **CountryCoefficients** parameter that determines the line characteristics must be configured via the *ini* configuration file.

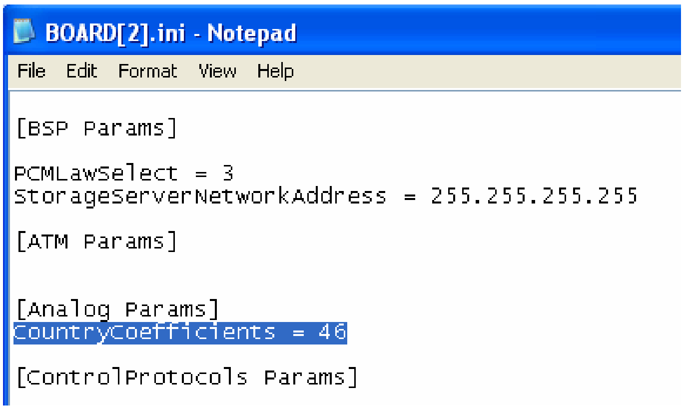
☑ Procedure to modify the *ini* file:



- Get the *ini* file from the gateway using the Embedded Web Server. From the Tool Bar select **Device Actions** -> **Save Configuration File**. Select **Save INI File** to save the configuration to the PC:
- Open the file (e.g. in Notepad) and add anywhere in the file the line

```
CountryCoefficients = xx
```

where xx is the country code found below; save and close the file. The example below shows the settings for Norway (46).



- Load the modified *ini* file back to the gateway (using the button **Load INI File**).

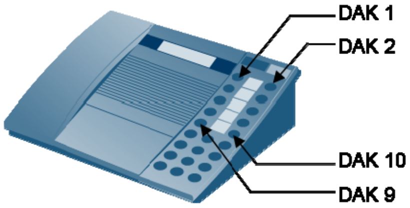
This method preserves the programming that already exists in the device, including special default values that were preconfigured when the unit was manufactured.

6.6.1 Country codes

The default value is 70 (United States).

Argentina	= 0	Finland	= 18	Lebanon	= 36	Russia	= 54
Australia	= 1	France	= 19	Luxembourg	= 37	Saudi_Arabia	= 55
Austria	= 2	Germany	= 20	Macao	= 38	Singapore	= 56
Bahrain	= 3	Greece	= 21	Malaysia	= 39	Slovakia	= 57
Belgium	= 4	Guam	= 22	Malta	= 40	Slovenia	= 58
Brazil	= 5	Hong_Kong	= 23	Mexico	= 41	South_Africa	= 59
Bulgaria	= 6	Hungary	= 24	Morocco	= 42	South_Korea	= 60
Canada	= 7	Iceland	= 25	Netherlands	= 43	Spain	= 61
Chile	= 8	India	= 26	New_Zealand	= 44	Sweden	= 62
China	= 9	Indonesia	= 27	Nigeria	= 45	Switzerland	= 63
Colombia	= 10	Ireland	= 28	Norway	= 46	Syria	= 64
Croatia	= 11	Israel	= 29	Oman	= 47	Taiwan	= 65
Cyprus	= 12	Italy	= 30	Pakistan	= 48	TBR21	= 66
Czech_Republic	= 13	Japan	= 31	Peru	= 49	Thailand	= 67
Denmark	= 14	Jordan	= 32	Philippines	= 50	UAE	= 68
Ecuador	= 15	Kazakhstan	= 33	Poland	= 51	United_Kingdom	= 69
Egypt	= 16	Kuwait	= 34	Portugal	= 52	UnitedStates	= 70
El Salvador	= 17	Latvia	= 35	Romania	= 53	Yemen	= 71

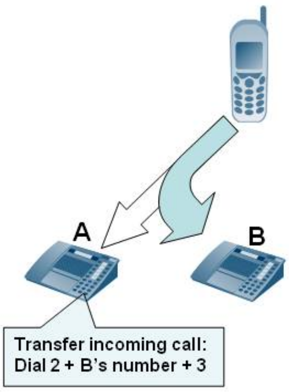
6.7 Feature Guide



- *Make a call from an intercom station*
  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 from AlphaPro*:
  I <prefix> P <phone number>
  Example: I 0 P 40002500



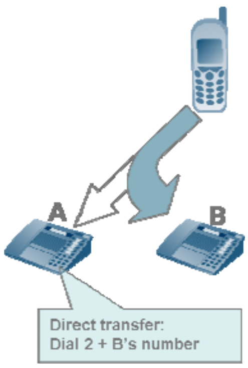
Call Transfer

Incoming calls from the line can be transferred to another station

- *During conversation, dial on the keypad:*
  2 + <intercom station> + 3
- From a preprogrammed DAK:
  D 2 I 104 M M D 3

Outgoing calls to the line can be transferred to another station

- *During conversation, dial on the keypad:*
  DAK9 + 2 + <intercom station> + 3



Option: Blind Transfer

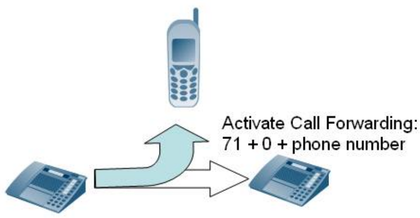
- In AlphaPro, Directory & Features menu, modify the Inquiry feature 2 from default feature 55/0 to feature 55/1.

- *During conversation, dial on the keypad:*
  2 + <intercom station>

Call Forwarding

An intercom call can be forwarded to a telephone

- *From keypad:*
  71 + 0 + <phone number> + M
- *From preprogrammed* DAK*:*
  I 71 I 0 P 40002500



Search List

A telephone number can be included in the Search List of a station.

- *Format:*
  I <prefix> P <phone number>
  Example: I 0 P 40002500

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