SeaCom 3000

User Manual

This manual covers the SeaCom 3000 telephone, talk-back command and intercom system. It includes all the telephone and intercom stations SC211, SC220, SC411 and SC421.

SEACOM

DNV

TYPE APPROVAL CERTIFICATE

Certificate No:
TAA0000054
Revision No:
2

This is to certify:

That the Automatic Telephone System and Hands Free Voice Talk Back System

with type designation(s)
SeaCom 2100, SeaCom 3000, Alphaconnect 128, AlphaConnect Classic

Issued to
Seacom ApS
Åbyhøj, Midtjylland, Denmark

is found to comply with
DNV rules for classification – Ships, offshore units, and high speed and light craft

Application :

See page 2.
Product(s) approved by this certificate is/are accepted for installation on all vessels classed by DNV.

Type	Temperature	Humidity	Vibration	EMC	Enclosure
SeaCom 2100	B	B	A	B	A
SeaCom 3000	B	B	A	B	A
Alphaconnect 128	B	B	A	B	A
AlphaConnect Classic	B	B	A	B	A

Issued at **Høvik** on **2021-12-06**

This Certificate is valid until **2026-12-05**.
DNV local station: **Denmark CMC**

Approval Engineer: **Steinar Kristensen**

for **DNV**
Digitally Signed By: Sjåvåg, Trond
Location: DNV Høvik, Norway

Trond Sjåvåg
Head of Section

This Certificate is subject to terms and conditions overleaf. Any significant change in design or construction may render this Certificate invalid.
The validity date relates to the Type Approval Certificate and not to the approval of equipment/systems installed.

DNV

Job Id:	**262.1-011116-4**
Certificate No:	**TAA0000054**
Revision No:	**2**

Product description

SeaCom automatic telephone system and hands free voice talk back system consists of automatic telephone exchanges and talk back, intercom and telephone stations.
Products are also available in an Alphatron branded version designated as Alphaconnect, consisting of the same hardware and software, but in different colours.
For the talk back, telephone and intercom stations the designation SC xxx represents the SeaCom version of the product, while Pxxx is the Alphaconnect version.
The following products are covered by this Type Approval [1]:

Equipment type	Module	Description	Article no.	SW	Env.[1]
Telephone exchanges	SeaCom 3000	Telephone exchange with maximum 144 telephone and communication station positions	10-092-0305_01xx	N.A.	Protected
	AlphaConnect Classic		10-091-0500_01xx	N.A	
	SeaCom 2100	Telephone exchange with maximum 128 telephone and communication station positions	10-092-0200_02xx	N.A.	
	AlphaConnect 128		10-091-0100_06xx	N.A.	
Circuit boards and modules for exchanges	PSU	Power supply unit, 18-32 VDC input,+/- 5V, -48V and 80V AC output	10-110-1020_03xx	02xx	
	CP2	Central processor board, including Ethernet, serial and USB ports [2]	10-110-1011_05xx	OS0301 +CP08xx	
	FIO2 Master	2 line trunk board with master clock and communications on board	10-110-1031_06xx	06xx	
	AEXT8	8 line analogue extension board	10-110-1041_06xx	05xx	
	PIM	Power input module for SeaCom 2100 and AlphaConnect 128	20-110-0060_02xx	N.A.	
	PIM	Power input module, 24V DC Mains and battery input for SeaCom 3000 and AlphaConnect Classic	20-110-0070_03xx	N.A.	
	CTU2	Cable termination unit, 16+2 lines	10-110-1061_02xx	N.A.	
	CTU24-24	Cable termination unit, 24 lines	10-110-1224_01xx	N.A.	
	CTU24-8	Cable termination unit, 8 lines	10-110-1208_01xx	N.A.	
	PDU	Power distribution unit, 24VDC for telephone and intercom stations	10-110-1200_01xx	N.A.	
	FIO4	Trunk Line Interface, 4 lines	10-110-1404_02xx	02xx	
	FIO4-2	Trunk Line Interface, 2 lines	10-110-1402_02xx	02xx	
	AEXT16-8	Telephone Line Interface, 8 lines	10-110-2020_02xx	01xx	
	AEXT16-16	Telephone Line Interface, 16 lines	10-110-2021_02xx	01xx	
	AEXT16-24	Telephone Line Interface, 24 lines	10-110-2022_02xx	01xx	
	Backplane	Backplane	20-110-0030_02xx	N.A.	
	LSP	System processor	10-110-1404_02xx	01xx	
Talk back station	SC 211	Simple water tight telephone station with 3 buttons	10-102-0211_06xx	01xx	Exposed
	P211		10-103-0211_06xx	01xx	Exposed
Telephone station	SC 220	Water tight and salt mist resistant industrial telephone station with built in call relay	10-102-0220_04xx	01xx	Exposed
	P220		10-103-0220_04xx	01xx	Exposed
Telephone and intercom station	SC 411	Flush mount telephone station with display, backlight and hands-free	10-102-0411_02xx	01xx	Protected
	P411		10-103-0411_02xx	01xx	Protected
	SC 421	Water tight telephone station with display, backlight and hands-free	10-102-0421_0300	01xx	Exposed
	P421		10-103-0421_03xx	01xx	Exposed
Accessories	Handset/ cradle	Watertight handset with cradle for use with SC211/P211, SC220/P220 and SC421/P421	10-400-1020_06xx	N.A	Exposed

[1] Actual configuration may vary based on requirements for individual installations. Only modules listed in this certificate are approved for installation.
[2] Ethernet, Serial and USB ports may only be used for connections internally within the system, and are not approved for connection to other systems/ networks

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Equipment type	**Module**	**Description**	**Article no.**	**SW**	**Env.[*)]**
	Handset/ cradle	Flush mount, SeaCom Flush mount, Alphatron	10-400-1050_01xx 10-400-1053_01xx	N.A	Protected
	Headset	Noise cancelling headset with 10m cable for use with SC211/P211, SC220/P220 and SC421/P421	10-400-0205_01xx	N.A	Protected

[*)] Location categories according to IEC 60945 (2002)

Application

The SeaCom/ Alphaconnect automatic telephone system and hands free voice talk back system may be used in cargo ships, passenger vessels and mobile offshore units for compliance with the following codes, rules and regulations:

- SOLAS
- MODU Code
- HSC Code
- DNV Statutory Interpretations DNV-SI-0364 [July 2021]

Type Approval documentation

DNV No	Document Id.	Rev.	Description
46	P21-0149	2021-12-01	Report: EKTOS, Supplementary EMC Test report 2-6GHz for SeaCom SC2100, SC3000, SC220, SC411, Alphatron P211 and P421
40	DOC20160908CJ02	08	Report: SeaCom, TA Test report AlphaConnect Classic and SeaCom3000
28	TMN20170417CJ01	0105	Manual: SeaCom, SeaCom3000 User Manual
26	117-22251-1	2017-11-29	Report: Force Technology, Test for marine type approval of AlphaConnect automatic telephone and intercom system
25	DOC110225CJ01_03	03	Report: SeaCom, Performance type test report for SeaCom communication system
24	TMN100510CJ01_03	0404	Manual: SeaCom, SeaCom system reference manual
23	2010-01-11-SeaCom-addendum	2011-03-22	Report: Jens-EMC, Extreme and excessive power supply conditions
22	20110329	2011-03-28	Report: EKTOS, Environmental test report, additional tests
10	20100902/CNI	2010-09-02	Report: EKTOS, Environmental test report
7	2010-01-11-SeaCom-Alphaconnect P411/ SeaCom411	2010-07-14	Report: Jens-EMC, EMC test report for SeaCom 411/ Alphaconnect P411
6	2010-01-11-SeaCom-Alphaconnect P220/ SeaCom220	2010-07-14	Report: Jens-EMC, EMC test report for SeaCom 220/ Alphaconnect P220
5	2010-01-11-SeaCom-Alphaconnect P211/SeaCom211	2010-07-13	Report: Jens-EMC, EMC test report for SeaCom 211/ Alphaconnect P211
4	2010-01-11-SeaCom-AC48/SeaCom1000 & AC128/SeaCom2100	2010-07-21	Report: Jens-EMC, EMC test report for AC48/SeaCom1000 &/ AC128/SeaCom2100
3	2010-01-11-SeaCom-Alphaconnect P421/ SeaCom421	2010-07-14	Report: Jens-EMC, EMC test report for SeaCom 421/ Alphaconnect P421

Tests carried out

- Environmental testing: DNV-CG-0339 (Jul. 2021) and IACS E10 (Rev 7.0, Oct. 2018)
- Performance testing: Functional tests according to DNV Type Approval programs 848.25 for Automatic Telephone System and 848.26 for Hands Free Voice Talk Back system.

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Certificate No: **TAA0000054**
Revision No: **2**

Marking of product

The Manufacturer and Type Designation to be applied to the equipment in a clearly visible location. In addition the equipment shall be marked with serial number, safe distance to magnetic compass, power consumption and/or supply voltage.

Periodical assessment

Periodical assessment is to be performed after 2 years and after 3.5 years. A renewal assessment will be performed at renewal of the certificate.
The scope of the periodical/renewal assessment is to verify that the production quality conditions stipulated for the type approval are complied with and that no alterations are made to the product design or its components and/or materials without appraisal by the Society.

This certificate is only valid if required periodical assessments are carried out with satisfactory results. To check the validity of this certificate, please look it up in https://approvalfinder.dnv.com

CONTENT

1. Introduction

Thank You for choosing SeaCom 3000 to cover Your on board communication needs.

This manual will explain how the system is designed, it will explain how each of the components used to form the system are working, and it will take You through how to set-up a system and how to do the system programming.

1.1 Purpose of this manual

This manual is a generic manual meant for salesmen and installers. It cannot be left alone as an end user manual.
It is the responsibility of the installer to create an end user manual, based on the numerous choices made during the design and architecture of a specific installation.

1.2 Liability

The information contained in the manual is distributed on an “As is” basis, without warranty. While every effort has been taken in the preparation of this manual, the manufacturer shall not be held liable with respect to any liability, loss, or damage caused by the instructions contained in this manual. The information contained in this manual is subject to change without notice.

1.3 Revisions

This manual addresses only systems updated in all aspects to the latest versions. If You have old hardware and software, some of the features described might not be implemented.

1.4 IMPORTANT SAFETY NOTES

- Telephone equipment makes use of DC voltages of 48V and AC voltages of 80V. It is the responsibility of the installer to make sure that the electrical parts of our system is properly covered, and that the PE (Protective Earth) is always taken back from all stations to the exchange system and properly wired to ship hull.

- Only authorized personnel should open any of our boxes.

- Always power the system off before handling

- The installer is reliable for proper fusing the system, and for designing the wiring and power supply capacity in a way that ensures that fuses blow first if a short circuit should happen.

- NEVER connect 230V AC to ANY terminal of this system. It is powered by 24V DC only.

- Relays of this product is designed to carry max. 50V, 1A. Make sure that the wires are fused by max. 1A.
  NEVER connect 230V AC to any relay terminals.

- NEVER leave the exchange without the cover put back on.

- The SeaCom 3000 is enclosed by a heavy steel box. Handle this with care, and make sure that the mechanical mounting is strong enough to carry the weight.

- Make sure that the exchange unit is mounted according to the instructions on a vertical
  bulkhead in order to assure proper ventilation of the system.

2. System overview

The SeaCom 3000, together with the communication stations SC211, SC411, SC220 and SC421, forms a full on board telephone, talk-back and intercom system to be used on ships.

It is designed to give seamen easy and reliable on board communication between all essential places on board a ship, as well as giving the possibility of making telephone calls to and from the ship, via satellite communication terminals.

2.1 Operational features

The list below gives an overview of the features to be found:

- On board communication
- Distribute satellite communication
- Talk back command calls
- Public Address (PA)
- Time distribution
- Wake up calls
- Conference calls

2.1.1 On board communication

On board conversations between bridge, ECR, cabins, offices, work shop, deck etc. All locations, no matter if it is dry, wet, noisy or dirty, can be covered by either an plain old analogue telephone or one of our communication stations with handset, headset or loudhailer.

2.1.2 Distribute satellite communication

Calls from the ship to shore can be conducted from all places on board. The seaman can have his privacy by calling the family from his cabin for example.

Calls from shore to ship can be distributed to all locations on board, typically to bridge, captains cabin or to ship office. You don't have to be on the bridge to answer Your incoming satellite calls.

2.1.3 Talk back command calls

Our communication stations SC211, SC411 and SC421 can be used to implement the classical talk-back functions. A call from bridge to mooring stations for example, can be set up from bridge, without hands-on on deck. Groups of stations can be addressed simultaneously when performing a command group conference call.

2.1.4 PA functions

The SeaCom communication system can distribute public address (acoustic paging) to all stations with loudspeakers.
The system gives possibility of defining groups with priority levels. The system can be set up to send a gong signal preceding the spoken message.

PA calls are initiated from telephones and communication stations, and does not require special control stations with goose neck microphones.

2.1.5 Time distribution and wake up

Using telephones with display gives the possibility for displaying the ships clock on the telephones. The time displayed can be set manually by a simple telephone call.

Wake up calls can be ordered from any telephone connected to the system.

2.2 Main system components

A SeaCom 3000 installation consist of the following main components:

- The SeaCom 3000 exchange
- Communication stations SC211, SC411, SC220 or SC421
- Telephones

2.2.1 SeaCom 3000

The SeaCom 3000 is the "heart" of the system.

It connects all stations and telephones by one cable each, and it hold the point of connecting the 24V DC power supply.

Key features:

- 15 simultaneous
- Priority override
- Caller number and caller name display
- Day mode / night mode
- Call transfer
- Wake-up
- Talk-back calls
- Conference calls
- Set date and time from telephone
- Call pickup
- Paging calls to telephones having speaker
- Answer back on PA
- Do not disturb
- Maximum 144 stations
- Power 24V DC main and backup
- Power consumption from 15W
- Analogue extensions lines with:
  - DTMF dialing
  - LD dialing
  - Caller ID
- Analogue trunk lines
  - direct in dialing
- Cable termination inside
- Dimensions 372 x 505 x 179 mm
- Weight 13 kg

2.2.2 SC211

Simple water tight station with 3 buttons. Typically used for deck stations. Also called the talk-back station.

Key features:

- Talk-back substation
- Water tight
- 10W speaker driver
- Power supply 24V DC

2.2.3 SC411

Full featured flush mount station with display, backlight and hands-free. Used on bridge and ECR.

Key features:

- Screw less flush mount
- Red dimmable backlight
- Hands-free operation
- Power supply 24V DC

2.2.4 SC421

Full featured water tight station with display, backlight and hands-free. Connect handset and headset. Used in engine spaces, and on deck.

Key features:

- Water tight
- Red dimmable backlight
- Hands-free operation
- 10W speaker driver
- Handset operation
- Headset operation
- Power supply 24V DC

2.2.5 SC220

Water tight and salt mist resistant industrial telephone with call relay build in. Used in engine spaces and on deck.

Key features:

- 2 wire only
- Handset
- Headset
- Call ringer relay

2.2.6 Telephones

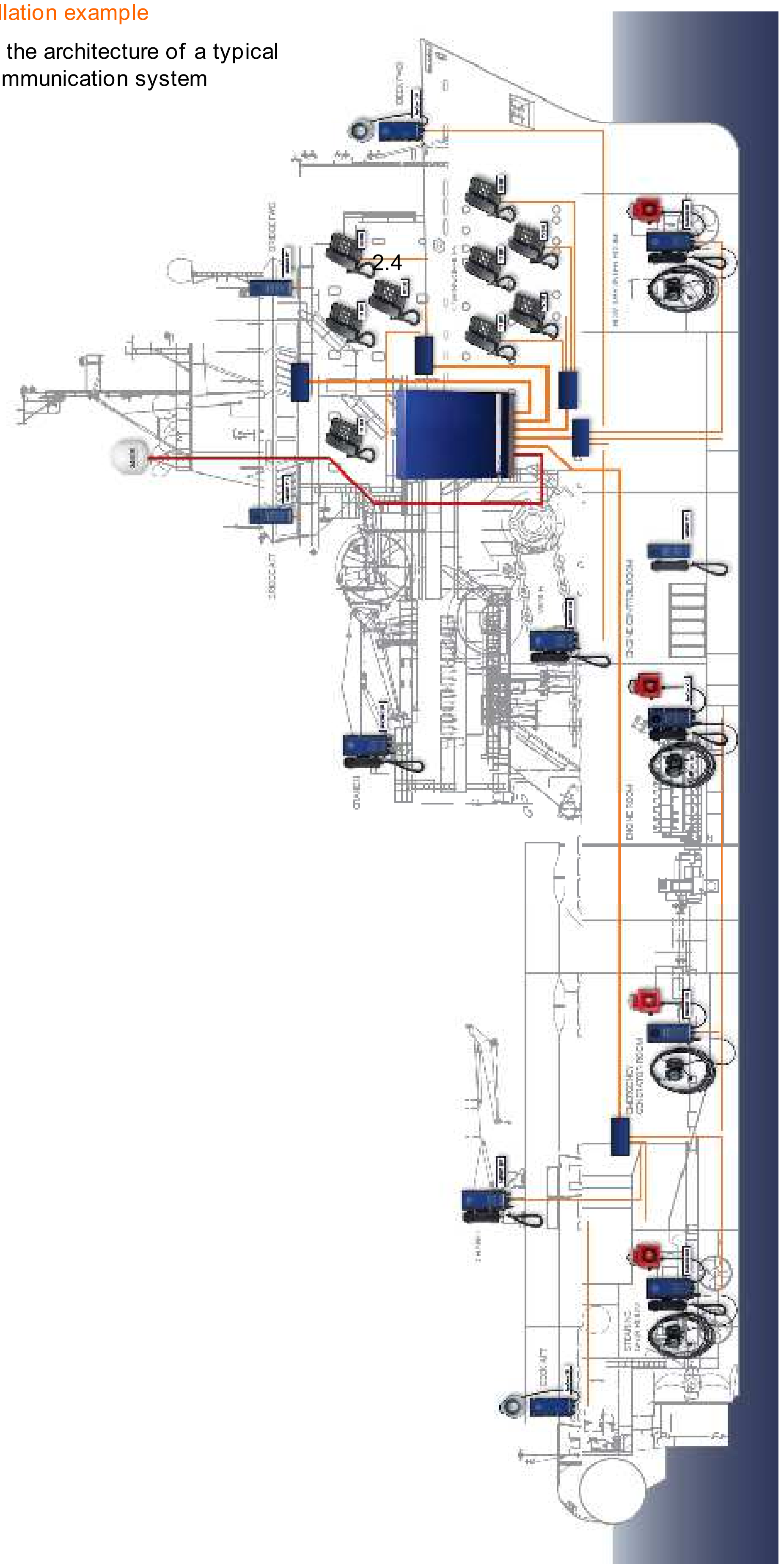
For dry and heated spaces on board SeaCom recommend our digital telephone TX325 or the direct-in/PA enabled SC325 telephone.

They are digital telephones displaying date/time, caller number and caller name. They have hands-free operation and can be either wall mounted or desk top mounted.

Note that these telephones are NOT covered by the DNV-GL type approval.

2.3 Typical installation example

The figure shows the architecture of a typical SeaCom 3000 communication system installation.



3. Designing a system

The intentions of this chapter is to be a guide to how to design an on board communication system, how to choose the right system components, and how to build the exchange system.

It is to be used by the people who makes quotations to ship yards and ship owners, as well as the technicians who is actually assembling and installing the components.

3.1 Analyzing

All ships have different needs. This is what makes design and installation of on board communication challenging. The SeaCom 3000 maritime communication system is a very flexible configurable system which allows You to meet by far the most requirements.

In order to architecture a proper system, You must first collect information about:

- Number of locations that need communication
- The environment to be expected at each location
- Type of communication to be carried out at each location
- Number of Satellite lines to be connected.

3.1.1 Counting locations

The number of locations needing telephones must be determined. This is done in cooperation with ship architect and owner. The number determines the size of the exchange system to be chosen.

3.1.2 Environment

On each location, the expected noise level must be known, and the location must be classified as protected or exposed, where protected areas are locations like cabins, offices and bridge, whereas exposed will be areas like engine room, mooring stations or workshops.

3.1.3 Type of communication

On each location a choice of which kind of communication is most convenient must be made. The SeaCom maritime communication system offers the below communication styles:

- Handset conversation
- Hands-free conversations
- PTT mode (push to talk)
- Headset conversation

Each station can offer the seamen one or more of these styles. It is important to select a style that fits to the needs and which can be operational in the given environment for the intended use.

3.1.4 Satellite lines

Count how many 2 wire trunk lines will be needed for connecting satellite communication equipment.

3.2 Selecting telephones/stations

Based on the analysis, the equipment to be placed at each location must be selected.

The table below shows typical noise levels to be encountered on ships.

Location	Noise level dB
Accommodation	< 60
Bridge	50-60
Engine control room	65-75
Engine spaces	80-100
Steering gear room	100-120
Close to engine or generators	100-130

The table shall be used as a guide only to choose among the numerous possibilities given by the SeaCom 3000 maritime communication system.
The below describes which type of station to be used in each type of environment.

3.2.1 Analogue telephones

Plain analogue telephones are to be used in protected areas only. The TX325, has the below operating range:

Hands free up to 65 dB noise

Handset conversations up to 75dB of noise.

3.2.2 SC411

This station gives the following possibilities in protected environment:

- Hands free with build in speaker and microphone up to 75 dB
- Hands free with external speaker up to 85 db of ambient noise
- Push to talk conversations with external speaker up to 100 dB of noise.

- Handset conversations up to 85 dB of noise.
- Operation with headset in up to 120 dB of noise.

3.2.3 SC211

The talk back station is a station to be used in exposed environment with the below capabilities:

- Push to talk operation up to 100 dB.
- Headset operation up to 120 dB of noise.

3.2.4 SC421

This is the most full featured station, to be used in exposed areas under the below conditions:

- Hands free with build in speaker and microphone up to 75 dB
- Hands free with external speaker up to 85 db of ambient noise
- Push to talk conversations with external speaker up to 100 dB of noise.
- Handset conversations up to 85 dB of noise.
- Operation with headset in up to 120 dB of noise.

3.2.5 SC220

This is a plain analogue telephone with optional handset or headset. To be used in exposed areas.
The capability for operation in noisy areas are:

- Handset operation up to 85 dB background noise
- Headset operation up to 100 dB of noise.

3.2.6 Features overview

The table below shows the features of the stations:

Feature	211	220	421	411
Handset connection		Y	Y	Y
Headset connection	Y	Y	Y	Y
Exposed door option	Y	Y	Y	
Call relay	Y	Y	Y	Y
External speaker	Y		Y	Y
External microphone			Y	Y
Display			Y	Y
Hands free operation			Y	Y
Keys	3	15	21	21
Speed dial			Y	Y
IP class	65	65	65	22
Mounting style	Wall	Wall	Wall	Flush

3.3 Designing the exchange system

The SeaCom 3000 is a scaleable system. It comes with 24 extension lines and 2 trunk lines, and can be build up to 144 extensions.

The components used to build the system are:

- AEXT16 extension line card
- FIO4 trunk line card
- CTU2 Cable termination card
- CTU24 Cable termination card
- PDU Power distribution card

3.3.1 AEXT16

This board adds 8, 16 or 24 extension lines to the system. Each telephone or communication station on board requires one extension line.

3.3.2 FIO4

This board adds 2 or 4 trunk lines to the system. Trunk lines are used to connect to satellite terminals, GSM terminals or shore lines.

3.3.3 CTU2 and CTU24

These boards are used to terminate the ship cables inside the SeaCom 3000. The CTU2 terminates 16 extension lines and the CTU24 terminates 24 extension lines.
When using the FIO4 board, the 2 or 4 trunk

lines will occupy the space of 8 extension lines on the CTU2 or CTU24.

3.3.4 PDU

24V DC has to be distributed to the SC411, SC211 and SC421 stations in use. The PDU allows for this including proper fusing of the lines. 18 stations can be powered by the PDU.

3.3.5 Exchange design example

On a ship we have found out that the below telephones and stations are required:

4 x SC211
2 x SC411
12 x SC421
4 x SC220
35 x TX325

A total of 57 positions

The SeaCom 3000 comes with 24 extensions, so we need to add another 33 extension lines. We choose the AEXT16-24 to add 24 and the AEXT16-16 to add the remaining 9.

Along with the two extra boards, we need a CTU24 for the additional 24 lines and a CTU2 for the 9 lines.

We have 4 + 2 + 12 = 18 positions that require 24V DC, and that can be covered by adding one PDU.

3.4 Power considerations

The SeaCom 3000 and the stations are all powered by 24V DC. This paragraph describes how to calculate the power required.

3.4.1 Power consumption - exchange

The power needed for the exchange system can be roughly calculated from the below formula:

P_{exh} = 15W + 0.25W * N_{ext}

where

N_{ext} is the number of extensions installed.
P_{exh} is the stand by power used by the exchange.

3.4.2 Power consumption - stations

Except for the analogue telephones and the SC220, all the communication stations also require 24V DC power. The power needed depends on the actual operation of communication, where especially the PA calls, which can engage all stations simultaneously, must be taken into account, in order to ensure that the peak power needed can really be delivered by the power supply chosen.

A guideline is:

P_{stby} = 2.0W * N_{st}

P_{peak} = 2.0W * N_{st} + 22W * N_{stpa}

P_{stby} is the idle power used
P_{peak} is the peak power used
N_{st} is the number of stations
N_{stpa} is the number of stations engaged in PA calls

3.5 Wiring schematic

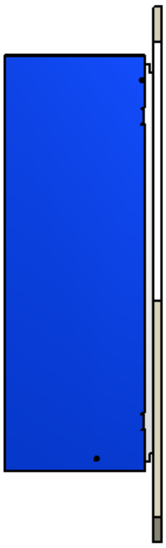
Based on the design of the system, it is the responsibility of the installer to create proper electrical schematic drawings and documentation to be used while installing. This manual is a generic manual, and cannot be an instruction and the documentation delivered to an end user on its own.

4. Installation

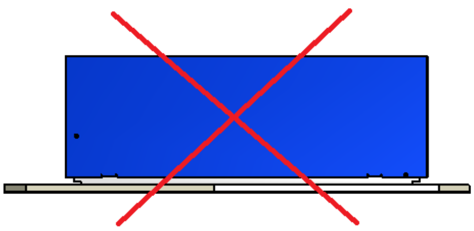
This chapter gives some guidance in how to install the SeaCom communication system. Considerations to be taken, and good practice to follow. For detailed mechanical drawings and details on electrical terminals, refer to later chapters.

4.1 The exchange

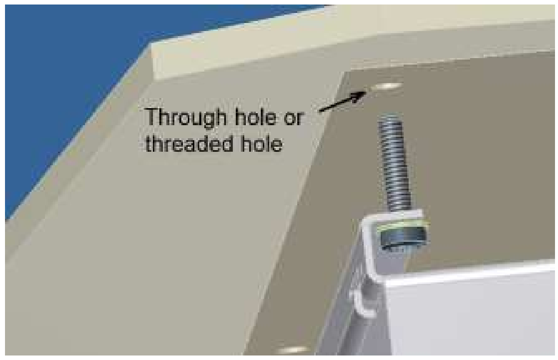
The SeaCom 3000 exchange must be located in a dry ventilated space with a maximum ambient temperature of 55 dgs, the cooler the longer lifetime shall be expected.
The exchange must be mounted on a bulkhead in a vertical style to ensure that the intended air flow can be realized.

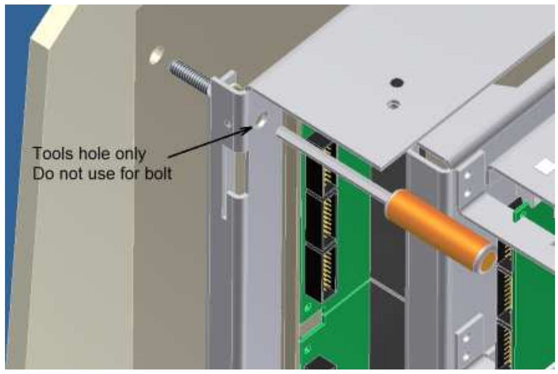


Do not mount the exchange horizontal.



The bolts to be used are 6 to 8mm torx or x-drive, and should be used like shown on the following pictures:



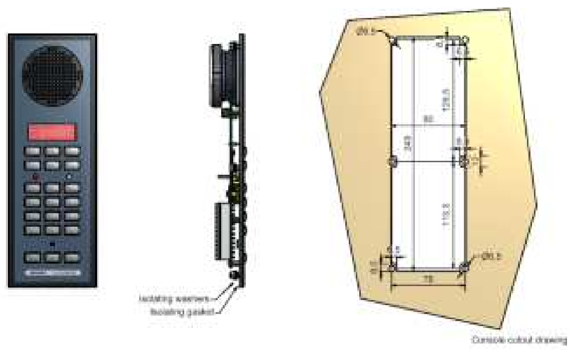


4.2 Mounting the stations

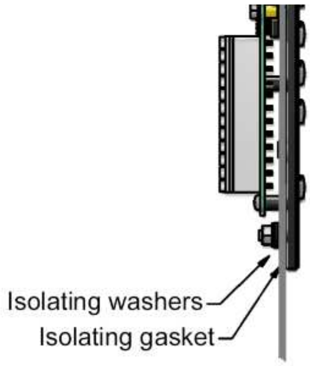
The stations are either flush mount or bulkhead mount.

4.2.1 Flush mount

SC411 is meant for screw less console flush mount.



It is important to make a precise cutout, in order to be able to make the required electrical isolation between the station and the steel of the console.



4.2.2 Bulkhead mount

The communication stations SC211, SC421 and SC220 are all boxed, and mounted on a bulkhead.

The mounting bracket is made of stainless steel, and it is possible to make use of the selection of 5mm mounting screw holes.

The mounting bracket is galvanic isolated from the box itself, so that there will be no galvanic corrosion of the aluminum box.

When mounting the stations care must be taken to use both PG entries in the bottom, or blind the one not used off.
And care must be taken to make a proper cable installation which ensures water tightness of the assembly.

Do not remove the Gore vent found on the lower right side of the stations.

This is meant as a pressure relief valve protecting moisture from entering the stations

4.3 Fuses

It is the responsibility of the designer and installer to make proper fusing. Always use fuses in the power supply circuit that can be blown by the actual power supply used, and make sure that the wires used will stand this current.

4.3.1 Common power supply only

Power to the communication stations MUST be taken from the exchange using a PDU. This ensures that 0V power on the exchange system is also 0V power for the communication stations.

Using local power supply for the communication stations is only allowed with the SC411 and SC421.

4.4 Cables

Choosing the right cables is essential for the system to work properly. All cables used for telephones and stations shall be twisted pair, shielded cable.

The below guidelines apply:

SC411 and SC421 with internal speaker only:

1 mm^2 per 600m length of cable.
max. 1800m

SC211, SC411 and SC421 with external 10W speaker in use:

1 mm^2 per 200m length of cable.
max. 1800m

TX325, TX325D and SC220:

Minimum 0.25mm^2 length of cable.
max. 1800m

When 24V DC power is take to the stations, these shall be done in the same cable as the telephone line. This means that a 2x2wire twisted pair shall laways be used for SC211, SC411 and SC421.

4.5 Shielding and protective earth

When installing the SeaCom 3000 maritime communication system, shielding is of the utmost importance.

The shield of the cables are used for two purposes:

- Reducing immunity to noise from cables taken in parallel with the telephone and communication wires
- Protective Earth (PE) used for making the system safe for the operator to use, protecting from high voltages by accident connected to the system.

4.6 Good practice shielding

The below figure shows schematically the way the shields should be connected. Note that there is only one point of connection to ship's hull / PE.

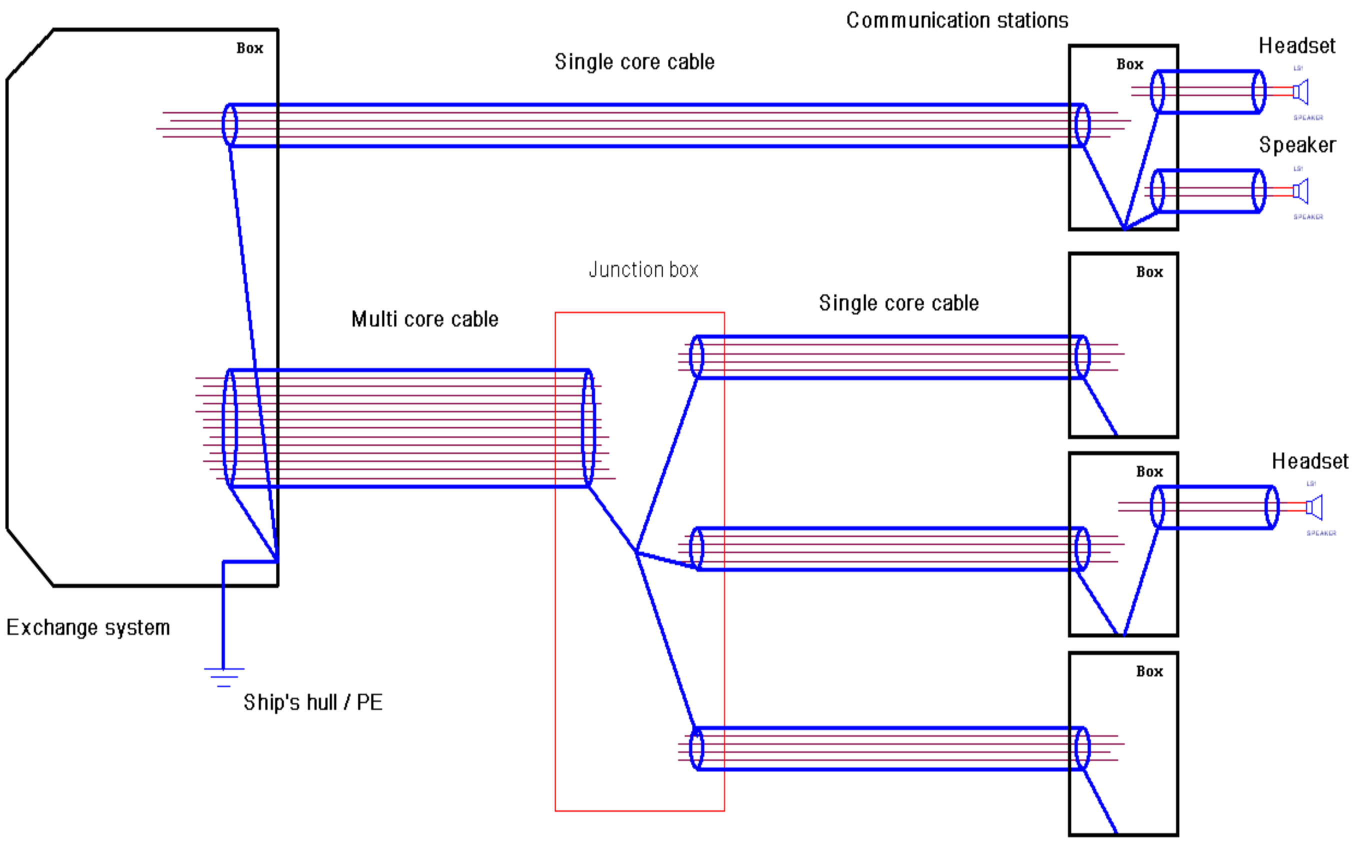
4.6.1 At the exchange

The box of the exchange have PE terminals which shall be connected using a Yellow/Green wire to PE of the ship or ships hull close to the exchange location.

4.6.2 At the stations

At the stations, the shield is taken inside the box and is connected directly to the backside of the enclosure where a PE stay 4mm will be found.
This PE stay is used for connecting the shield of headset cable and speaker cables as well.

Do NOT connect the PE terminal of the stations to ship's hull.



5. Operation

This chapter is a generic user manual, which tells about the way the system works, and the way system functions behave. The chapter is to be used as a starting point for making the end user manual, which must be delivered with each installation.

5.1 Making and transferring a call

Making calls are made as with any telephone system. Lift the handset of a telephone and dial the number that You want to call.
Terminate the call by hanging up again.
When in conversations, calls can be transferred by pressing the R button, which makes the system present a new dial tone upon which You can dial the number to transfer a call to.
A transfer can be made with or without presentation.

5.2 Ringing signals

The SeaCom 3000 system generates 3 types of ringing signals to extensions and communication stations:

Incoming call from trunk line
(1s ringing with 5 s pause)

Local call
(2 short ringing with 5s pause)

Talk back call
(3 short ringing with 5s pause)

Conference
(2 short one long with 5s pause)

5.3 Tone signals

The SeaCom 3000 system generates 3 types of tone signals heard when making calls.

Ringing to B extension
(1s tone with 5 s pause)

Busy or error or wrong number dialed
(intermittent tone)

Transfer caused by Do-Not-Disturb
(3 short tones)

5.4 Priority call

The priority call number is the * (asterisk) followed by the call number to interrupt.

Example: *20

If a call is made to an extension that is occupied, then hang on and make a new call to the extension where the dialing is preceded by the *. This will make a priority call to the extension. The ongoing conversation is parked, and the breaker can speak to the interrupted extension. When the priority call has ended, the interrupted conversation is re-established.

5.5 Direct in

The Direct-in feature can be used with the intercom stations SC211, SC411, SC421 and the SC325.
It is made by dialing the priority call number * twice followed by the call number.

Example: **20

Such a call will make the called station hook off without any hands on by the receiver. The SC411 and SC421 enters hands-free mode and conversations or listening is effective.
Direct in to the SC211station is to be considered a single station PA call, as it is only possible to speak from the caller to the SC211.

5.6 Direct in to ringing group

A special feature is direct in calls to a ringing group. Also only with SC211, SC411 and SC421.

When a bridge is equipped with more than one station with the direct-in call possibility (typically SC411s), and they are all included in a ringing group which is used for calling the bridge (example: number 9). Then a direct in call to the bridge by dialing **9 will be directed to the last one of the bridge telephones that called number 9.
It is thus possible for the seamen to chose an

active position to receive direct in calls, simply by indicating activity by dialing the bridge group call number.

5.7 Setting system time

The system time of the SeaCom 3000 system can be adjusted by a calling the below numbers:

#120 + 8 digits date (YYYYMMDD)

#121 + 4 digits time (HHMM)

If the time is valid, a dial tone will be heard. If the time is not valid the busy tone will be heard.

5.8 Ordering a Wake-up Call

Wake up calls can be ordered by the below call numbers:

#110 Cancel wake up

#111 + 4 digits (HHMM) Order wake up

#112 + 4 digits (HHMM) Order repeated

Wake-up can be ordered to be executed only once, or daily wake-up can be ordered by using the repeated call.

When the wake up time has arrived, a call from the system to the extension will be made. The display (if any) will show "Wake-up call". If the wake-up is acknowledged by is lifting the handset during the wake-up call, the call will be terminated.
If a wake-up call is not acknowledged in this way, it will be repeated after 5 minutes. This will be done 2 times, after which the wake up call is terminated.

5.9 Ringing groups

If a bridge has 3 telephones, these can be included in a ringing group. (example call number 9). When number 9 is called, all 3 telephones on the bridge will ring. The first one picked up will answer the call, which in turn stops the ringing on the 2 other phones.

If one of the bridge telephones is busy, and a call to the group comes in, it will be heard on the bridge that a second call is coming in. In this situation it is possible to hang up on the busy phone, which shortly after will start ringing in parallel with the other telephones of the group. The call can be answered just by lifting off again.

5.10 Paging calls

Paging calls can be received by the communication stations with speakers - SC211, SC411 and SC421 only.

Dial #08 which turns on a selected set of stations and plays the ding dong sound to these stations. Hereafter a spoken message can be send to these stations.

Paging calls can be made from any extension or station given the privilege to do so.

5.11 PA group conference

The PA calls can be answered by any telephone connected to the PA groups. All stations of a system can be assigned to a PA group, and thereby be able to answer back on a PA call.
When a PA call is answered back by one of the called parties, the PA call can stay open so that all parties receiving the PA call will listen to the conversation between the PA initiator and the PA answering back station. This is called a PA group conference because any number of stations can participate. While having a PA group conference, any of the parties listening can take over the right to speak to the PA initiator, just by hooking off. If the parties called is of type SC411, pressing the microphone key on such station will answer the PA call, and releasing will turn the station into listening.

5.12 Alarm calls

Alarm calls can be received by the communication stations with speakers - SC211, SC411 and SC421 only.
Dial #09 which turns on a selected set of stations and plays the alarm sound to these stations. If the alarm shall be interrupted by a spoken message, this can be done by pressing the R button. This will stop the alarm tone, and the spoken message can be send. When pressing the R button once again, the alarm tone continues. Alarm calls can be made from any extension or station given the privilege to do so.

5.13 Call pickup

When a ringing telephone is heard on board, dial #05 to pick up the call.
Calls incoming from satellite services can be picked up, as well as local on board calls.

5.14 Music when free

If an audio in channel is configured, the SC411 and SC421 can call the music when free call number #061, which makes the station open the speaker and listen to the music input channel. If a call comes in or the keyboard is activated for an call, the music is halted. When the conversation is ended, the music channel is opened again until the call number #060 is used to cancel the music when free option.

5.15 Talk-back command calls

This is the classical “talk-back” call.
The call number #07 implements a “semi-duplex conference” call, which is used to implement this feature.

If the call is made from a SC411 to one or more SC211s, the speech direction of the call can be controlled only by the microphone key of the SC411 initiating the call.

The command call can be set up as a fixed group or as a dynamic group, where the first has the SC211 participating pre-programmed in a list, whereas the last type gives the user the possibility of choosing the participants when making the call.
In both cases the initiator is the master, and the SC211s are all slaves.
The master will be in conversation with the primary slave, which is the one that latest have pushed a PTT button or which is chosen by the master to be the primary slave part, by the masters press of one of the number keys representing the station.
The master of the command talk-back call will control the speech direction by pressing or releasing the microphone key, or a slave will push the PTT button and thereby force itself to be the primary slave part speaking. All SC211 not being the primary slaves will just listen to the conversation, and can hear both the master and the primary slave.

A Semi-duplex conference call can be initiated from one of the SC211 substations. In such case a conference master of type SC411 must be defined in the system programming. When the SC211 makes the call to the conference group, the master station will be ringing. If the call is answered by the master station, the full conference group will be connected, just as if the call was made by the conference master itself. During such “reverse” talk-back call, the master controls the conference speech direction by the microphone key, and substations can be selected by the numerical keys or the substations can achieve the right to speak using its PTT button.

5.16 Conference

A conference call having up till 11 members can be made. The system can handle **one conference only.** In a conference call, all participants can speak and listen at the same time. To be able to make conference calls, a call number #07 must be programmed.
To start the conference, on extension must be the master or initiator. When the initiator calls the #07 conference number, a call is set up to the participants. During a conference call, participants can leave or join the conference.
An extension that like to join a conference shall call the #07 call number.
When the initiator leaves the conference, the full conference is closed.
A conference can be set up to a fixed group, or the members can be selected by the initiator by dialing a list of participants.
A conference can be set up so that the participating extensions will ring until answered, or direct-in can be chosen so that the conference is set up automatically to all participants beeing able of receiving direct-in calls. (SC211, SC411, SC421 and SC325).

5.17 Do not disturb

Dial the #02 call number to make the telephone enter do-not-disturb mode. In this mode calls are denied, and the caller will hear the 3 short tones, indicating that the person called like to be left without telephone calls.
The do-not-disturb mode has a timeout of 8 hours, after which normal mode is re-entered.
In do-not-disturb, calls can be transferred to another extension, either fixed or predetermined. It is also possible to enter do-not-disturb mode with the exception that certain extensions can still call in.

5.18 Day-mode night-mode

Call #031 to activate night-mode, and call #030 to activate day-mode.
This is a highly flexible functions typically used to set where incoming calls are directed during day time and during night time.

5.19 Calls via trunk lines

Outgoing calls, via satellite or shore lines, are made by first dialing the call of trunk line connected to the communication equipment to be used. Trunk lines have a default call number of *00.*
If the trunk is free, the connection to the external communication equipment is established. The dialing instructions applicable to the external communications equipment must then be used.

6. SeaCom3000

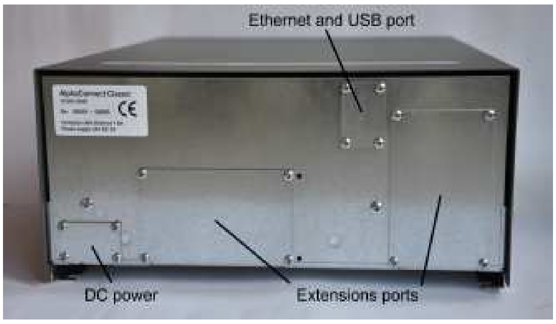
This chapter is a description of the SeaCom 3000 exchange system and the circuit boards to be used with it.

6.1 SeaCom 3000

The SeaCom 3000 comes in an elegant steel cabinet:

6.2 Cable entries

When seen from below, the 4 cable entries can be seen:



Leftmost is the entry for the two 24V DC powers.

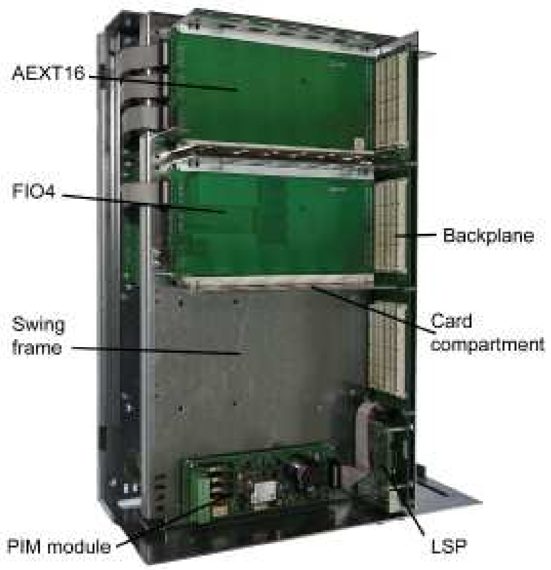
The two large ports are used for extension cables, and can either be opened totally, or cable glands can be used.

Topmost is the Ethernet and USB port. This port is opened when programming or service have to be undertaken.

NOTE: THIS PORT MUST BE LEFT CLOSED ALL TIME WHEN NOT USED.

6.3 The system inside

The figure below shows the system inside as delivered.



The system is two layered. The figure above shows the swing frame layer, holding all circuit boards.

Bottommost is the Power Input Module – PIM. This is the place to connect 24V DC power.

The backplane board holds connectors for interconnecting extension boards and trunk boards. A total of 9 slots are available.

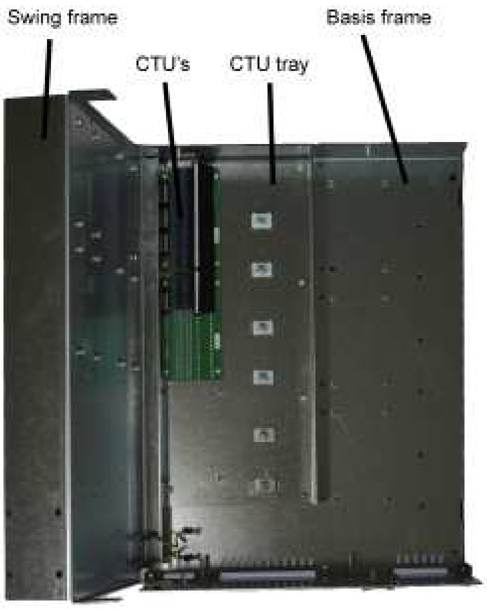
3 card compartments are available, out of which only 2 are occupied in the basic system.

Topmost is the Analogue Extension Card - AEXT16-24. This is the card implementing the 24 extension lines.

In the card compartment below is the Flexible Input Output card - FIO4-2 – found, which implements the 2 trunk lines and 2 audio out ports.

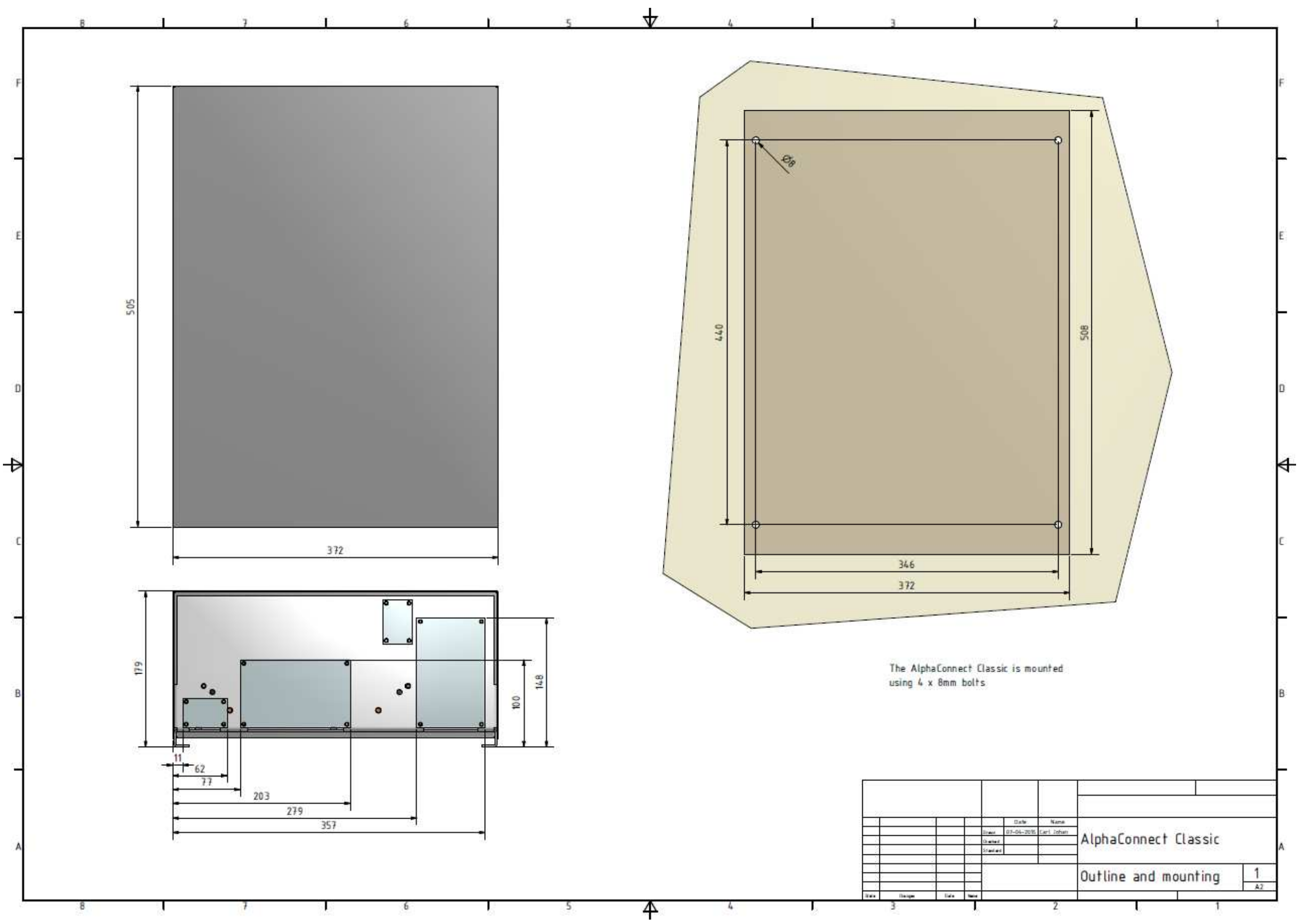
At the bottommost part of the backplane, the LSP (Linux System Processor) is found. This is the circuit board being the very brain of the system by running the CP software. This board holds the Ethernet port and the USB port used for system programming.

The figure below shows the system when the swing frame is taken out.



The basis frame is revealed. The basis frame is used to hold the Cable Termination Units – There are 6 CTU positions, out of which 2 is used in the basic SeaCom 3000 system. One CTU board terminates up to 24 wire pairs. The CTU's are connected to the circuit boards of the swing frame layer via ribbon cables. In order to be able to use the rightmost 3 CTU positions, the leftmost CTU positions are lifted by the CTU tray, and the ribbon cables are taken below the tray to reach the right located CTU's.

6.4 Mechanical dimensions



6.5 Circuit boards.

This chapter holds detailed technical information about all circuit boards used in the exchange systems.

The following boards are found:

- AEXT16
- FIO4
- PIM
- CTU24
- LSP
- Backplane

6.5.1 The AEXT16

The picture below shows the **A**nalog **Ext**ension card:

The AEXT16 is the telephone line card of the system. The AEXT16 that comes with the SeaCom 3000 has 24 lines, but the AEXT16 line card comes in 8 or 16 lines versions as well, in order to more closely adapt the need of the single customer.

6.5.1.1 Board layout

The board layout is shown below:

To the left, the 3 lines connectors are found. These have the 2 wire analogue telephone lines.

To the right, the 64 pole DIN connector is found. This the connector used to interconnect the system via the backplane. This connector carries 24V DC power, PCM data streams, reset signals and inter board communication.

6.5.1.2 Jumpers

If You are adding cards to an SeaCom 3000, this paragraph has no relevance, as new board do not have any jumpers to be set.

On the board there is a set of jumper fields used to define the operation of the board.

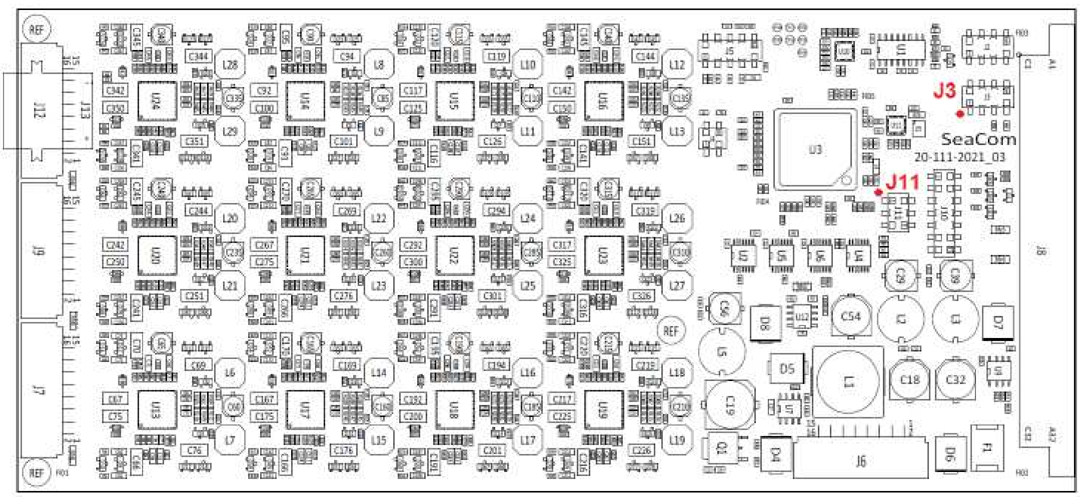
One, and only one, board of an SeaCom 3000 system must behave like a master board. The master must be located in backplane slot 0 as well.
This board is responsible for generating the necessary clock signals on the backplane and as being the center of all inter board communication, including the communication the CP/LSP, and it must have J11 jumpers and J3 jumpers set correctly to work:

J11 Master functions:

1,2 (red dot)	Master
3,4	C2,C4
4,5	PSU2

J3 CP/LSP communication:

1,2 (red dot) 3,4	RS232 level Used with CP2
5,6 7,8	0-5V level Used with LSP



6.5.1.3 Indicators

On the backside of the AEXT16, just below the 3 lines connectors, the line activity indicators are found. These are yellow led's that will show activity on the line. Indicators are arranged so that they are aligned with the two pins of the connector which they shows the activity off. This to make debugging an installation easy.

The following code for the indicator:

- off	line is idle
- flash 1 Hz	wait for B answer
- flash 2 Hz	dialing
- flash 4 Hz	Extension is ringing
- on	conversation

6.5.2 The FIO4

The picture below shows the **F**lexible **IO** card:

The FIO4 is the trunk line card of the system.

FIO4 line card comes in 2 or 4 lines versions. The SeaCom 3000 has 2 trunk lines.

The FIO4 card, also implements 2 Audio I/O lines and a couple of relay outputs and two digital inputs.

To the left, the 3 connectors are found. These have the 2 wire analogue telephone lines, together with all the audio I/O, relay contacts and digital inputs.

To the right, the 64 pole DIN connector is found. This the connector used to interconnect the system via the backplane. This connector carries 24V DC power, PCM data streams, reset signals and inter board communication.

6.5.2.1 Jumpers

If You are just adding cards to an existing SeaCom 3000, this paragraph has no relevance, as new board do not have any jumpers to be set.

A described for AEXT16, one card must be master. The FIO4 can act as master, and if is has to do so, the jumpers below must be set correctly.

J11 Master functions:

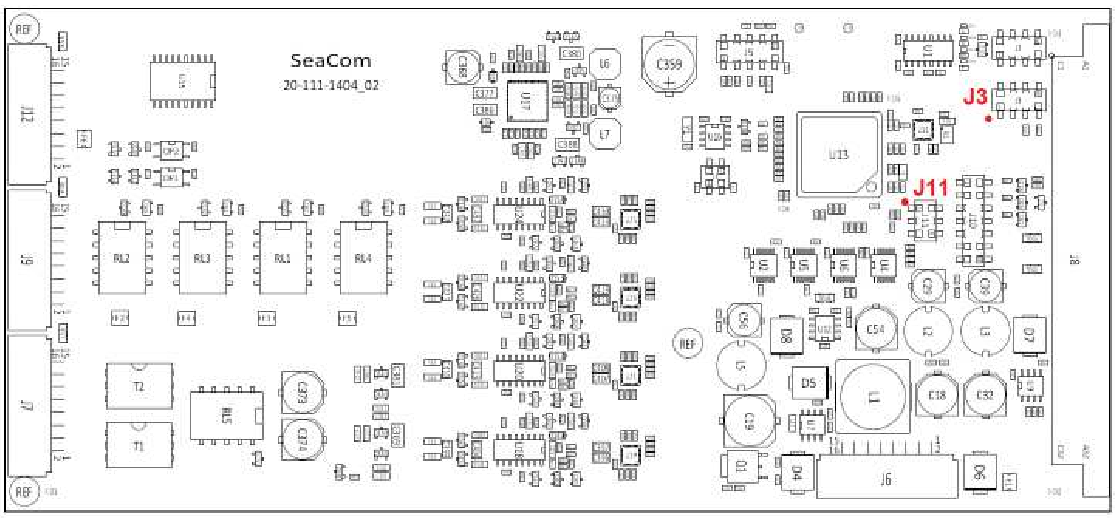
1,2 (red dot)	Master enable
3,4	C2,C4 to backplane enable
4,5	PSU2 communication enable

J3 CP/LSP communication:

1,2 (red dot) 3,4	RS232 level Used with CP2
5,6 7,8	0-5V level Used with LSP

6.5.2.2 Board layout

The board layout is shown below:



6.5.2.3 FIO4 Connectors

The 3 tables below shows the pin out of the 3 lines connectors.

J7 Trunks and audio I/O:

Pin 1,2	Trunk line 1
Pin 3,4	Trunk line 2
Pin 5,6	Trunk line 3
Pin 7,8	Trunk line 4
Pin 9,10	Audio I/O 1 (parallel J9)
Pin 11,12	Audio I/O 2 (parallel J9)
Pin 13,14	Relay contacts 1 (parallel J9)
Pin 15,16	Relay contacts 2 (parallel J9)

J9 Audio I/O, relays and digital in

Pin 9,10	Audio I/O 1 (parallel J7)
Pin 11,12	Audio I/O 2 (parallel J7)
Pin 13,14	Relay contacts 1 (parallel J7)
Pin 15,16	Relay contacts 2 (parallel J7)
Pin 9,10	Relay contacts 3
Pin 11,12	Relay contacts 4
Pin 13,14	Digital input 1
Pin 15,16	Digital input 2

J12 Zone relay control

Pin 2	Zone relay 1 drive
Pin 4	Zone relay 2 drive
Pin 6	Zone relay 3 drive
Pin 8	Zone relay 4 drive
Pin 10	Zone relay 5 drive
Pin 12	Zone relay 6 drive
Pin 14	Zone relay 7 drive
Pin 16	Zone relay 8 drive
1 and 3	24V DC 100mA out
13 and 15	GND for relays and 24V DC

6.5.2.4 Trunk lines

The FIO4 implements good old 2 wire trunk lines with the following main specifications:

- Line feed 24 to 48V DC
- 20 mA off hook loop current
- DTMF dialing in and out
- Dial tone detection
- Galvanic isolation up to 1,5kV

6.5.2.5 Audio I/O

The two audio I/O are galvanic isolated audio transformer type I/O:
Main specifications are:

- 600 ohm
- Bidirectional
- 0dB level (770mV)
- Galvanic isolation up to 1,5kV

6.5.2.6 Relays

All relays dry contacts galvanic isolated from everything else. Main specifications are:

- stands 48V DC
- Fused by 500mA auto reset fuse
- Galvanic isolation up to 1,5kV

6.5.2.7 Digital input

The two digital inputs are opto isolated inputs. They require 24V DC between the input terminals to be activated. Main specifications are:

- 15 - 48 V DC input for activation
- Galvanic isolation up to 1,5k

6.5.2.8 Zone relay drive

These are 0-24V DC drive outputs. Their intentional use is to drive external relays that can be used to switch 100V line speaker signals. This can be used for forming zoned PA systems.

6.5.2.9 Indicators

On the backside of the FIO4, just below the 3 lines connectors, the line activity indicators are found. These are yellow led's that will show activity on the line. Indicators are arranged so that they are aligned with the two pins of the connector which they shows the activity off.

Indicators will also show when relays are activated and when digital input receives signal.

6.5.3 PIM

The picture below shows the **P**ower **I**nput **M**odule:

The PIM is used to connect 24V DC to power the system. It holds the main fuses, and it holds the switch over circuit responsible for switching to the backup power supply when main power fails.

6.5.3.1 Board layout

The board layout is shown in figure below.

To the left on the board, the power connector is found. This takes two inputs: the main bower input and the backup power input.
We have one power output , which is used to take power to distribution units for powering the intercom stations used with the system. In this way these are also switched over when power fails. A maximum of 10A input and output is allowed.

The alarm relay output is a dry contact output which is normally closed when system is running OK. If any errors occurs, or switch over to battery occurs, the relay contact will open and the alarm indicator will turn on.

On the PIM module two power converters are also found making +-5V auxiliary power to the backplane. These voltages are used by the LSP and by FIO2 when used.

24V DC and the auxillary voltages are taken to the backplane via J6 rightmost on the board.

6.5.3.2 Power connector

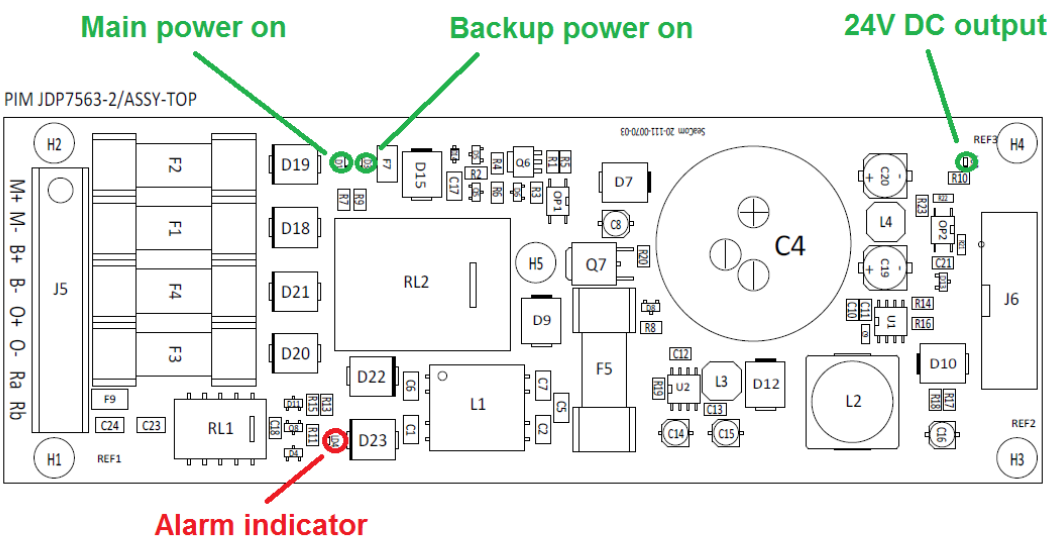
M+	Main 24V DC input positive terminal
M-	Main 24V DC input negative terminal
B+	Battery 24V DC input positive terminal
B-	Battery 24V DC input negative terminal
O+	24V DC Output 10A max
O-	24V DC Output 10A max
Ra	Alarm relay
Rb	Alarm relay

6.5.3.3 Fuses

F2 and F2	Main 24V	10A
F3 and F4	Battery 24V	10A
F5	24V out to backplane	2A

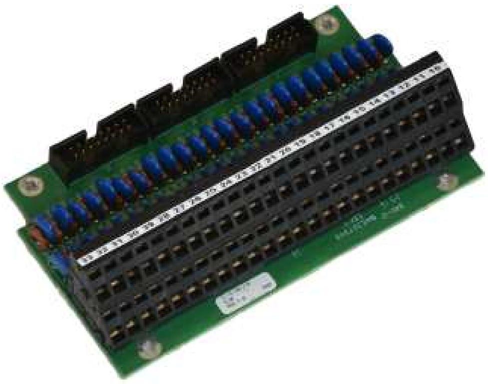
6.5.3.4 Indicators

There are indicators showing the presence of powers, and one indicator showing the state of the alarm relay.



6.5.4 CTU24

The picture below shows the **C**able **T**ermination **U**nit CTU24:



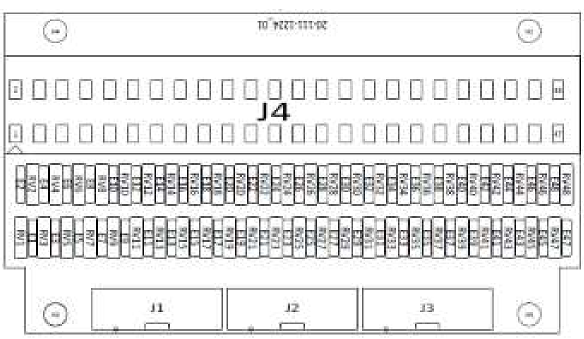
The CTU24 is used to be the interface between the ribbon cables and the ship cables. And it takes care of being the EMC barrier of the system shorting transients and lightening to the basis frame of the cabinet.

The CTU24 comes in two variants: 24 lines and 8 lines.

6.5.4.1 Board layout

The board layout is shown in the figure below:

The topmost connector, J4, is the ship cable connector, and the bottommost J1, J2, J3 are the ribbon cable connectors



6.5.5 LSP

The picture below shows the LSP (**L**inux **S**ystem **P**rocessor).

The LSP is a power full 64bit quad core 1.2 GHz computer, that takes care of being the "brain" of the SeaCom 3000.

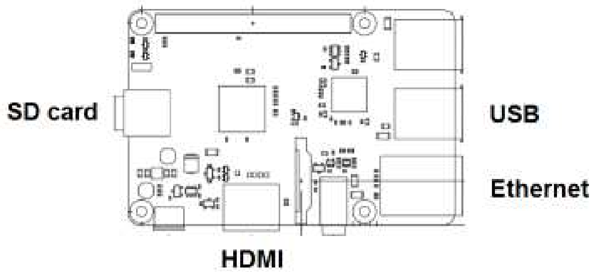
It executes the **Processor.exe** software, and holds the **Configuration_file** which holds the number list of the system.

The computer is running Linux.

Refer to later chapter “LSP in details” for information regarding operating system and disk layout and use.

6.5.5.1 Board layout

The board layout is shown below:



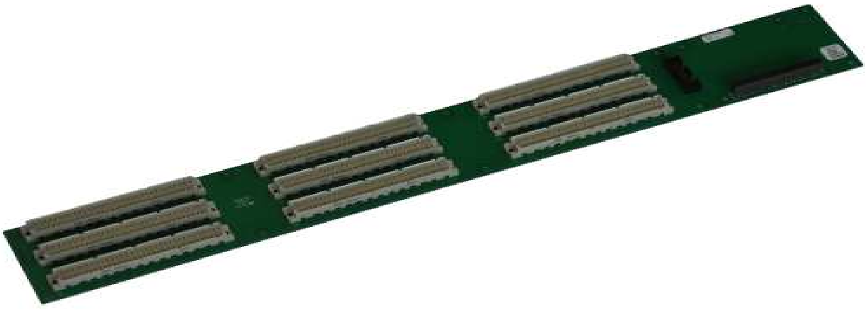
To the left, the SD card connector is found. this card is the "hard disk" of the system, holding the Linux operating system, and all the required system software and files.

To the right the Ethernet port and the USB port are found. These are our main interfaces when the system programming is undertaken.

As an option, also a HDMI connector is found.

6.5.6 Backplane

The picture below shows the backplane:



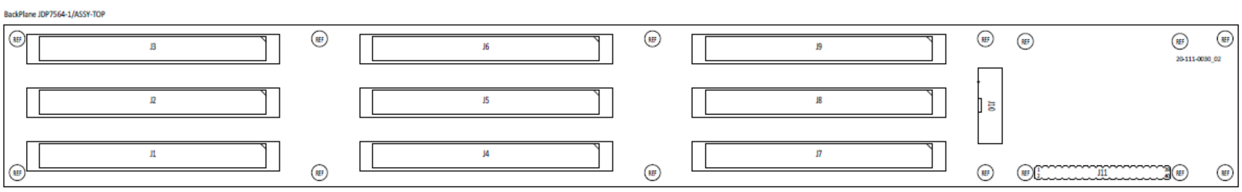
The purpose of the backplane is to interconnect all circuit boards, and to distribute power. The backplane also carries the LSP board.

9 card slots are available.

6.5.6.1 Board layout

The figure below shows the board layout.

J10 is the 24V DC power input, J11 is the LSP board connector, and the 9 large connectors are all for circuit boards in the 3 circuit boards compartments



7. Communication stations

The SC211, SC220, SC421 and the SC411 are communications. These communication stations share the same mechanical layout principles , the same PCB base design, the same connector pin-out and the same keyboard layout, which makes interchanging and reuse possible and easy.

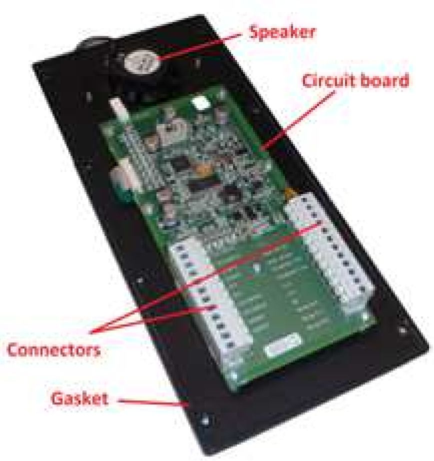
A general introduction to the common mechanics of the SC211, SC220, SC421and the SC411 flush mount stationis found in this paragraph,

7.1 Mechanics and mounting

The enclosure of the stations consist of a back part, which can be mounted on a bulkhead, and a front part carrying keyboard, speaker, microphone etc. depending on the actual station. The box is made of sea water resistant aluminium, which is passivated by anodization. Below picture shows a station in its basic configuration.

7.1.1 The keyboard

Next picture below shows the front part as seen from behind, showing the circuit board, connectors, speaker and the very important gasket making the whole thing water tight when assembled.



The picture show the connectors without the cable part to which the ship cables are connected.

7.1.2 Ins and outs

On a basic station, the below ins and outs are found:

The box is watertight due to the use of PG cable entries with O-rings. Openings optionally used, are on delivery closed by grommets, which are removed when the handset or headset is to be attached to the station. There are two optional cable inlets for the handset and for the headset water tight connector set.

The two main cable entries are used for taking the cable from the exchange through, and for connecting a wired directly headset, an external speaker or a horn/flash light.

Blind off the one that is not used if only a single PG glands is in use.

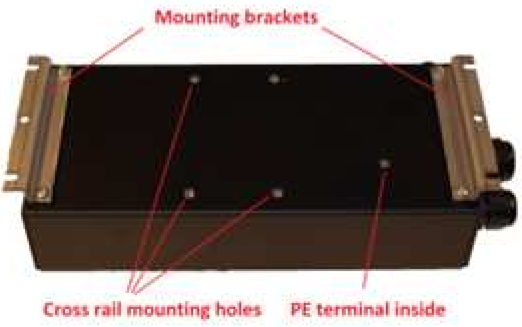
7.1.3 Gore vent

In order to make a pressure relief when temperature changes, a Gore vent is used. This vent is essential for keeping the interrior of the box free from moisture, as it allows dry air to enter the box wereas damp air is taken through to the outside. For the vent to be effective, all gaskest, grommets and PG glandas must be properly mounted.

Do not remove the vent or use the hole for cable entry.

7.1.4 Mounting brackets

On the backside of the box two stainless steel mounting brackets are found.



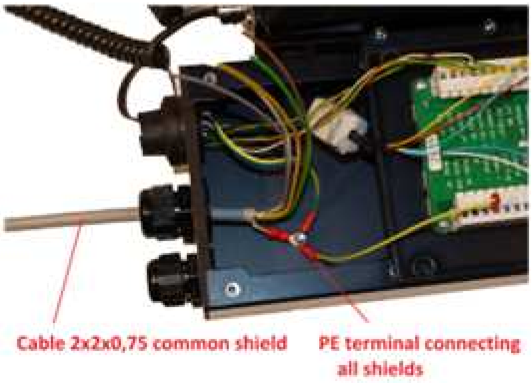
These are used for hanging the station on a bulkhead. In order to make it legal to terminate the cable shield to the PE terminal of the encapsulation, the two stainless mounting brackets are isolated from the aluminum box by isolating washers as seen on the below picture.

Do not corrupt the isolation.

7.1.5 Cables and shielding

Recommended signal cable to the exchange is 2x2x0,75 twisted pair cable with common shield. If a station has a speaker or loudhailer connected, the cable must be shielded. And if a headset is connected via one of the cable glands, the headset cable must be shielded.

In all tree cases, the shield of these cables must be connected to the PE terminal inside the enclosure as shown on the picture below.



Refer to chapter 4.5 discussing shielding.

The PE terminal should NOT be connected to ships hull near station

7.1.6 The silica gel bag

When installing the enclosed stations, a silica gel bag is put inside the station to remove moisture entering the enclosure. The enclosures are perfectly damp and water tight, and the silica gel bag is only a safeguard extending station lifetime.

The silica gel bag is delivered in a small plastic bag, and is activated by opening the bag an putting the silica gel bag – as shown on the picture – inside the station cabinet.

7.1.7 The handset

The range of stations can be connected to our water resistant and salt mist tolerant handset. The handset is an option, and has to be ordered separately.
The handset has a build in magnetic reed hook switch, and is placed in a cradle fixing the handset even on ships in rough seas.
The. Handset has a 300 ohm speaker and a 300 ohm microphone, both dynamic types.

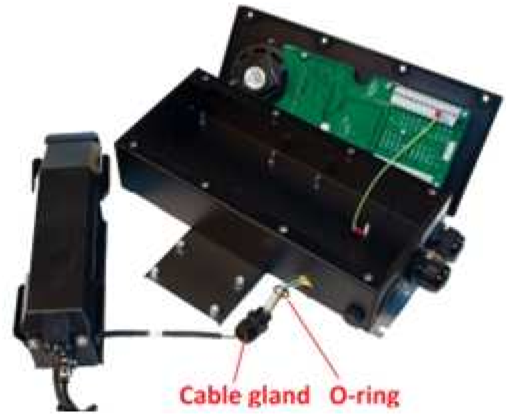
The handset can be mounted on the box of the stations using a cross rail fixing bracket in order to form a station with handset.
The picture below show such.

7.1.7.1 Mounting the handset

First mount the cross rail on the back of the station that need a handset. The cross rail is ordered separately.

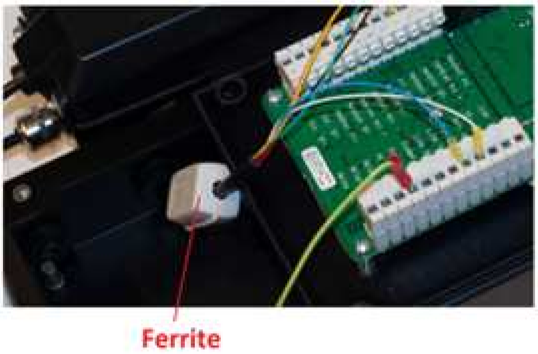
Use the 4 screws and the 4 nylon washers. To fasten the cross rail on the backside of the aluminum box.

Next put on the PG9 cable gland and the O-ring. Pass the cable through the handset cable hole.



Make fast the handset to the cross rail, using the 4 screws and the nylon washers.

Tighten the PG9 cable gland so it forms a water tight cable pass through, and put on the ferrite delivered with the handset.



Connect the handset to the connectors of the circuit board. The use of the wires are:

Black / Red	-	Microphone
Green / Rellow	-	Earpiece
Blue / White	-	Hook switch

7.1.8 Headset

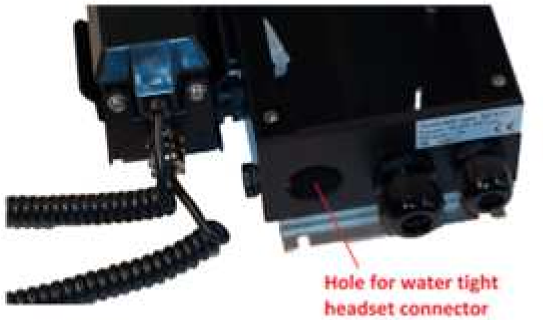
All stations connects a headset with a PTT button and 10m cable. 300 ohm speakers, 300 ohm dynamic microphone.

The headset cable can be taken directly into the stations via one of the PG glands, or the optional water tight headset connector set can be used. This set, which has to be ordered separately, makes it possible to disconnect the headset and stowe it when not in use or easy replacement when worn out.

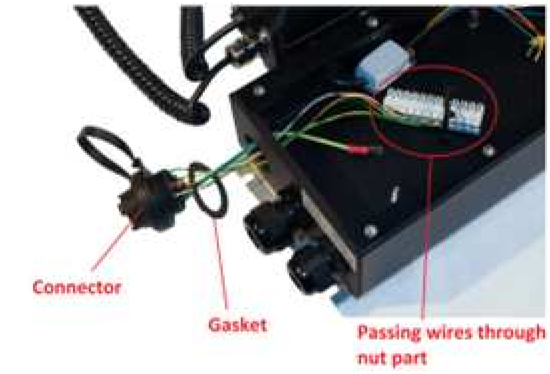
7.1.8.1 Mounting headset connector

The figure below shows the connector set:

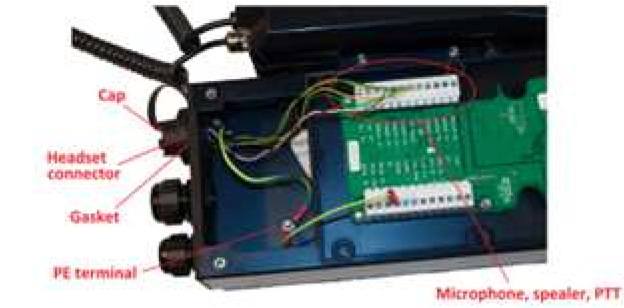
Remove the grommet covering the headset connector hole.



Pass the white connector through the hole, and make sure that the gasket is in place.



Tighten the nut, connect the PE terminal and connect the PCB connector part to the PCB.



The headset with the cable part water tight connector is now ready to be attached to the station when installed.

7.1.9 Exposed equipment

The SC211, SC220, SC421 can be used in exposed areas when the optional front covering door is mounted on the station. This door gives additional mechanical robustness while still being acoustical open, so the SC421 can be used in hands-free. With the door mounted, the stations can withstand the effects of light breaking seas heavy rain and splashing. The door must be opened to operate the station

The door is mandatory when the stations shall be classed exposed

The door can also be used to give additional mechanical robustness also in areas not affected by waves and salt water.

The picture below shows a station with door, closed and open for operation.

7.1.9.1 Mounting the door

The door comes as a kit, just like the handset. The kit contains the door and screws for mounting it. First assemble the black balls and the studs as shown below.

Then mount the stud in both top and bottom of the station in the left thread hole.

And finally mount the door using the black and white bushing and washer as shown below. One mounting in the bottom and one in the top of the station.

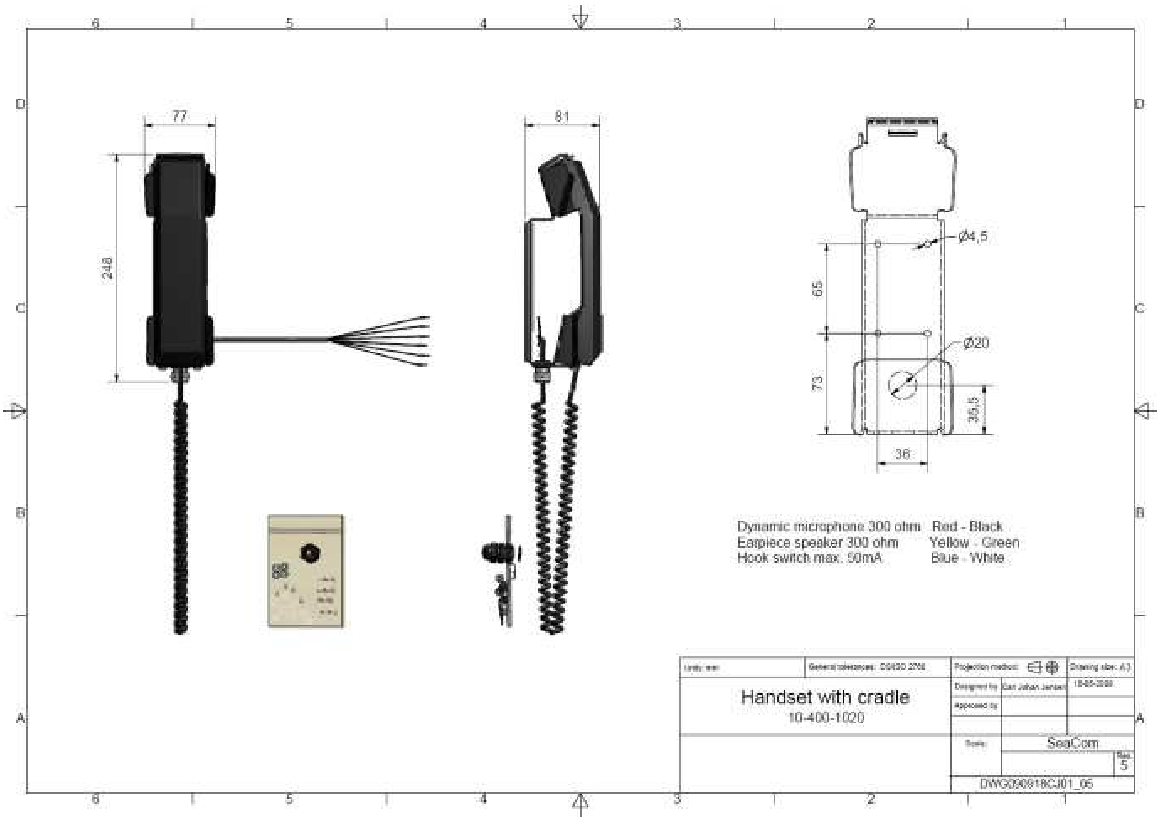
Close the door and make sure that it closes with a robust click. The door is held closed by the spring force of the tongue and the black ball.

7.1.10 Features overview

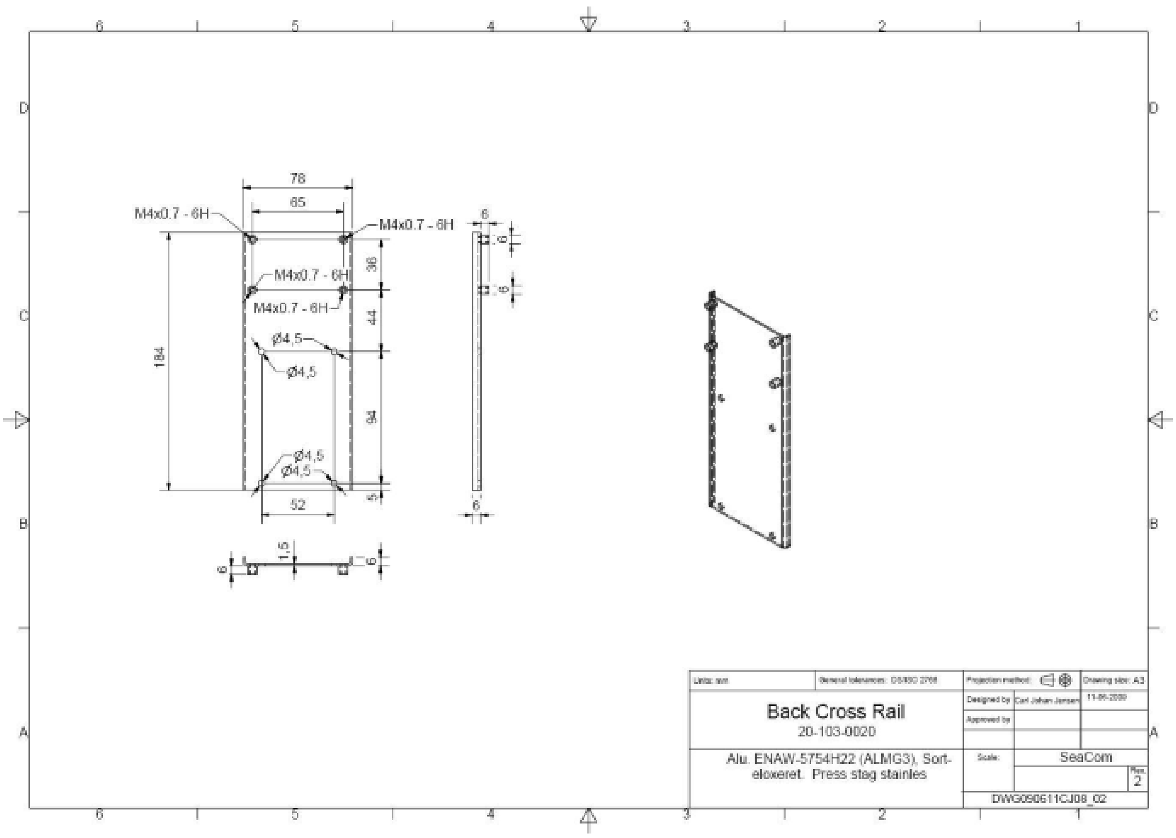
The table below shows the features of the stations:

Feature	211	220	421	411
Handset connection		Y	Y	Y
Headset connection	Y	Y	Y	Y
Exposed door option	Y	Y	Y	
Call relay	Y	Y	Y	Y
External speaker	Y		Y	Y
External microphone			Y	Y
Display			Y	Y
Hands free operation			Y	Y
Keys	3	15	21	21
Speed dial			Y	Y
IP class	65	65	65	22
Mounting style	Wall	Wall	Wall	Flush

7.1.11 Handset mechanical outline



7.1.12 Cross rail drawing



7.2 SC211

The picture below shows the SC211 in a full equipped configuration with headset and loudhailer.

This station is also called the talk-back station.

7.2.1 Description

The station is used on mooring stations, in workshops etc, where it is never necessary to call other than for example Bridge and/or E.C.R.

The station can be called by any telephone on board, and calls shall be answered by pressing either the PTT button of the headset, speaking into the headset microphone, or by pressing one of the 3 keys, speaking into the horn speaker, which is used both as speaker and as the microphone.

Calling from the station is initiated by a press to the PTT button of the headset, or by pressing one of the 3 keys. When making such a call, it is always necessary to press a key while speaking, and release it while listening.

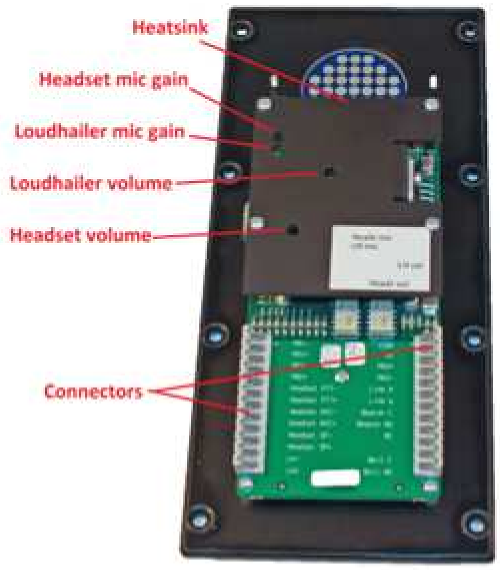
A command call (talk-back) call can be set up from a SC411 or SC421, where the call will immediately open the horn speaker, and the speech direction is entirely controlled by the PTT key of the calling stations. No hands on at the SC211 end. Such a call can be made to a group of up till 10 of the SC211 stations, forming a command group call where all stations are listening when the initiator is speaking.

7.2.2 Specification

The SC211 has the below features:

- 3 call buttons (plus headset PTT)
- Connects loudhailer
- Connects headset (8 ohms 1W)
- Does NOT connect handset
- Receives PA calls
- Receives Alarm calls
- Receives talk-back calls as slave station
- 18-32V DC operation
- Line impedance 600 ohms
- Stand by power 280mW
- Operating power max 15W
- Loudhailer > 8 ohm
- Headset > 16 ohm
- Headset microphone dynamic
- -25 to 70 ºC operation
- Relay for driving horn and rotating flash light (24V DC 1A)
- IP65 enclosure
- EN60945 compliant

7.2.3 Inside



7.2.4 Electrical connections

The station is connected using 2 12 pole screw terminal connectors:

J4:

Pin	Use
1	Optional push button 1
2	Optional push button 1
3	Optional push button 2
4	Optional push button 2
5	Headset PTT button
6	Headset PTT button
7	Headset microphone (dynamic)
8	Headset microphone
9	Headset speaker (>16 ohm)
10	Headset speaker
11	Loudhailer (> 8 ohm)
12	Loudhailer

J5:

Pin	Use
1	24V DC power +++
2	24V DC power ---
3	Optional push button 3
4	Optional push button 3
5	Telephone line
6	Telephone line
7	Relay contact for rotating light N.O.
8	Relay contact for rotating light N.O.
9	PE (protective Earth)
10	
11	Relay contact for horn sounder N.O.
12	Relay contact for horn sounder N.O.

7.2.5 Jumper field

J3 is a jumper field used to select or program some functionality.

J3	Use
1	When set, full duplex operation is selected for the headset
2	When set, the BELL relay will hold during ringing, When not set, the BELL relay will close following the ringing signal
3	

7.2.6 Volume settings

4 volume setting potentiometers are available. They are all accessed via holes in the heat sink, and a small thin screw driver is required. Be careful not to short any circuits on the board when adjusting the volumes.

The adjustments are used as:

VR	Description
1	Loudhailer speaker volume
2	Loudhailer as microphone sensitivity
3	Headset microphone sensitivity
4	Headset ear cup speakers volume

7.2.7 Operating

This paragraph describes how to use the station as seen from the end users view.

7.2.7.1 Making calls

Press one of the buttons of the SC211 or the PTT button of the headset. This will cause the SC211 to call the programmed telephone number (typically the bridge or E.C.R.).
The calling tone will be heard. When the call is answered, the answer will be heard in the loudhailer and headset.
If the call is not answered, press the button again, and the call will be terminated.

7.2.7.2 Receiving calls

When anyone calls the SC211station, the loudhailer will sound the ringing tone, the rotating light beacon relay will be activated, and the bell relay will be activated according to the type of ringing signal received. The ringing tone will be heard in the headset too.
Answering the call is done by pressing one of the buttons of the SC211 while holding the button pressed speaking into the loudhailer, which is used as microphone, or by pressing the PTT button of the headset speaking into the microphone of the headset.

7.2.7.3 Conducting calls

The SC211 will select the microphone active based on the last push button pressed. If any of the 3 buttons on the keyboard are pressed, then the loudhailer will be the microphone to speak into, and when the PTT button of the headset tis pressed, then the headset microphone will be activated.

7.2.7.4 Terminating a call

The SC211 will terminate the call automatically when the far end terminates.

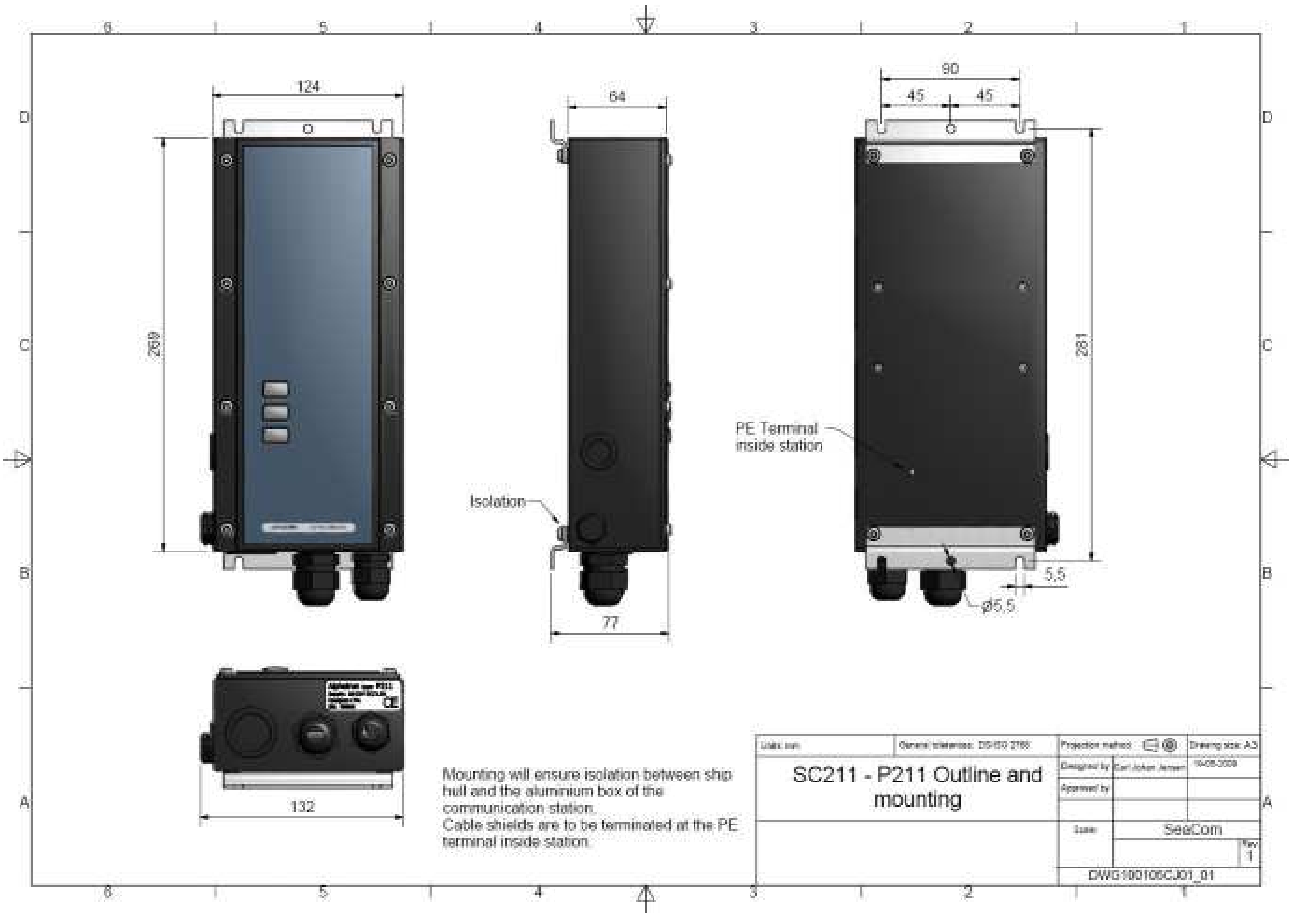
7.2.7.5 Receiving paging calls

Public address calls are received by the SC211. The loudhailer and the headset will be activated. The rotating light beacon relay is activated during the paging call. No user action has to be taken.

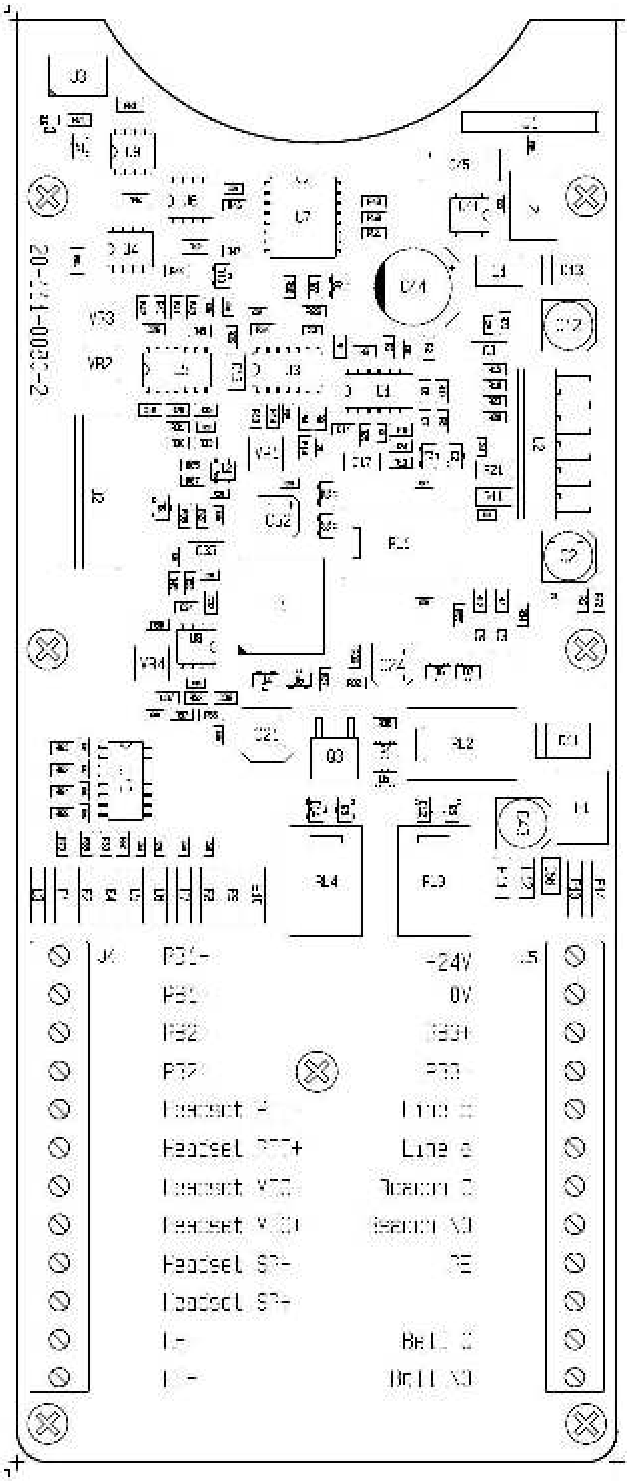
7.2.7.6 Command group (talk-back)

The SC211 is to be used in a command group conference, also called a talk-back call. A command group call typically initiated from the bridge telephone, and up till 10 members can be included in the conference. Initially the speaking part is the bridge telephone, sending an order to one or more SC211 stations. All stations are listening to the current speaker. If any of the participants have something to report back, they will press one of the PTT buttons at their station. This will turn their station into the speaking part of the conference. All other stations will listen to the speaking part. The bridge telephone can regain the right to speak by pressing the PTT of the phone. Refer to system programming chapter for information on how to configure and set up a talk-back call.

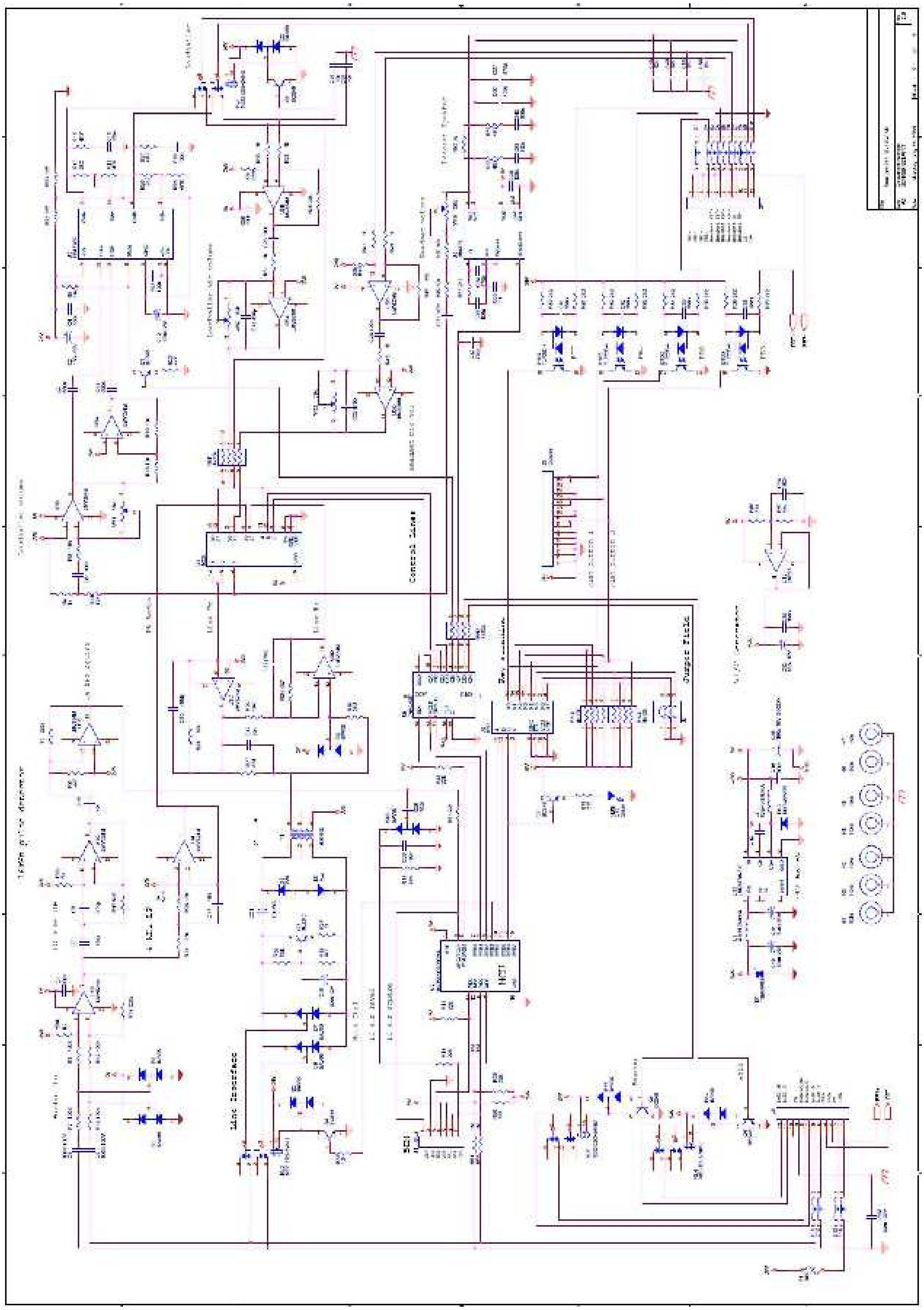
7.2.8 Mechanical outline



7.2.9 PCB layout



7.2.10 Schematic



7.3 SC220

The picture below shows the SC220 water tight, salt mist resistant telephone in a full equipped configuration with the optional headset and optional handset.

This station is also called the industrial telephone station.

7.3.1 Description

The telephone is used, in workshops, engine room emergency generator room etc. where a cost optimal communication solution is needed.

The station is used like a normal telephone, with the additional feature of using a headset where noise conditions requires.

The telephone has a build in call relay, which is activated when the telephone is ringing, and deactivated when the telephone answers a call. The relay has a hold over function with 5 seconds delay.

Although the station with a headset allows for operation in noisy areas, the power to the headset speakers is limited by the available line power. If this is not enough, use the SC421 instead.

The telephone has a very use full function: busy tone disconnect. If the seamen forget to hang on, the build in busy tone detector will do it. This means that communication to critical positions on board is not blocked, just because the seamen forget to han on.

7.3.2 Specification

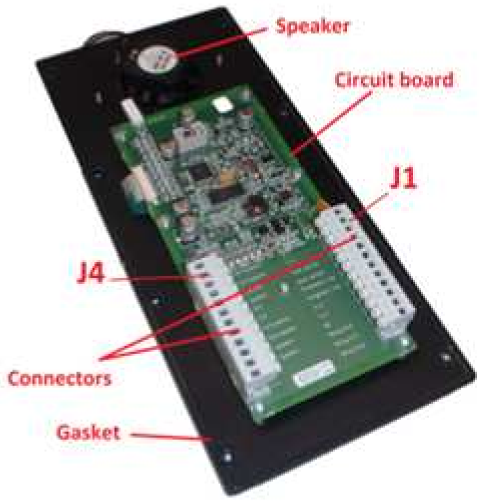
The SC220 has the below features:

- Full numeric keyboard
- Connects handset
- Connects headset (>150 ohm speaker) 10mW
- Headset hook off button
- Full duplex and PTT mode
- Programmable gains and volume
- Powered by telephone line only
- Line voltage 20-50V DC
  ringing 40-90Vrms @20-50Hz
- Line impedance 600ohm
- Busy tone disconnect
- Dynamic microphones 300 ohm
- Headset speakers > 150 ohm
- -25 to 70 ºC operation
- Build in ringer sound 75 dB 1m
- Relay for driving horn and rotating flash light (24V DC 1A)
- IP65 enclosure
- EN60945 compliant

7.3.3 On the front

The front of the SC220 has the below functions:

7.3.4 Inside



7.3.5 Electrical connections

The telephone is connected using 2 12 pole screw terminal connectors:

J4:

Pin	Use
1	Handset microphone
2	Handset microphone
3	Handset speaker
4	Handset speaker
5	Headset PTT button
6	Headset PTT button
7	Headset microphone (dynamic)
8	Headset microphone
9	Headset speaker (>150 ohm)
10	Headset speaker
11	
12	

J1:

Pin	Use
1	
2	
3	Handset hook switch
4	Handset hook switch
5	Telephone line
6	Telephone line
7	
8	
9	PE (protective Earth)
10	Relay contact N.C.
11	Relay contact C.T.
12	Relay contact N.O.

7.3.6 Programming

The telephone has a set of configuration parameters which are set up by the installer by use of the keyboard.

In order for programming, hang on the headset and handset. Press the R button for 2 seconds. When the programming mode is activated, the red indicator starts to flash. Then press the # key together with the parameter number. The green indicator the starts to flash, indicating the readiness for the parameter value. Next enter the parameter value. When accepted, the red indicator starts flashing, indicating that a new parameter can be chosen. The programming mode times out automatically. The parameter 0 must be entered if the factory settings are recalled.

Example: set the gain of the headset microphone to 6:

- Press R for 2 seconds (red flash)
- Press # and 2 (green flash)
- Enter 6 (red flash)
- Wait (no flash)

The below parameters are available:

Number	Use	Default
0	Recall factory, always set to 0	0
1	0 = no headset, 1 = headset is connected	1
2	Gain of headset microphone (0..8)	5
3	Volume of headset speaker (0..8)	5
4	0 = no hold over on relay 1 = 5 seconds hold time on relay	0
5	Gain of handset microphone (0..8)	5
6	Volume of handset speaker (0..8)	5
7	Volume of ringing signal (0..8)	6
8	Conversation time-out 0 = no timeout in force 1..9 = 10..90 minutes	0
9	Timeout on no digits dialed 0 = no timeout in force 1..9 = 10..90 seconds	0
*	Setting time from Busy tone detected to phone hook on 0 = disabled 1-9 = 10-90 seconds	0

7.3.7 Operating

This paragraph describes how to use the SC220 station as seen from the end users view.

7.3.7.1 Handset calls

Lift off handset, await the dial tone, and dial the number to call. When the call has ended, return the handset to its rest position.

A call can be transferred to the headset simply by pressing the headset hook off key.

7.3.7.2 Headset calls

Take on the headset and press the headset hook off key. The dial tone will be heard in the headset. Dial the number to call. When the call has ended, press the headset hook off key again. Be sure that the green call active indicator is extinguished.

7.3.7.3 PTT mode

The PTT mode is used with both headset and handset when the ambient noise is so loud that it can be troublesome to hear the far end. In this case a better signal to noise ratio can be obtained by using PTT mode, because the microphone it not picking up noise while listening.
To enter PTT mode, press the handset PTT button on the keyboard or press the PTT button of the headset in case it is a headset call that is ongoing.
PTT mode is indicated by the green indicator flashing.
When in PTT mode, the PT button, whether it is the microphone button on the keyboard or the headset PTT must be pressed when speaking.

From PTT mode, full duplex can be re entered by a single quick press to the PTT button.

The full duplex mode is indicated by the green indicator steady on.

7.3.7.4 Receiving calls

When anyone calls the SC220 station, speaker will sound the ringing tone, the red indicator will flash and the relay will be activated so a horn will sound or a rotating light beacon will flash.

Answering the call is done by lifting the handset, in which case a handset call is entered, or by pressing the headset hook off key on the keyboard, in which case a headset call is entered.

When a headset is used with 10m cable, it will be possible to answer a call by pressing the PTT button on the headset. Note that this will also activate the PTT mode. If a full duplex call is wanted, the PTT has to be pressed quick to enter full duplex conversation.

Such a call can be terminated by pressing the PTT button 3 times quickly after each other. Note that you have to learn the pace of the 3 push to be made. This allows the user of a long headset cable to hang on without returning to the station itself.

7.3.7.5 Timed termination

The SC220 station has the possibility of making timed call terminations. This is used to make sure that a call is not hanging not terminated because the user has forgotten to hang on after using the headset. Timeouts are programmable using parameter 8 and 9.

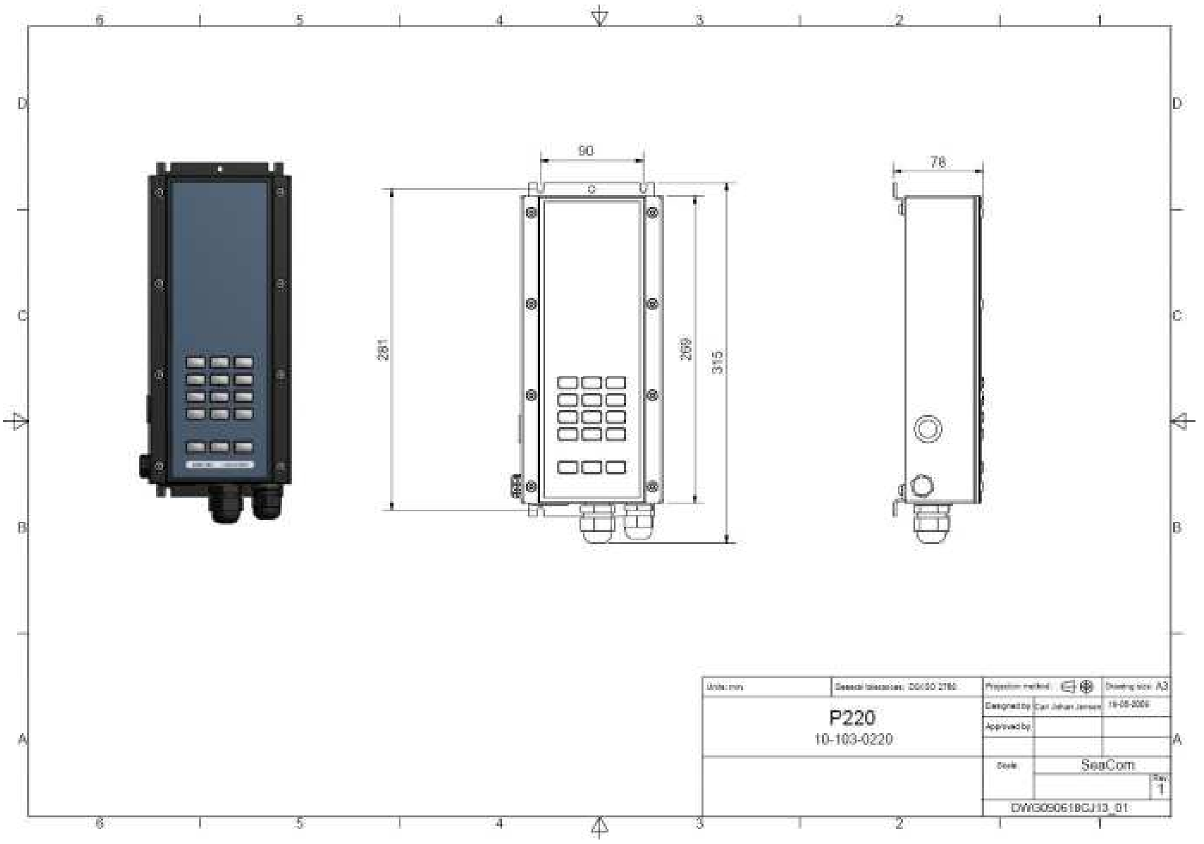
7.3.7.6 Busy tone termination

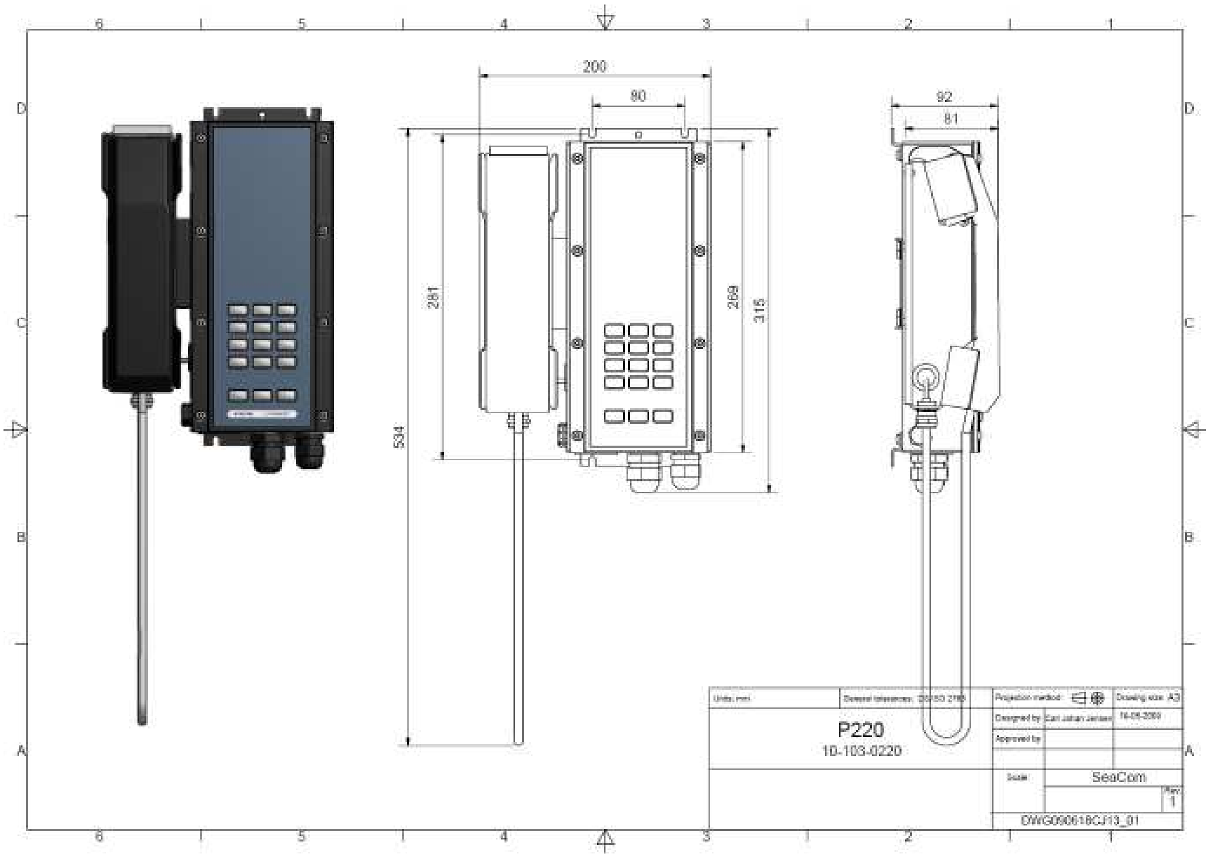
The SC220 station has the possibility of detecting the busy tone from the exchange to which it is connected, and make automatic hangon when a busy tone is sounding. SeaCom style, Alcatel and Ericsson style busy tones can be recognized.

7.3.7.7 Call transfer

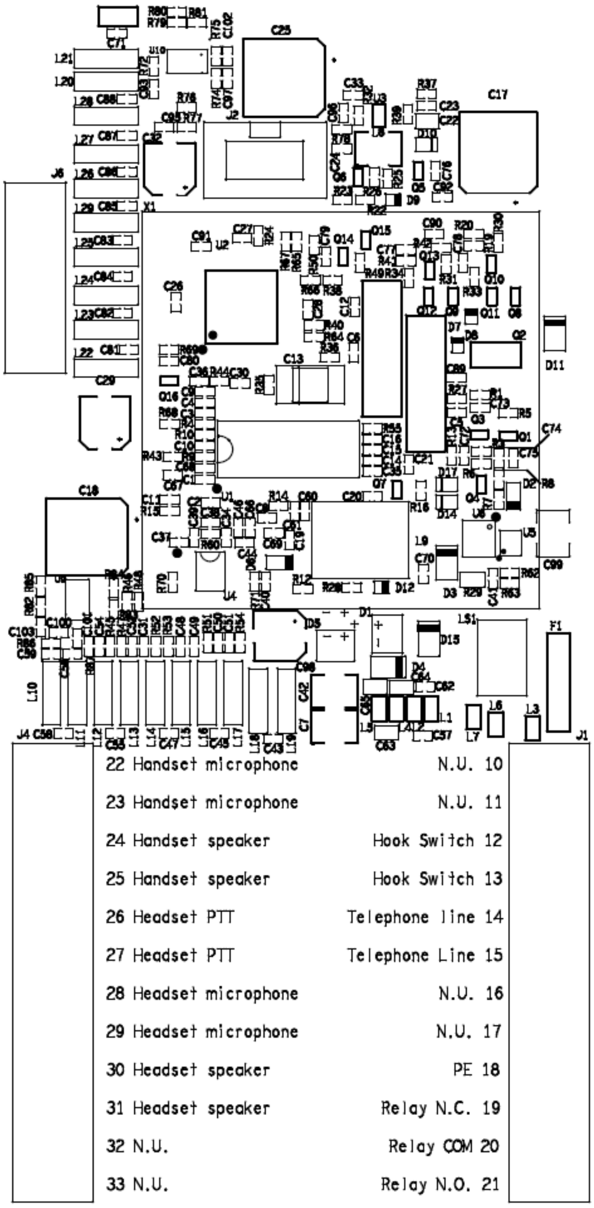
If a call has to be transferred to a third party, then press the R button and wait for the dial tone. Key the third part to call. Hang on the handset or headset, or wait for the third party to answer.

7.3.8 Mechanical outline

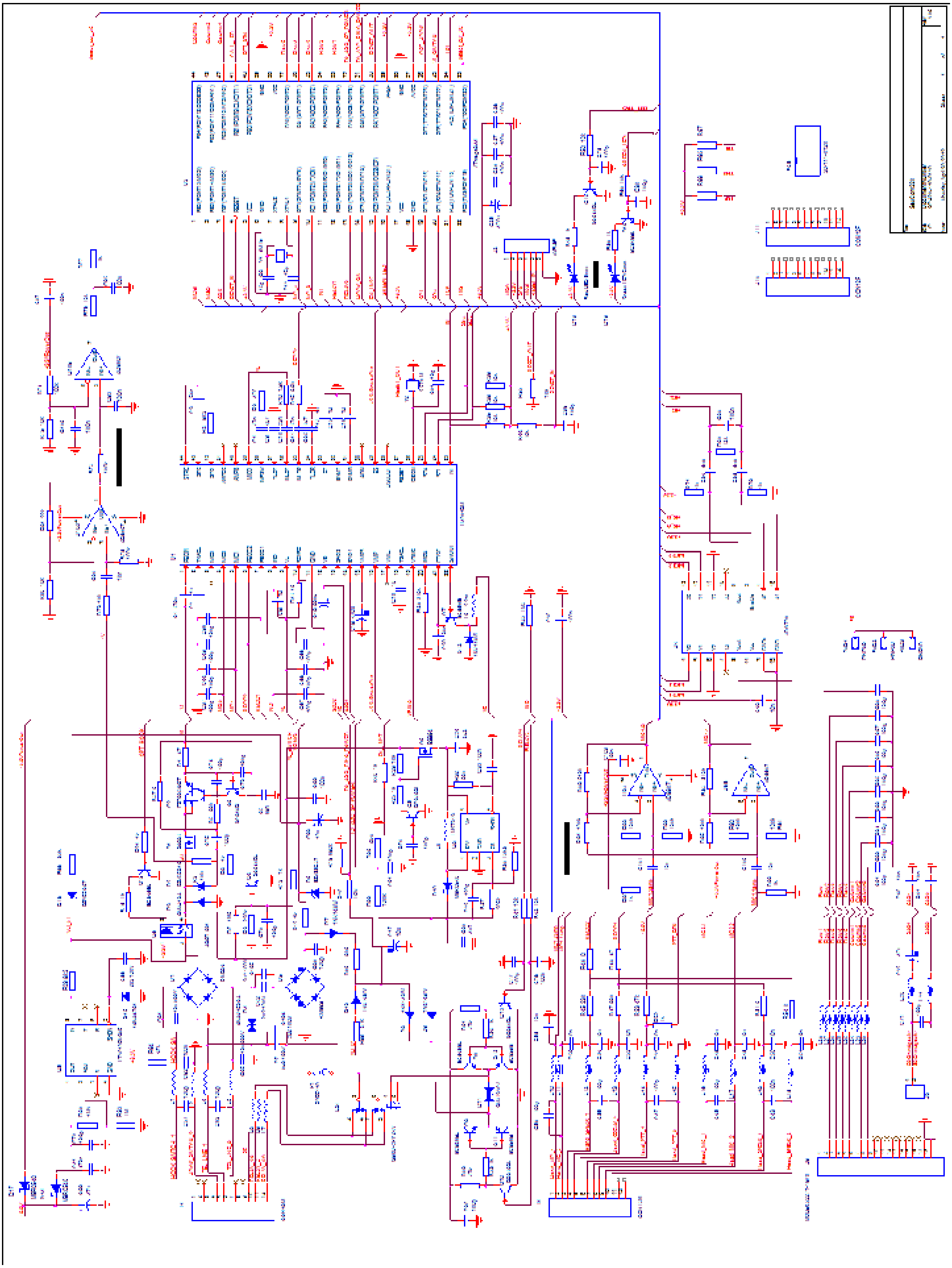




7.3.9 PCB layout



7.3.10 Schematic



7.4 SC411 / SC421

The SC411 and the SC421 are our topmost stations will all features.

The picture below shows the SC411 in a basic version without any accessories.

In this version, the station is used entirely in hands free.

The picture below shows the SC421 station, also in a water tight IP65 enclosure and hands free operated only.

These stations are also called the intercom stations.

The two stations are sharing the same PCB layout and software, but is different in that way that the SC411 is a flush mounted IP22 station with an extra high quality speaker, whereas the SC421 is an IP65 enclosed station looking very much like the stations described in the above chapters.

7.4.1 Description

The SC411 is made for use on the bridge, where a nice screw-less design is required and red backlight with automatic ambient light controlled dimming function is a must. The station is meant to be used in hands free only, but it connects an optional handset which can be used for more private or undisturbed conversations. Although the SC411 has an 8W build in high quality speaker, the hands free operation and communication comfort can be dramatically increased using an external speaker connected to the 10W build in amplifier.

The stations can act as the master for controlling a talk back command group session where one or more SC211 talk back stations are controlled. The speech direction will be controlled by the microphone key of the SC411 or the SC421 station.

Direct-in calls can be made to these stations, where no hands on are required by the party receiving a call. The stations simply opens the conversation directly in hands free. This can be used for example for calling a station at the anchor winch in order to listen to the sound from the winch.

The SC421 is meant for hands free operation on deck or in damp areas. The speaker, the keyboard and the hands free microphone is all designed to withstand moisture and salt mist.

The SC421can be equipped with handset, headset and external speaker or horn, and a call relay is build in for activation horns and rotating beacons in noisy areas.

The substantial number of configuration options available with these stations are accessible through an easy to use menu system.

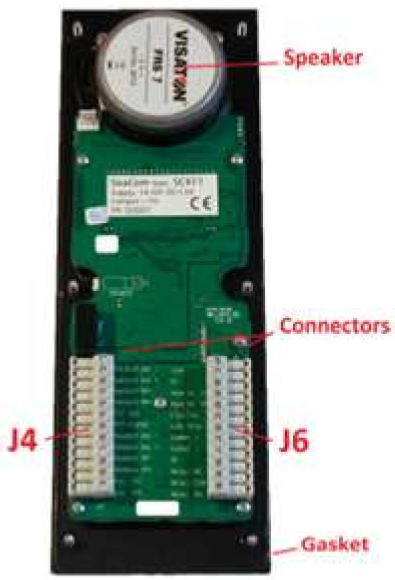
7.4.2 Specifications

- Full duplex, hands-free and PTT
- Connects handset,
- Connects headset (8 ohms 1W)
- Connects 10W external speaker
- Connects external microphone and footswitch
- Voice activated hook off
- Auto dimmed red backlight
- Volume adjustment
- 3 speed dial buttons
- Busy tone disconnect
- 4 ringing sounds
- 18-32V DC power
  250mW standby max. 15W operating
- 25 to 70 °C operation
- IP22 and IP65 enclosure
- Line voltage 20-50V DC
  ringing 40-90Vrms @20-50Hz
- Line impedance 600ohm
- DTMF & LD dialing
- Relay contacts 24V DC 1A.
- Hands free speaker output 85dB 1m
- Hands free microphone max 110dB
- Headset speaker out 1W 8ohm
- External speaker out 12W > 8ohm
- EN60945 compliant

7.4.3 On the front

The front of the SC411 and the SC421 is shown below:

7.4.4 Inside



The picture shows the station without the shielding mounted, so that the PCB can be seen.

7.4.5 Electrical connections

The stations are connected using 2 12 pole screw terminal connectors:

J4:

Pin	Use
22	Handset microphone ---
23	Handset microphone +++
24	Handset speaker
25	Handset speaker
26	Headset PTT button
27	Headset PTT button
28	Headset microphone ---
29	Headset microphone +++
30	Headset speaker (>16 ohm)
30	Headset speaker
32	External speaker 10W
33	External speaker 10W

J6:

Pin	Use
10	24V DC ++++
11	24V DC ---
12	Handset hook switch
13	Handset hook switch
14	Telephone line
15	Telephone line
16	
17	
18	PE (protective Earth)
19	Relay contact N.C.
20	Relay contact C.T.
21	Relay contact N.O.

7.4.6 J2 - RS422 interface

J2 is used for a 4 wire RS232 serial interface. It is used for production testing and for interfacing to the MFC panel. The MFC panel is a computer which through the serial interface can control the SC411 and the SC421 station

J2:

Pin	Use
1	Transmit ++
2	Transmit --
3	Receive ++
4	Receive --

7.4.7 Keys

This paragraph has a short description of the special keys provided.

7.4.7.1 F1 to F3

These keys are used for speed dial, which is a one touch dialing to a pre-programmed telephone number. Refer to the menu system for programming.

7.4.7.2 Up down keys

In idle these keys adjust up and down the backlight. In conversation these keys adjust the volume up and down.

7.4.7.3 M key

This key is used for accessing the dial memories, and for entering the menu system when pressed for an extended period.

After entering the menu system, the M key is used for entering into menus and for accepting entries.

7.4.7.4 Speaker key (hands free)

Hook off into hands free is made by pressing the speaker key, this initiates a call in hands free.

When in conversation, the speaker key can be pressed for an extended period to enable the external speaker.

After making a telephone call, a press to the speaker key will terminate the call acting as a hook on key.

7.4.7.5 PTT key

The PTT key shows a microphone symbol. As the speaker key, it can be used for initiating a call, but in opposition to the speaker key the conversation will be started in PTT (push to talk) mode. In this mode the user must press the PTT key whenever speaking to the called party.

The PTT key cannot be used for terminating.

When the station is a command master (talk back master), the key is used for controlling the speech direction. This condition exist when a semi duplex conference call is made from a SC411 to one or more SC211 stations.

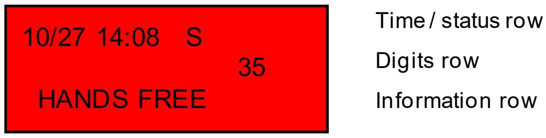
The informative text “COMMAND MASTER” will be shown in the display.

7.4.7.6 R key (transfer)

As for the other stations and telephones this key is used when transferring a call to a third party.

7.4.8 Display

The SC411 and the SC421 have a display which shows incoming calls, shows dialled digits and holds some status and information fields.



The display have 3 rows. The topmost holds the clock and the status field. The mid row holds digits incoming as well as outgoing. The bottom line holds miscellaneous information.

7.4.8.1 Status field

The status field can contain the below characters:

- H In hands free
- M In PTT mode
- S External speaker on
- R,T,I Receive, Transmit or Idle

7.4.9 Call indicator

The call indicator is a red LED showing:

- Quick flash - Ringing
- Steady on - Line is busy
- 1Hz flashing - Line error
- Short flash - Missed calls (5mn timeout)

7.4.10 Backlight control

The stations are constantly monitoring the ambient light level and adjusting the display and keyboard backlight accordingly.

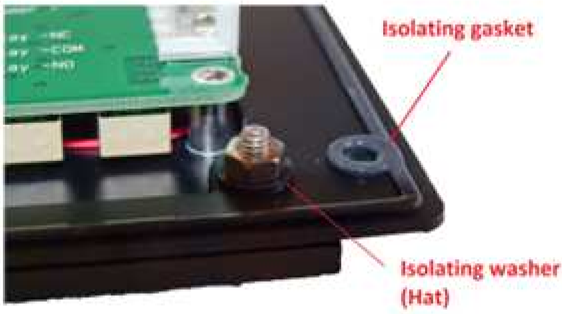
Use the up/down keys to add an offset to the backlight level selected.

7.4.11 Mounting the SC411

The SC411 is differing from the other stations by being a flush mount station. This chapter describes the installation. The SC411 is and is meant to be mounted without any screws visible from the top.
Refer to the console cut out drawings chapter 7.4.14

It is recommended that the station is isolated from the console frame mechanics. This is accomplished as the SC411 has an isolating gasket at the rim of the aluminum frame

And it is supplied with 6 isolation washers (“hats”), which are used when mounting the nuts from behind.

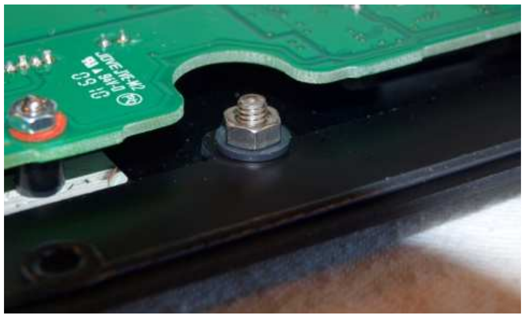


The cable coming from the exchange is a shielded 2x2x0,75 with common shield. The shield must be terminated at the PE terminal of the station.

The station can be mounted with an optional mounting plate, in case it is not possible to access the console from behind. The pictures below shows the mounting plate.

And its mounting

The isolating washer fits into the holes of the mounting plate.



7.4.12 Operating

This paragraph describes how to use the stations.

7.4.12.1 Modes of conversation

The SC411 and the SC421 can be in conversation in the following modes:

- Hands free mode
- PTT mode (push to talk - simplex)
- Handset mode
- Headset mode

A hands free call is initiated by the speaker button.
A PTT call (push to talk) is initiated by the PTT key.
A handset call is started when the handset is lifted.
A headset call is started either when pressing the PTT button of the headset or by pressing the F1 key for 2 seconds. (in which case must not be program-med as speed dial)

7.4.12.2 Dialing

In all modes digits can be dialed either before or after starting the call using the numeric keys.

7.4.12.2.1 Speed dial

For quich and fast dialing, the F1-F3 works as speed dial keys – if programmed as such.

Pressing one of them and the phone will call one of the 3 preprogrammed numbers.

7.4.12.2.2 Memory dial

Press M and then F1-F3 and the display will show name and number of the selected memory location. Press F1-F3 for selecting memory location or use '▼' or '▲' to scroll through M1 to M6 or 'M' to jump from M1-M3 to M4-M6.

When the right number is selected, then lift the handset or press the speaker key to start the call.

The contents of the memories must be programmed using the menu system described chapter 7.4.13.2.15.

7.4.12.2.3 Last number re-dial

To re-dial last number, press 'R' and lift the handset or press speaker.

7.4.12.2.4 Call lists and re-dial

There are two call lists in the memory. One for outgoing calls, one for incoming calls. Each list has 10 entries indexed from 1 to 10 where 1 is the latest. When entering the list, latest call is shown.
Use '▼' or '▲' to scroll through the list. The list will roll from 10 to 1 or from 1 to 10.

Calls marked with '!' in incoming call list, are missed calls.

If the phone is indicating a lost call when idle, '▼' or '▲' works as short cut to incoming call log.

7.4.12.3 How to answer a call

An incoming call is accepted by pressing the speaker button – entering hands free, by pressing the PTT button – entering PTT mode conversation, by lifting the handset – entering a normal handset conversation, or by pressing the F1 or PTT of the headset.

7.4.12.4 Direct in calls

If the station received a direct in call, it automatically turns into hands free and the conversation can beging without any hands on.

7.4.12.5 Voice activated answering

The station do have a build in voice detector. When enabled, the detector tries to detect distinct clapping or shouting between ringing signals. The user can in this way answer incoming calls by voice activation.

7.4.12.6 Terminating a call

All calls are terminated either by hanging up the handset, pressing the speaker button or automatically by the SeaCom exchange system.

7.4.12.7 Switching mode of conversation

During a conversation it is possible to change the mode of conversation. If the call is started in hands free, lifting the handset will turn the conversation into a handset call.

Switching to PTT mode is done by pressing the PTT button, and returning to hands free from PTT mode is done giving the PTT a short push. The display will show the mode active.

7.4.12.8 Switching between PTT and full duplex

When making a hands free call, turning the call into a PTT call (push to talk) can be done by pressing the MIC button. When first pressed, speaking can only be heard by the called party when the MIC button is pressed. Returning to full duplex hands free is done by giving the MIC button a short push. Watch the display to follow the mode You are currently in.

When making a handset call, the MIC button is used in the same way to make the call a PTT call or a full duplex handset call.
The SC411 can be used as talk-back conference master using the handset in PT T mode.

When making a headset call, the PTT button of the headset (or F1 button) is used in the same way. Pushing turns the conversation into a PTT call, and pushing shortly turns the conversation into a full duplex conversation.

7.4.12.9 PA call and PA volume

The stations can receive PA calls. The volume for priority PA is fixed, whereas the volume of non priority PA calls can be adjusted during the PA call or by using the configuration menu.

7.4.13 The menu system

The SC411 and the SC421 has a comprehensive menu system which is described in this chapter.

7.4.13.1 Navigating in the menu

To enter the menu system, press and keep 'M' pressed for approx. 2 sec.

Use key up and down for navigating, and select a menu using the M key.

To exit the menu system, wait 10 sec, or pres the speaker key.

7.4.13.2 The menu – overview

Below an overview of the menus available:

ENTRY

- Speaker volume
- Backlight
- Ringer volume
- P.A. volume
- Bell signal
- Calls out
- Calls in
- SET_UP_MENU
  - Auto answer
  - Auto busy
  - Direct in
  - Hands free
  - External speaker
  - Date & Time
  - Speed dial
  - Memory 1-6
  - SPECIAL_SETUP
    - Dial signal
    - Loop dial type
    - Remote PTT
    - Remote Hook switch
    - Lcl Ext Mic
    - Headset microphone type
    - Headset VOX.
    - Local microphone gain.
    - Handset microphone gain.
    - Headset microphone gain.
    - External microphone gain.
    - Relay mode.
    - Relay hold off.
    - Contrast (display)
    - F1 Mode
    - Voice hook off

The following paragraphs describes the use of each of the set up items.

7.4.13.2.1 Speaker volume

Used to preset the speaker volume.

7.4.13.2.2 Backlight

Adjust level of backlight.

7.4.13.2.3 Ringer volume

Set the level of the ringing sound.

7.4.13.2.4 P.A. volume

Set the volume of a non priority PA call.

Note that the volume of a priority PA call is fixed and cannot be changed.

7.4.13.2.5 Bell signal

Choose the type of ringing signal sounding when and incoming call is active.

A sample is played when the type is altered.

7.4.13.2.6 Calls out

This is the outgoing call list. If list entry is empty, time will show '- - -'

Picking from the list for making a call can be made by selecting an index and initiate a call.

7.4.13.2.7 Call in

Same as 'Calls out' but shows received call.
Set up menu

7.4.13.2.8 Auto answer

Set the number of ringing signals received before the station automatically hooks off and enters hands free conversation.

Use the auto busy function to make it detect busy tone and hang on again.

This feature is used to simulate direct-in when the stations are used with a non SeaCom exchange, which is not capable of communicating the direct-in to the stations.

7.4.13.2.9 Auto busy

When this feature is enabled, the station will try to detect a busy tone on the line. If a busy tone is detected, then an automatic hook on will be performed.

This feature is to be used with the auto answer when making a simulation of direct-in on a non SeaCom exchange, which is not capable of communicating the direct-in to the stations.

7.4.13.2.10 Direct in

This set up parameter can be used to enable and disable the direct-in calls to a station.

7.4.13.2.11 Hands free

Set the stability margin used when the station is in hands free.

The range is 10dB to 40dB.

Use this setting to fine tune an installation for maximum hands free performance.

With a station only using build in hands free microphone and build in hands free speaker, this parameter seldom need adjustments. But if an external speaker is installed and used in hands free, this parameter must always be tuned.

A low value gives the maximum hands free comfort, but also the highest risk of howling and feed back. Setting a high value reduces the risk of feed back, but also reduces the comfort of using hands free, as the switching between speech directions is heavier and more accentuated.

The right setting is the lowest value giving stability and no feed back under all circumstances.

Using an external speaker located well in distance of the hands free microphone gives the optimal change of being able to use a low stability margin, and thereby the highest quality of hands free conversations.

7.4.13.2.12 External speaker

Enable the external speaker to be on by default. When starting conversations and when ringing.

7.4.13.2.13 Date & time

Adjust the date and time. To be used if the exchange to which the station is connected is not capable of sending dsta and time using FSK.

Data format MUST be text string like : YY/MM/DD HH:MM.

NOTE: Settings will be overwritten by any FSK data issued from the telephone exchange and is not preserved when power is off.

7.4.13.2.14 Speed dial

This menu is used to program the 3 F1,F2,F3 speed dial buttons.

7.4.13.2.15 Memory dial

The menu to use when programming the 6 memory dial locations.

Use the ‘F1’-‘F3’ to select M1-M3 / M4-M6 or use ’▲’ or ’▼’ to change to M4-M6 / M1-M3.

7.4.13.2.16 Dial signal

Choose loop disconnect dial (pulse dial) or DTMF signaling for dialing numbers to the line.

7.4.13.2.17 Loop dial type

Choose international or Swedish coding of digits for loop dialing numbers to the line.

7.4.13.2.18 Remote PTT

When this function is enabled, the headset PTT input can be used as a PTT button for using the external speaker as microphone.
Just like for the SC211 station.

7.4.13.2.19 Remote Hook Switch

When this function is enabled, the handset hook switch input can be used as a PTT button for using the external speaker as microphone.
Just like for the SC211 station.

7.4.13.2.20 Local Extern Microphone

Enabling this parameter will change the function of the headset PTT and the headset microphone inputs.

The function is meant to be used on a bridge where the SC411 station cannot be located at the position wherefrom communication is needed. On that location a goose neck microphone can be installed – connected to the headset microphone input - and a push button or foot switch installed -connected to the headset PTT input.

The push button or foot switch is used for accepting incoming calls, just as if pressing the PTT button on the SC411 itself.
If hands free communication is wanted, a short push will turn the SC411 into hands free.

Obviously calls cannot be made from this position, as this requires dialing at least some digits.

7.4.13.2.21 Headset microphone type

Select if the headset microphone is an electrets or dynamic type.

7.4.13.2.22 Headset VOX

This parameter, when set to enable, activates the headset VOX function. This is a voice controlled PTT function on the headset.

Using the PTT function reduces noise from the station, and reduces the noise heard in own headset ear cups, as the microphone of the headset is first turned on when speech is detected.

7.4.13.2.23 Local microphone gain

Enter this menu in order to adjust the gain of the hands free microphone.

This is used for adapting to different ambient noise levels, where the gain can be reduced for noisy environments, and the amplification can be increased where silence is present.

Be aware that increasing the gain automatically adds to the necessary hands free stability margin, so choosing a high gain adds to the feeling of the hands free speech direction switching, whereas reducing gain increases the hands free comfort.

Adjusting the gain is assisted by a level meter displayed when the menu is chosen.

A good practice is to speak to the microphone, and increase the gain until the P (peaking) is seen, after which the gain is reduced two steps.

7.4.13.2.24 Handset microphone gain

A menu used for adjusting the gain of the handset microphone.

Hint: Lift the handset and wait 20 seconds for timeout, then enter the menu while having the handset in Your hand.

7.4.13.2.25 Headset microphone gain

A menu used for adjusting the gain of the headset microphone.

Note: Headsets are used in noisy areas, and when noise reaches 120 dB, this parameter need to be adjusted to its minimum in order to keep the input stages operating in their linear range.

7.4.13.2.26 External microphone gain

This menu is used for adjusting the gain of the input amplifier when the local speaker is used as microphone.

7.4.13.2.27 Relay mode

This menu item is controlling the behavior of the relay:

Name	Relay activated on
Ring	Ringing
PA	PA call received
Ring + PA	Ringing or PA call received
Invert	Active always
Inv. Ring	Not activated when ringing
Inv. PA	Not activated when PA call is received
Inv Ring + PA	Not activated when ringing or PA call is received
Call	Activated when the telephone is active in a call (Used for PA speaker mute)

7.4.13.2.28 Relay hold off

Enable this function if the relay shall have a hang over time keeping it activated between ringing signals.

7.4.13.2.29 Contrast

Adjusting the contrast of the display.

7.4.13.2.30 F1 mode

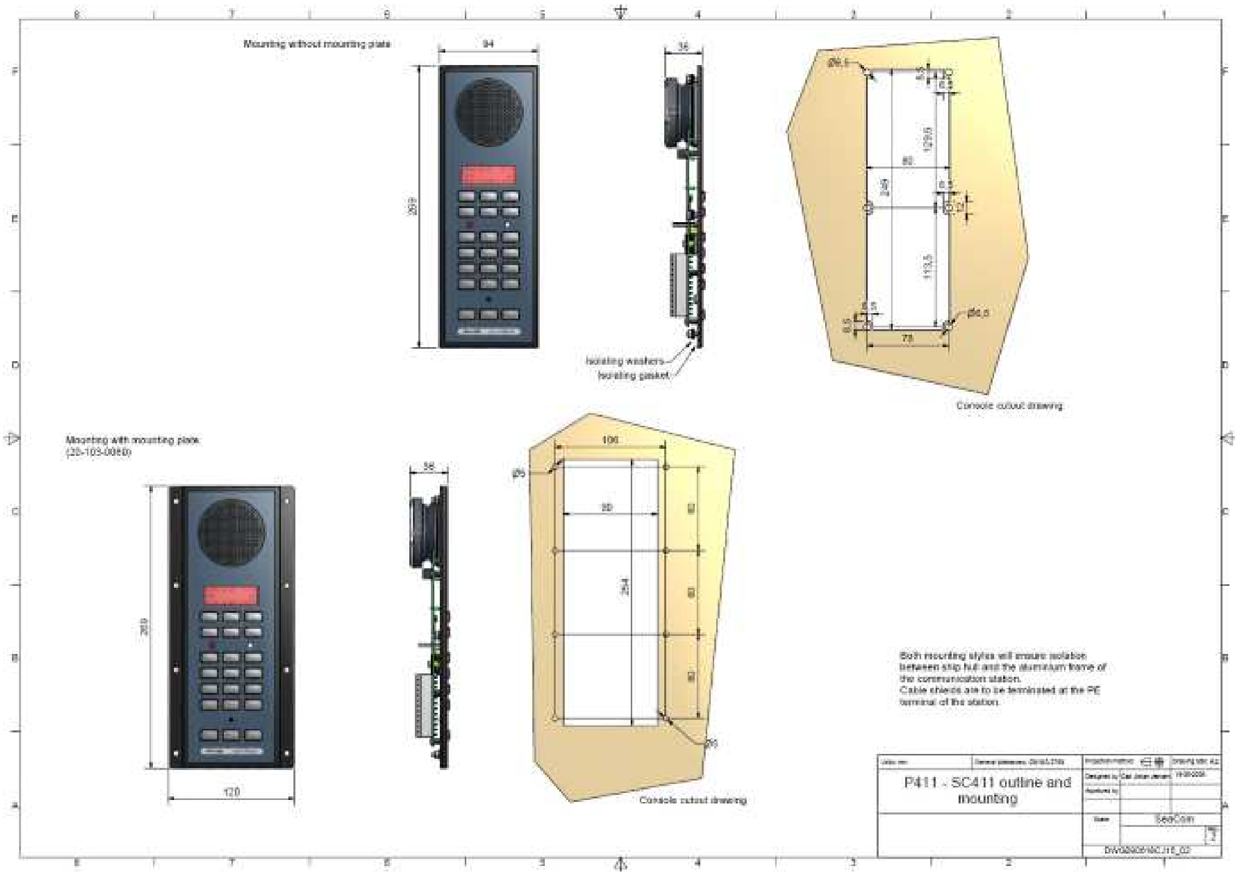
This parameter has two settings: headset and hands-free.
When set to headset, pressing F1 for more than 2 seconds starts a headset call, whereas the speaker key starts hands-free call.
When set to handsfree, pressing F1 for more than 2 seconds starts a hands-free call whereas pressing the speaker key starts a headset call.

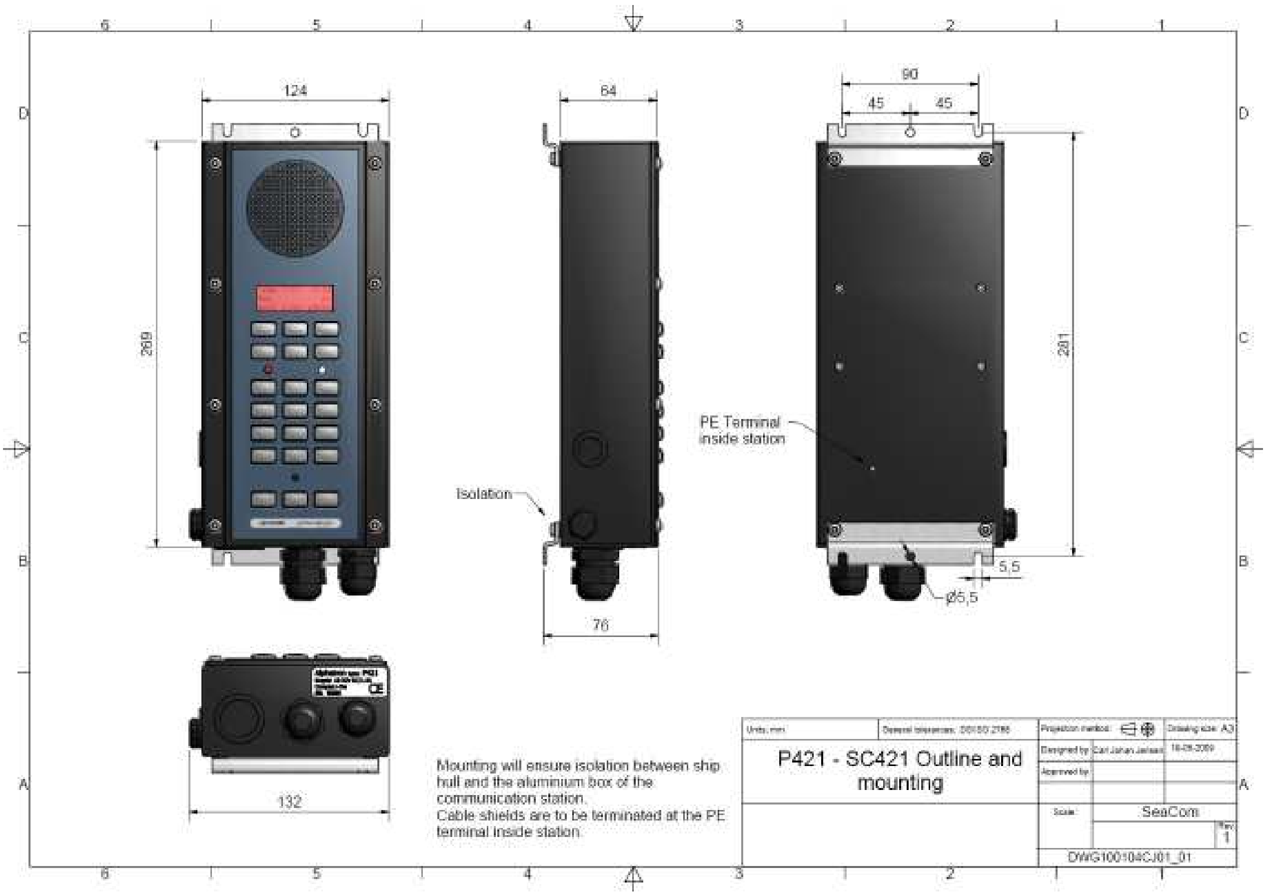
This is used for stations in always noisy areas, where only headset calls have a meaning, and so will it be possible to use headset and start and stop conversations using the speaker key.

7.4.13.2.31 Voice hook off

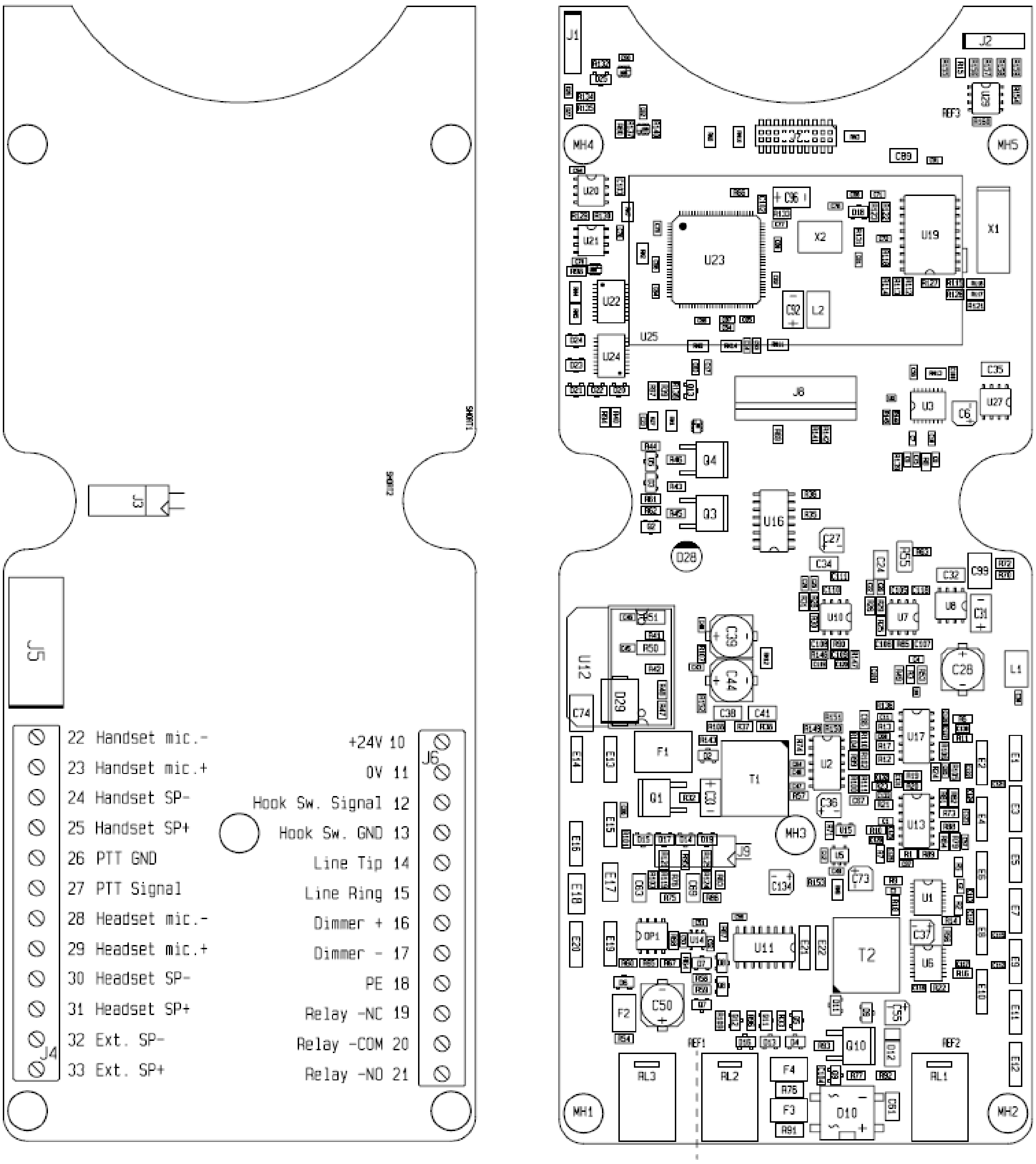
This parameter enables the build in voice detector. When enabled, this detector is active between ringing signals. If a distinct clapping or shouting is detected, the station will hook off into hands free conversation mode. Conversation must be terminated by the calling party.

7.4.14 Mechanical outline

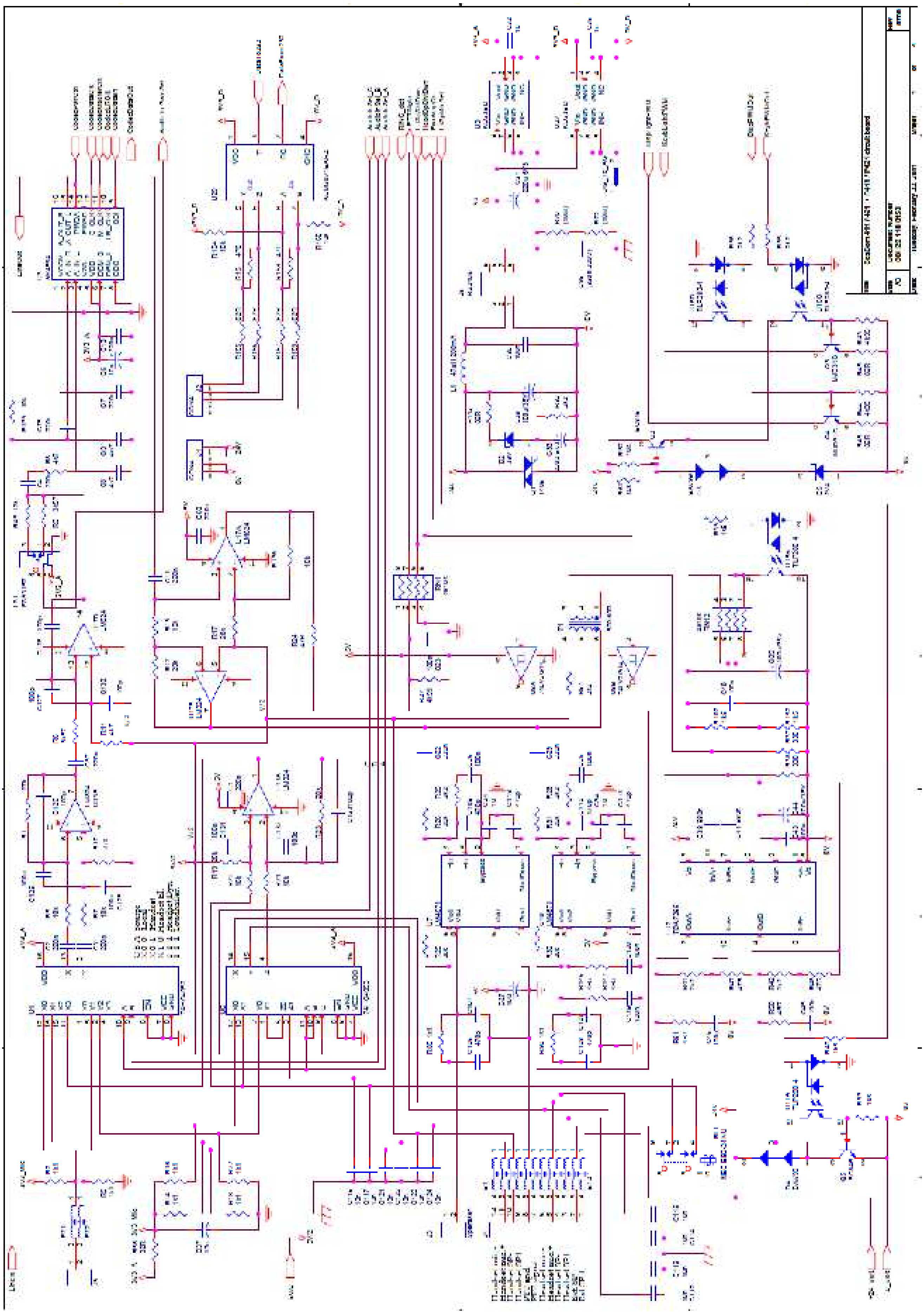


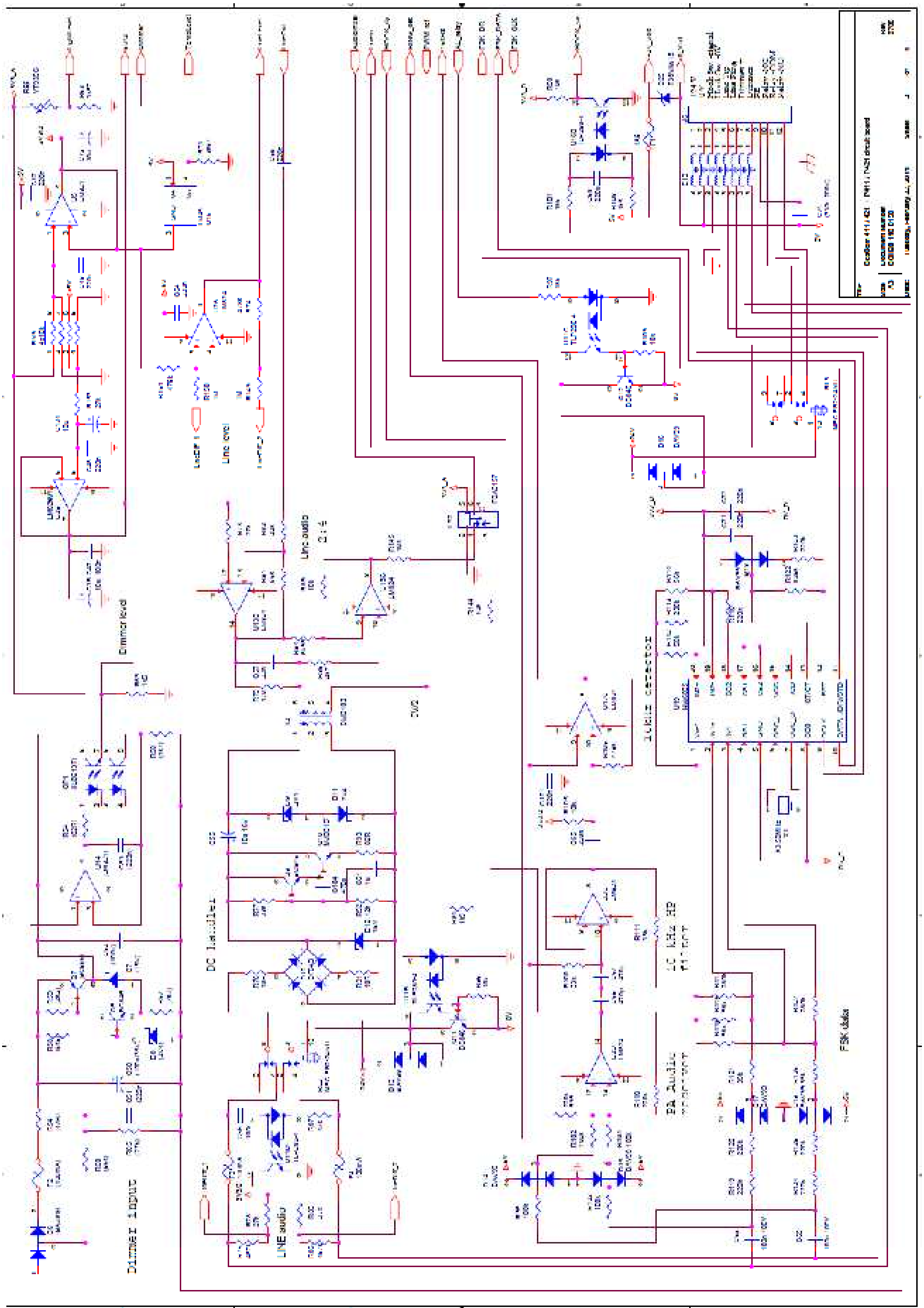


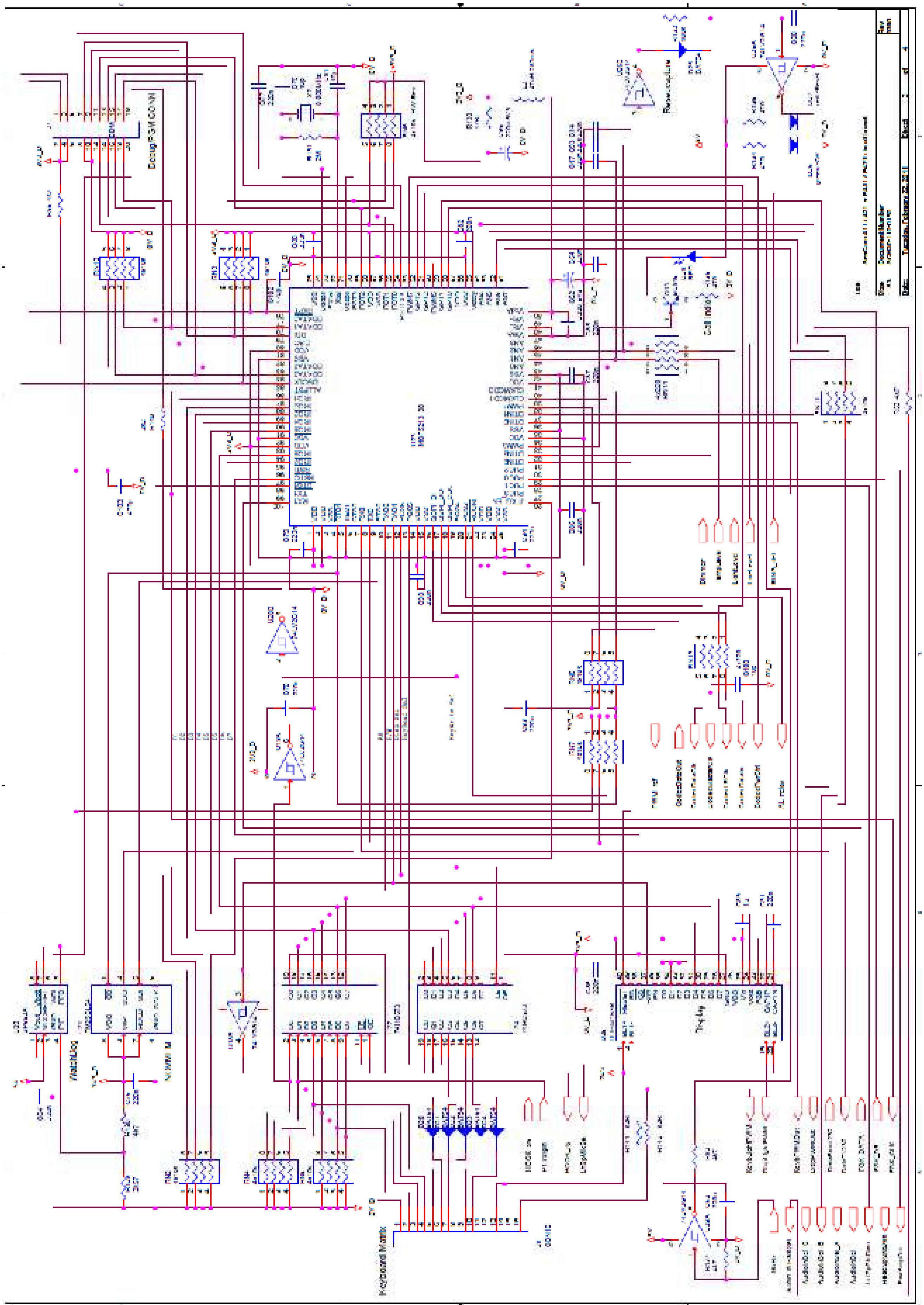
7.4.15 PCB layout



7.4.16 Schematic







8. System programming

The SeaCom must be programmed during commissioning, in order to determine the call numbers of telephones, in order to grant access rights to satellite communication e.t.c. Programming is done using a Windows based configuration tool.

This chapter describes how to get access to the configuration file and how to use the configuration software.

8.1 Getting access to the configuration

There are 3 ways to get access to the configuration of the system:

- Using the shared folder via Ethernet
- Using USB stick
- Using remote desktop via Ethernet

8.1.1 Shared folder access

The SeaCom 3000 provides a shared folder, which hold the configuration tool and file.
To access this You will need a Windows computer and a network cable.
Open the network port in the bottom of the SeaCom 3000, and connect the Ethernet cable from there and directly to Your Windows computer.

The SeaCom 3000 has a DHCP server, which means that You normally don't have to do any configuration of You network port.

The system has the following access data:

Computer name / IP:

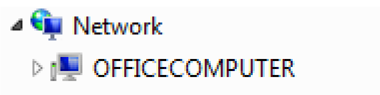
phonesystem / 192.168.0.53

user name and password:

admin 1017

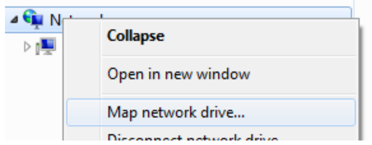
On the Windows computer, open a file Explorer (NOT the internet explorer).

Find the display of network places:

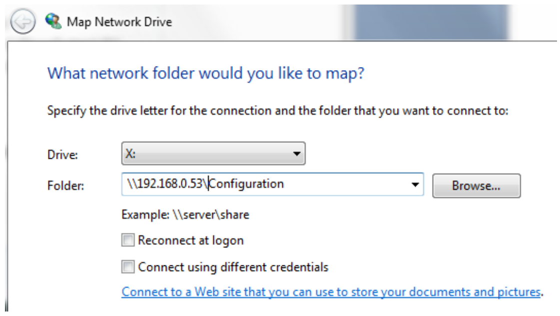


If the **phonesystem** is shown, just continue to browse to the folder **Configuration**.

If the **phonesystem** is not shown, You can connect the shared folder by using the "map network drive":



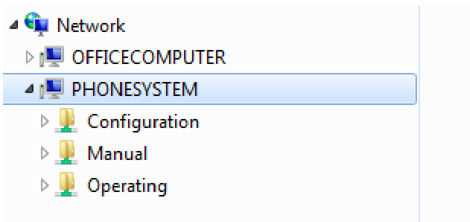
and enter *\\192.168.0.53\Configuration* or *\\phonesystem\Configuration*



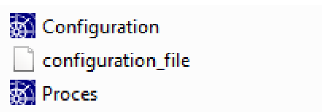
You will be faced with the user name and password dialog

And here You enter *admin* and *1017*

After that the **phonesystem** might be shown in the file browser:



Double click the **Configuration** folder, and You will find these files:

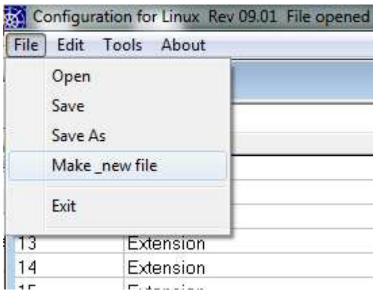


The **Configuration** is the tool software to be used to edit the **configuration_file.**
Proces is not used here, but this is the CP software matching the two other files.

To start the configuration, simply double click the **Configuration**. You will NOT be asked for any file to open, as the file opened file is always the one being in the current folder.

Editing the file and using the configuration software is described in details later in this chapter.

When closing the configuration tool, the file is saved onto the disk of the SeaCom 3000 system. But it is NOT set into operation until the next reboot of the system.
If You like to force a reboot of the system, then You can use the menu *Make_new_file:*



This will make the system store the edited file and ask the CP software to set it into operation immediately. The process will take approximately one minute.

Details about how the file flow is can be found in chapter 9.4.

8.1.2 USB access

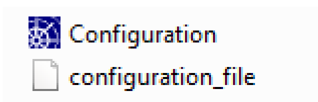
Another way of getting access to the configuration tool and file is via USB stick.

You will need an USB disk that the system can recognize. It recognizes by far the most, but to be sure, use the SeaCom approved USB sticks.

What You do, is to insert the USB disk into the USB port of the SeaCom 3000 system, and the wait for the disk activity indicator to stop flashing. This takes less than 10 seconds.

Then remove the USB stick, and insert it into a Windows computer.

Use the file browser to find the USB disk, and You will see the below content:

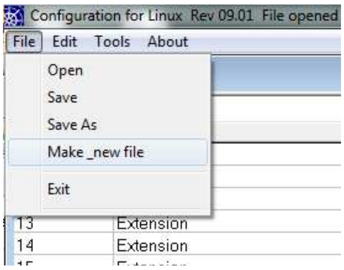


This is the Configuration tool and a copy of the running **configuration_file**

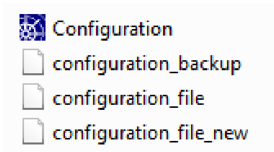
Now double click **Configuration** file, and the tool will open the **configuration_file** found on the USB stick.

Editing the file and using the configuration software is described in details later in this chapter.

When editing is finished, You MUST use the menu *Make_new_file:*



This will make the system store the edited file together with a copy named **configuration_file_new** on the USB disk.:



Eject the USB disk from the Windows system properly. This is important to avoid corrupting the files.

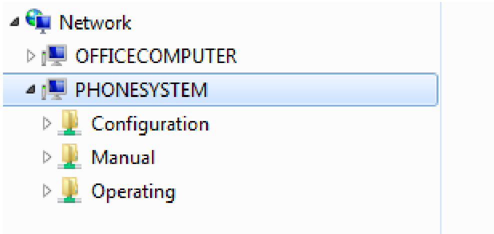
Re-insert the USB disk into the USB port of the SeaCom 3000 system, and wait to the disk activity indicator stops flashing.

The system will now take the new updated **configuration_file_new** and bring it into operation- The process takes about one minute.

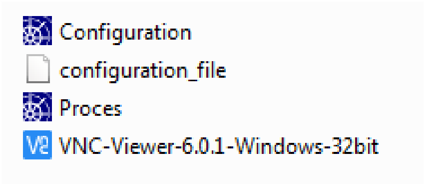
Details about how the file flow is can be found in chapter 9.4.

8.1.3 Remote desktop access

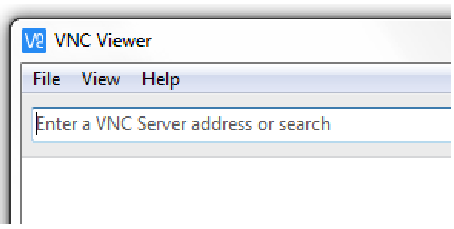
The SeaCom 3000 system gives the possibility of connecting via Ethernet using remote desktop. The VNC_viewer is used.
The VNC_viewer is a piece of software running on a Windows computer which is connected to the SeaCom 3000 via a networtk cable.
To get a copy of the VNC-viewer You can visit the *www.realvnc.com* or You can get it from SeaCom 3000 system disk where it is store id the folder **Manual**.
Use the instruction of paragraph 8.1.1, but instead of mapping or locating the **Configuration** folder, then map or locate the **Manual** folder.



In this folder You will find:

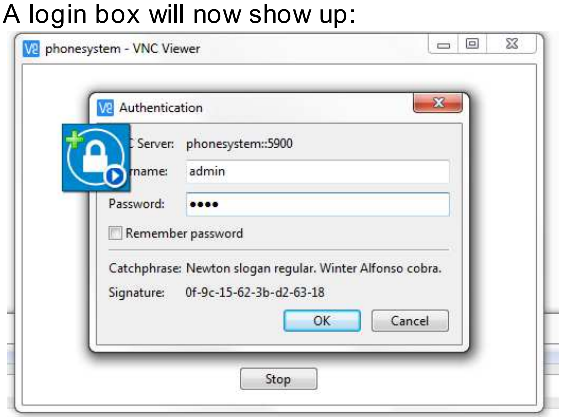


Copy the VNC-Viewer-6.0.1-Windows to the desktop of Your Windows computer, and start the software by double clicking the icon.
Then You will be faced with this window:



Enter either **192.186.0.53** or **phonesystem** in the field.

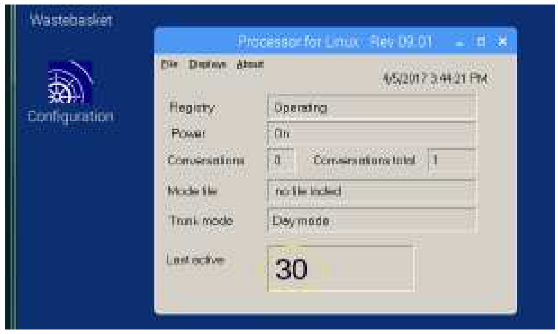
A login box will now show up:



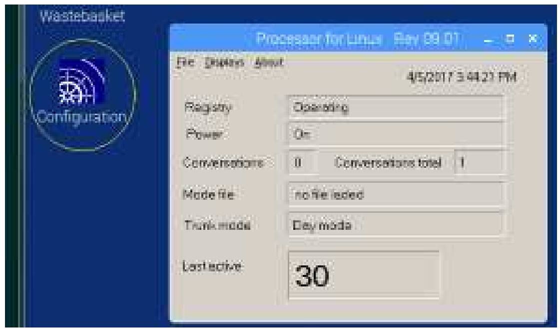
Enter user name **admin** and password **1017**

After than the remote desktop will be show:

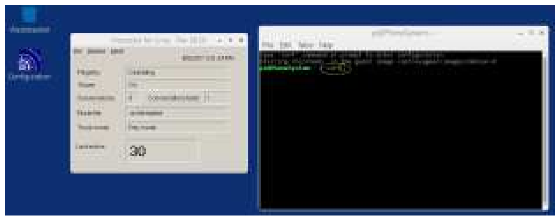
Remote desktop is a very power full tool, and it is really handy when installing a system, as You can see the last active extension.



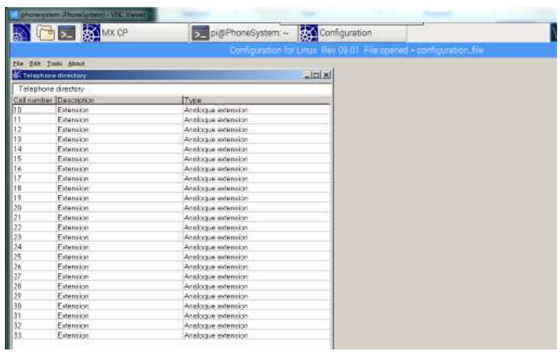
But You can also start a configuration session by double clicking the icon.



This will open a terminal window:

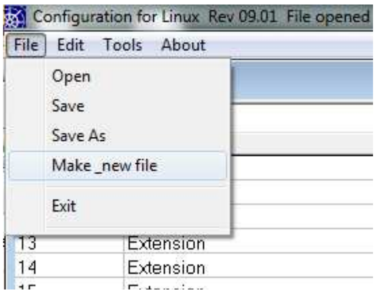


Now at the command line enter the command *conf.*
This will start the **Configuration.exe** file in a new window in the remote desktop session:



The file being now edited is located in the **Configuration** folder. This means that on this step we are doing exactly the same thing as when using the shared folder access method, but the software is running on the SeaCom 3000 system itself.

When editing is finished, the edited file is found in the **Configuration** folder.
But it is NOT set into operation until the next reboot of the system.
If You like to force a reboot of the system, then You can use the menu *Make_new_file:*



This will make the system store the edited file and ask the CP software to set it into operation immediately. The process will take approximately one minute.

Details about how the file flow is can be found in chapter 9.4.

8.2 General concepts

The following many pages will describe the use of the configuration tool in all details.
W start by som basic set-up concepts.

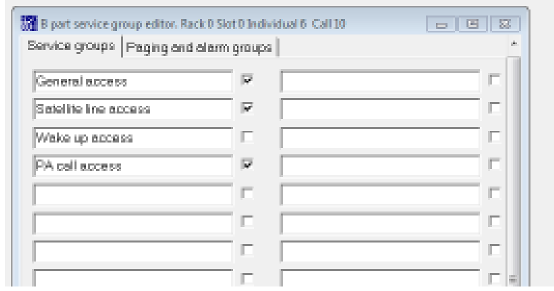
8.2.1 Call numbers / descriptions

All call numbers - extensions, trunk lines as well as system call numbers, will hold a call number of maximum 10 digits. The call numbers can be combined from any number of ASCII characters. But for a number to be "dial-able", it must only consist of the characters 0..9.

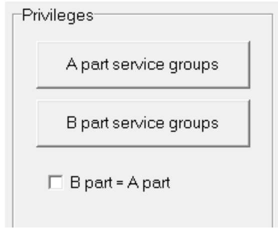
And all call numbers have a description attached, which is displayed in display telephones, and which can be used when printing telephone directories.

8.2.2 Service groups

All call numbers will be linked to a list of 32 service groups in which the call number is entitled to participate. This system is used to restrict access to for example satellite services, wake–up calls, right to set system time etc.



To open the editor used to include a call number to a service group, the below buttons are available for all call numbers:



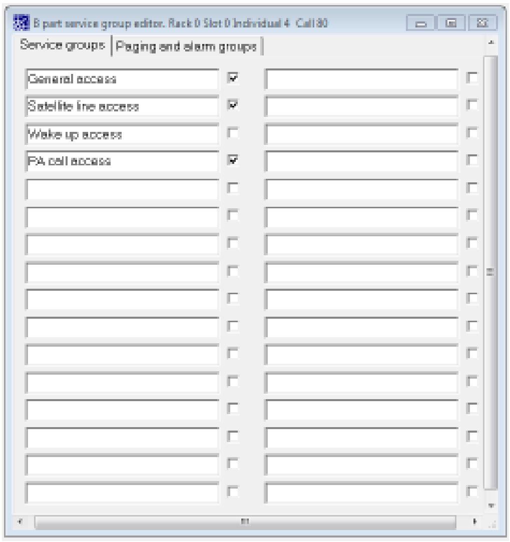
Assigning a service group to a call number can either be made for both directions – when the A=B is clicked, or a division between outgoing and incoming rights can be implemented when the A=B is unchecked.

A telephone is the A part when it is taking initiative to a call, and will be the be part when a call is received.

The use of this can be for allowing all telephone to receive call coming in from a satellite service, and restricting some telephones from making outgoing satellite calls.

8.2.3 The service group editor

The service group editor for all call number looks like:



32 groups are available. The names of the service groups are to be determined by the installer, and can be edited globally from any open service group editor.

The check marks however only relates to the call number for which the editor is open, and by checking / un-checking, the call number is included or excluded from participating in a service group.

The service group editor has two pages, of which the second is used for including telephones into paging and alarm calls groups. As for the service groups, the paging group names are free to choose and globally edited from any paging group editor, whereas the check marks belongs to the single individual of which the service group editor was opened.

8.2.4 System call numbers

System call numbers are call numbers representing a dialed functions which does not have any hardware related to it.
A good example of a system call number is the priority dialing. The user dials *, which is the default call number of this type system call. When the processor of SeaCom encounters that the dialed number is a priority call, then it presents the caller of a new dial tone, and makes a priority call to the dialed number.

The following system call numbers are available:

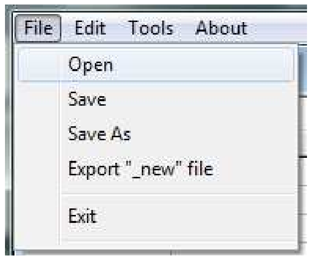
- Priority dial
- Set date and time
- Wake up ordering
- Ringing group
- Short number dialer
- Number alias
- Standard dialer
- User Account and PIN code checker
- Paging call
- Call pickup
- Semi-duplex conference group
- Alarm distribution call
- Mode select
- Do not disturb
- Direct in request
- Day mode / Night mode

8.3 The Configuration application

This chapter describes the use of the **Configuration.exe** application. All screen shots are made with the expert access level in force, so the full menu is seen.

8.3.1 The Files menu

The file menu is used to open, save and to make saved copies of configuration files.



8.3.1.1 Open

Use this menu when another configuration file is to be opened. The standard File Open dialog box will be used.

8.3.1.2 Save

Use this menu when the current file is to be saved. This will overwrite the current file with new data. The system always saves the file when the **Configuration.exe** application is closed.

8.3.1.3 Save As

Use this menu to save the current file under a new name. This can be used for making backup copies of the file or mode files.

8.3.1.4 Export "_new" file

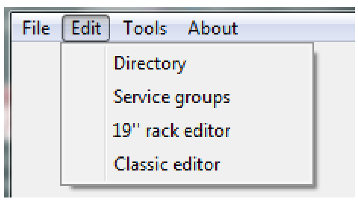
Used to save the current file and export it to the file: **configuration_file_new**
This is used as a signaling file to the system requesting immediate use of a new file and rebooting.

8.3.1.5 Exit

Used to close the application. This will automatically save the file.

8.3.2 The Edit menu

The edit menu is used to choose the way of accessing the configuration.

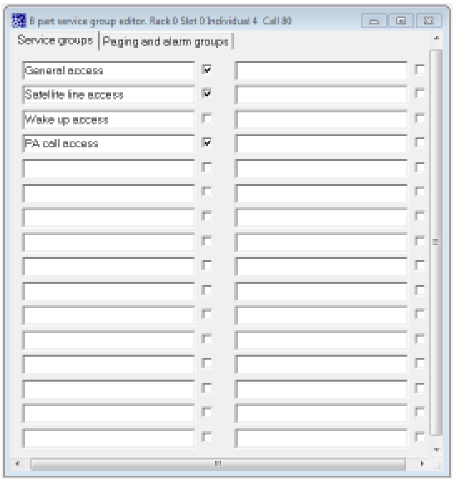


8.3.2.1 The Directory menu

This menu is used to open the telephone directory. The directory window is used for most of the system maintenance, and is a tool of such importance that is covered by its own chapter.

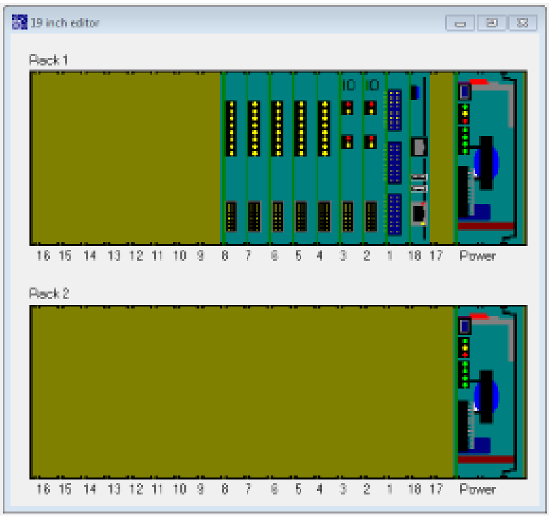
8.3.2.2 Service groups menu

This menu will open the service group name editor.



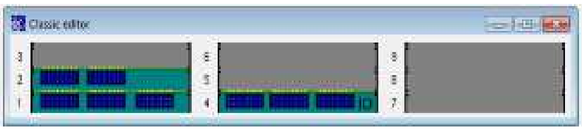
8.3.2.3 19'' rack editor

This menu should NOT be used with the SeaCom 3000.
It is used when the CP software operates in a 19'' environment.



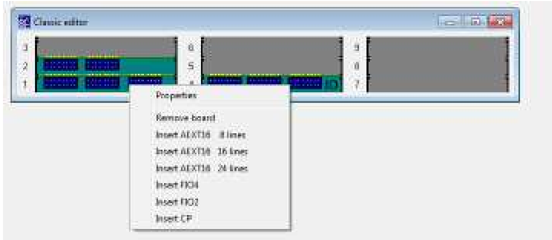
8.3.2.4 Classic editor

This is the physical editor to be used with the SeaCom 3000 system.



8.3.2.5 Selecting a new board type

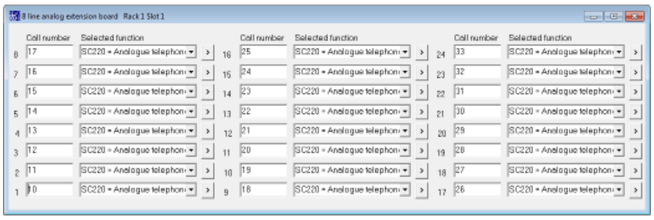
Each board position can be configured with one of the existing board types. There is a right mouse button menu used to insert a new boards.
Let the mouse point to the board position where to insert a new board, right click and chose the board type from the list.



When a board is inserted into the system, its propertied can be edited by double clicking on the board in the Classic editor. The properties windows of boards will be shown below..

8.3.2.5.1 AEXT16 board

The AEXT16 analogue telephone line editor looks like:



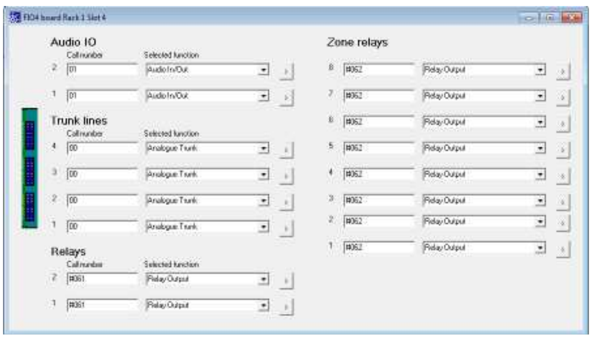
This board contains 24 individuals, which can be selected to operate either unused or as a basic analogue extension or as one of our communications station families.

Choosing the equipment type connected to an individual of the extension board is the first and most basic programming to be made when configuring a system.

The properties of each individual type is discussed in chapter 0

8.3.2.5.2 FIO4 board

The FIO4 trunk board editor looks like:

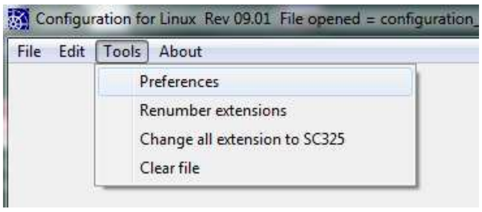


There is one individual for each port of the FIO4.

The properties of each individual type is discussed in chapter 0

8.3.3 The tools menu

This menu is used for miscellaneous purposes:



8.3.3.1 Preferences

This menu opens the preferences editor form.

8.3.3.1.1 Installation identifier

This is a text field to be used by the engineer installing the system to describe the system The text field will by default be found in the header of telephone directory printouts.

8.3.3.1.2 Serial number

A number identifying the system. Free to be used or not as an identifier of the delivery.

8.3.3.1.3 Last extension number

This number shows the last extension number used by the auto number assignment system. Can be altered in order to make the auto number system start from a selected extension number.

8.3.3.1.4 COM Port

Serial port used by the **Proces.exe**

8.3.3.1.5 Error log level

Selecting the error log level. Normally *operating*, but some difficult problem is to be traced, the level can be changed to *debug*.

8.3.3.1.6 CP watchdog enable

The PSU holds an overall system watchdog, which can be enabled by clicking this check box.

8.3.3.1.7 Use PCCPV2

Default PCCPV2, but if the Proces.exe should be used with old FIO2 boards, the PCCPV1 can be choosen.

8.3.3.1.8 Alarm when extension error

Click this field if the alarm relay has to be activated and a log generated whenever an extension line is hanging off hook for more than 30 minutes.
Clicking this field will enable the extension supervisory function.
The alarm relay will return to no alarm state when the off hook condition is fixed.

8.3.3.1.9 Trunk day mode / night mode

By clicking here You can choose the day mode or night mode on configuration time.

8.3.3.1.10 This is a mode file

Click this field if the file is a mode file. This will open some extra settings and menus used with mode files.

8.3.3.2 Renumber extensions

You can renumber all extensions of a full exchange. Numbering will be done starting with a number specified by the user, and the order will be the physical order in which the extensions are found.

8.3.3.3 Change to SC325

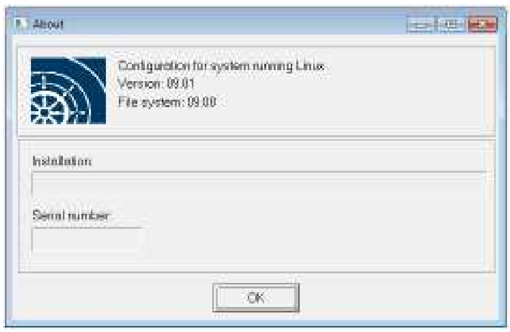
This is used if one like to change all telephone types to SC325.

8.3.3.4 Clear file

Make a brand new file. This will delete all.

8.3.4 The About menu

The About menu will open the about box showing the installation identifier and serial number plus information on the revisions of code and file system



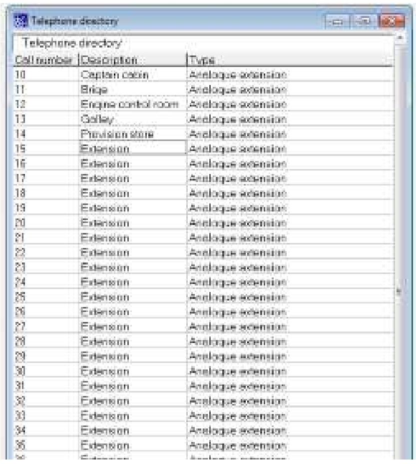
This form gives a number of system information's.

Installation: An identifier entered by the engineer installing the system

Serial number: A number entered by the engineer installing the system

8.4 The Telephone directory

By far the most installation programming work can be undertaken from the telephone directory window. This window is shown, when starting the **Configuration.exe** tool.



8.4.1 Header field

The text bar just below the drag bar is used for entering the text to be used as the header of printouts.

8.4.2 The directory grid

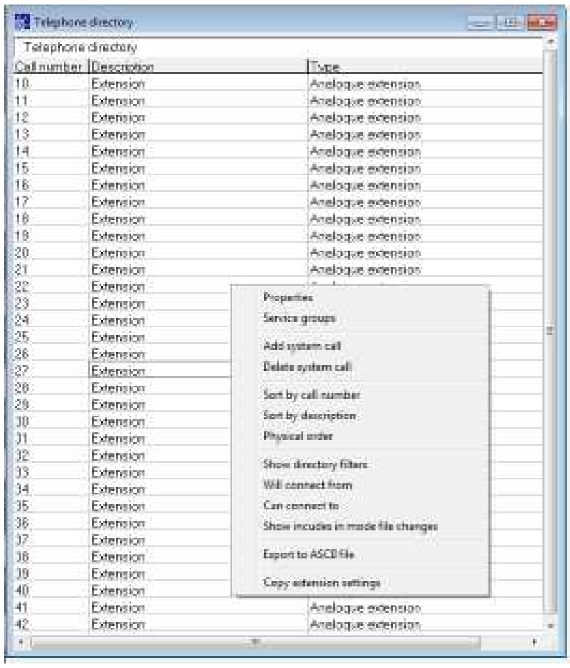
The directory grid is used to display 3 columns: The call number column, the description column and the type column.

The first 2 columns can be edited directly in the string grid, whereas the type column just displays information about what kind of individual is covered by the call number and description.

Double clicking an item in the string grid opens the properties editor of the individual.

8.4.3 Pop up menu

Working with the telephone directory is primarily done through the use of the right mouse button pop up menu.



The items of this menu will be described in the following text. The daily user will have access only to a subset of the menu.

8.4.3.1 Properties menu

Using this menu opens the properties editor of the individual currently selected in the directory grid. This is equal to double clicking the individual in the directory grid.

8.4.3.2 Service group menu

This menu opens the service group editor of the individual selected in the directory grid. This is a short cut for opening the properties of the individual, and then opening the service group editor.

8.4.3.3 Add system call number

When using this menu, a small selection box appears. This box is shown:

Click the button of the system call type to add. This will ad one system call number to the bottom of the telephone directory.

To edit the properties of a system call number, then double click the call number in the directory grid.

8.4.3.4 Delete system call

This menu will delete the system call number currently selected in the directory grid.

8.4.3.5 Sorting the telephone directory

A small number of sorting mechanisms can be activated by the use of the 3 sorting menus. When first started, the telephone directory shows the call number is their physical order. This order is determined by the board positions in the board magazine. Systems are shown in the order they appears in the configuration file.

8.4.3.6 Sort by call number

This menu will sort the telephone directory by call number

8.4.3.7 Sort by description

This menu will sort the telephone directory alphabetically by the descriptions.

8.4.3.8 Physical order

This menu will restore the order to the physically order as when the telephone directory was first opened.

8.4.3.9 Show includes in mode file changes

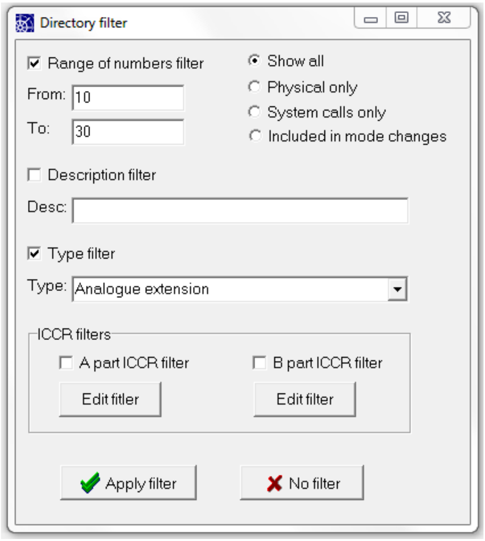
This menu will sort out the call numbers that will be included in mode file copying.

8.4.3.10 Filtering the telephone directory

A versatile tool for filtering the telephone directory is available.

8.4.3.11 Show directory filter

This menu opens the telephone directory filter form.



Filling items of the form will cause the telephone directory to filter it content according to the rules entered. There is a logical AND between the items of the form.

8.4.3.12 Will connect from

This menu is used to get a quick view of individuals that can connect to the currently selected individual in the telephone directory.

8.4.3.13 Can connect to

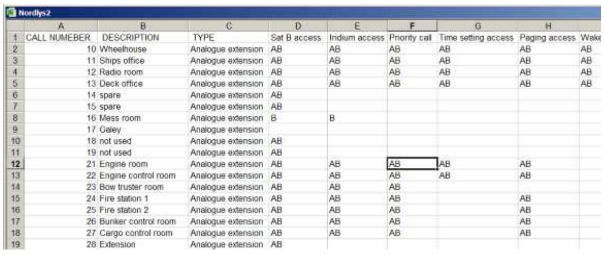
As the above menu, this menu is used to make the telephone directory display the individuals that can be connected from the currently selected individual.

8.4.3.14 Export to ASCII file

The directory displayed in the directory grid can be exported to a comma separated ASCII file.

```
CALL NUMEBER, DESCRIPTION, TYPE,Sat B access,Iridium access,Priority call,Time setting access,Paging access,Wake up access
10,Wheelhouse,Analogue extension,AB,AB,AB,AB,AB,AB,
11,Ships office,Analogue extension,AB,AB,AB,AB,AB,AB,
12,Radio room,Analogue extension,B,B,B,B,B,B,
13,Deck office,Analogue extension,AB,AB,AB,AB,AB,AB,
14,spare,Analogue extension,AB,,,,,,
15,spare,Analogue extension,AB,,,,,,
16,Mess room,Analogue extension,,,,,,,
17,Galey,Analogue extension,,,,,,,
18,not used,Analogue extension,AB,,,,,,
19,not used,Analogue extension,AB,,,,,,
21,Engine room,Analogue extension,AB,AB,AB,AB,AB,,
22,Engine control room,Analogue extension,AB,AB,AB,AB,AB,,
23,Bow truster room,Analogue extension,AB,AB,AB,,,,
24,Fire station 1,Analogue extension,AB,AB,AB,,AB,,
25,Fire station 2,Analogue extension,AB,AB,AB,,AB,,
26,Bunker control room,Analogue extension,AB,AB,AB,,AB,,
27,Cargo control room,Analogue extension,AB,AB,AB,,AB,,
```

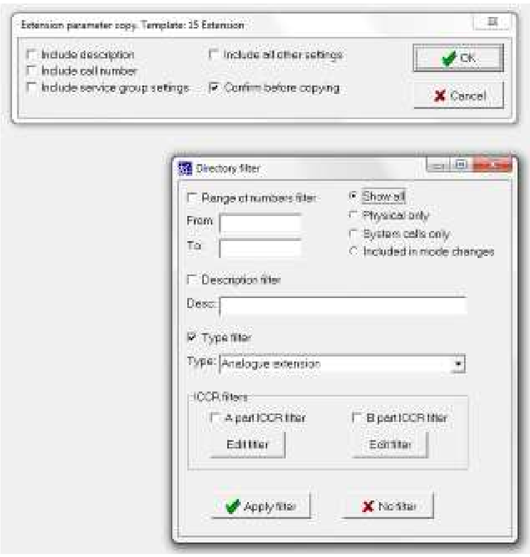
And when imported into an excel sheet it can look like:



8.4.3.15 Copy extension settings

When many extensions are to share the same settings, an effective way to set the parameters for these extensions is to configure one extension, and then copy the settings to all the other extension going to have same settings.

To copy the settings of an extension, select it on the number table grid. Then use the right mouse button to select the “Copy extension settings” menu. Selecting the menu will open two forms: the copy extensions parameter form and the general directory filter form. The first form is used to select which parameters are to be copied. Most often copying the call number has no great meaning, but copying the description such as “passenger cabin” can be convenient. 4 levels are available:



Check the “include all other settings” to copy all data that are not description, call number or service group settings.

The directory filter form are opened to give the possibility of choosing which extensions are receiving the settings. Use the filter box before pressing OK in the copy extensions parameter box. The extensions shown in the directory are the extensions receiving data.

Create a test file in order to familiarize with the copying function before using it on a real data file.

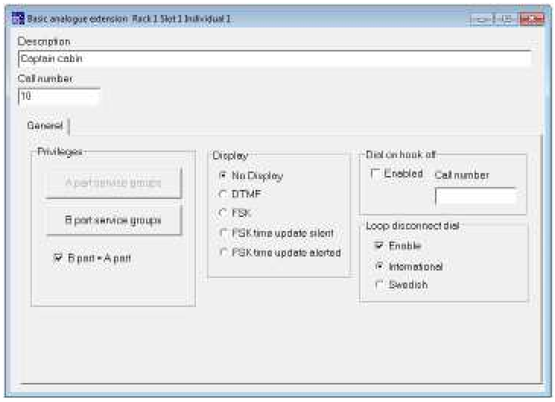
8.5 Properties of individuals

This chapter holds descriptions of the configuration of individuals of the SeaCom system.
The chapter includes both physically individuals such as extensions and trunk lines and system call individuals such as short number system and priority calls.

The properties editors of individuals can be opened either from the physical editor form, typically done when making the first time system programming, and from the telephone directory form, typically done when maintaining the system.

8.5.1 Basic analogue extension

This editor shows like:



8.5.1.1 Call number / Description

These items are standard individual items found on all individuals

The call number field contains the call number to be used by the individual. Max 10 digits or characters can be entered.

The privileges window contains 2 buttons for opening the A part and the B part service groups. If the B part = A part check button is checked, only the A part editor can be opened, and the contents of the B part will be a copy of the A part.

8.5.1.2 Display

This box selects how the display is handled.

No display	- for telephones with no display
DTMF	- for telephone using DTMF protocol
FSK	- Telephone will receive caller ID using the FSK protocol

FSK silent - FSK caller ID display plus time updating with no alert ringing

FSK alerted - FSK caller ID display plus time update with alert ringing

8.5.1.3 Dial on hook off

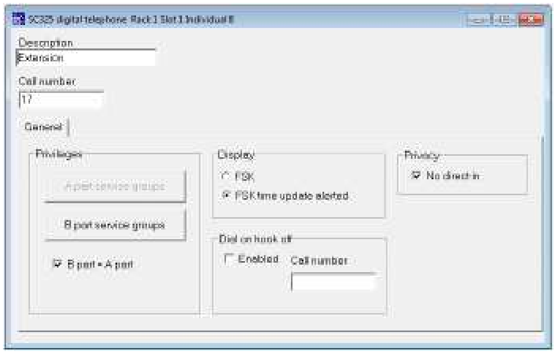
The window Dial on off hook is used when you want the extension to connect directly to a call number whenever it is hooked off.

8.5.1.4 Loop disconnect dial

Enables the loop disconnect = pulse dial method used wit old telephone sets.
2 types of decoding are possible, Swedish or International. The difference is the count of disconnects that signals the zero digit.

8.5.2 SeaCom 325

Our digital telephone does have its own individual type:



8.5.2.1 Call number / Description

These items are standard individual items found on all individuals

The call number field contains the call number to be used by the individual. Max 10 digits or characters can be entered.

The privileges window contains 2 buttons for opening the A part and the B part service groups. If the B part = A part check button is checked, only the A part editor can be opened, and the contents of the B part will be a copy of the A part.

8.5.2.2 Display

This box selects if the telephone shall receive time updates.

FSK - Telephone will receive caller ID using the FSK protocol but NOT timeupdates

FSK alerted - FSK caller ID display AND time update with alert ringing

8.5.2.3 Dial on hook off

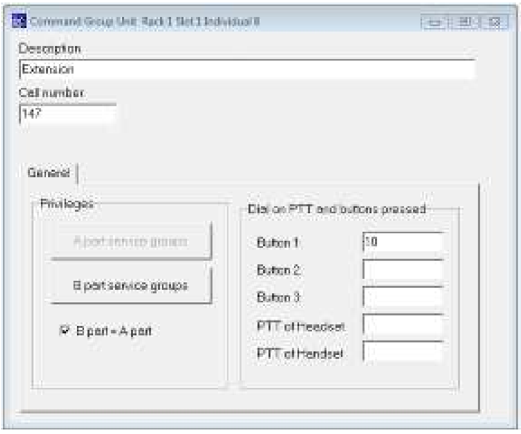
The window Dial on off hook is used when you want the extension to connect directly to a call number whenever it is hooked off.

8.5.2.4 No direct-in

Check this to prevent the phone answering direct-in calls. This could be used for maximum privacy.

8.5.3 SC211 Talk-back station

This individual type is used for the SC211 station.



8.5.3.1 Call number / Description

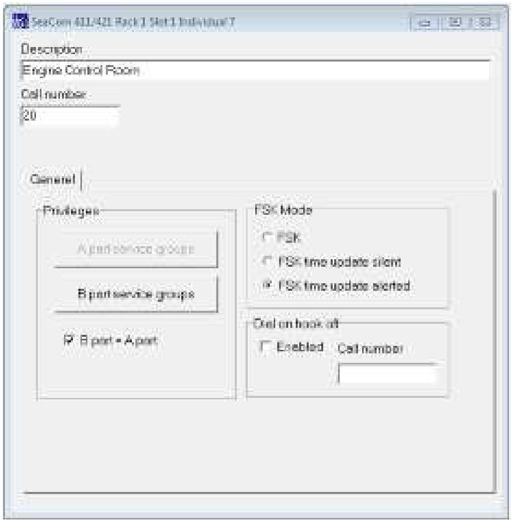
These items are standard individual items found on all individuals

8.5.3.2 Dial on PTT and buttons pressed

The SC211 startion has 3 push buttons. When the unit is in idle, pressing one of these buttons starts a connection sequence calling the numbers put into the boxes.

8.5.4 SC411 and SC421 stations

These individual types covers the stations SC411 and SC421



8.5.4.1 Call number / Description

These items are standard individual items found on all individuals

8.5.4.2 FSK Mode

FSK - Telephone will receive caller ID using the FSK protocol but NOT time updates

FSK time update silent - FSK caller ID display AND time update with no alert ringing

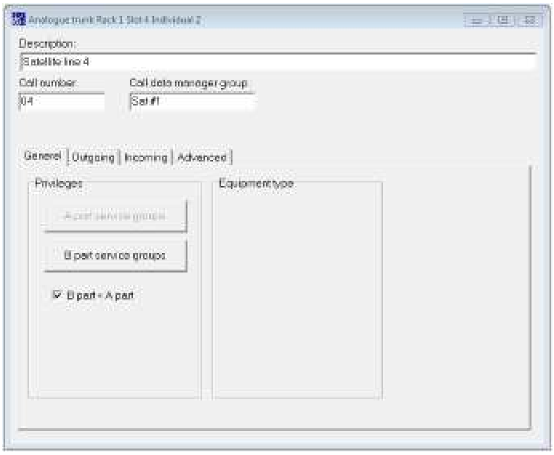
FSK alerted - FSK caller ID display AND time update with alert ringing

8.5.4.3 Dial on hook off

The window Dial on off hook is used when you want the extension to connect directly to a call number whenever it is hooked off.

8.5.5 Analogue trunk

This is an individual of the FIO2 board.



8.5.5.1 Call number / Description

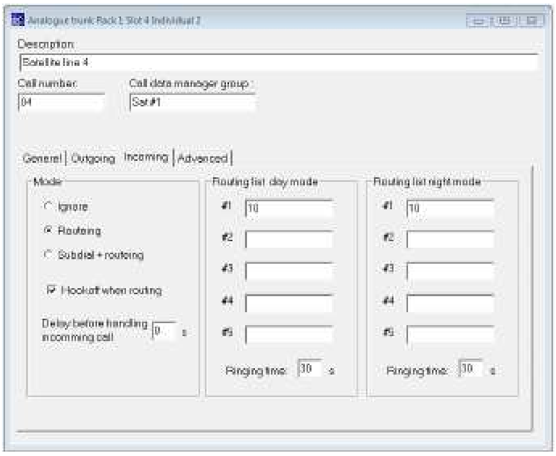
These items are standard individual items found on all individuals

8.5.5.2 Trunk group

This item is used for stating the type of the trunk. This is used when calculating the price of a call via the trunk. All trunks sharing the same group name will share the same charge calculation values. Not used by SeaCom 3000 systems.

8.5.5.3 Incoming calls handling

How the trunk line handles incoming calls can be determined by altering the set-up items of the Incoming tab page.



8.5.5.3.1 Incoming mode

The incoming mode window gives 3 possibilities:

- Incoming calls are ignored
- Incoming calls are routed using the call numbers shown in the Routing list window
- Incoming calls will be faced with a new dial tone for a period equal to the general dial time out, during the period of which the caller can make sub-dialing to a local number. If the caller does not make any sub-dialing, the Routing list will be used.

8.5.5.3.2 Hook off when routing

When this box is not clicked, the hook relay of the trunk line will remain open as long as the routing is going on. When anyone answers the routing (B answer), the hook relay will immediately be activated in order to connect the line.

When clicked, the hook relay will be activated during the hunting period.

8.5.5.3.3 Delay before handling incoming call

When equipment are connected in parallel with the trunk line, it can be useful to delay the handling of incoming calls in order to wait for the paralleled equipment to answer. Enter a delay time in seconds in order to include such a delay.

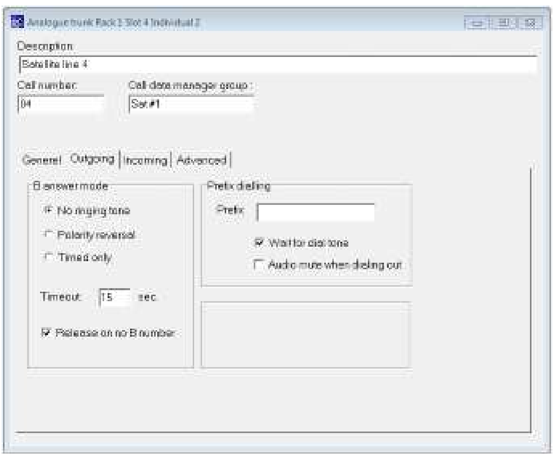
8.5.5.3.4 Incoming day mode / night mode

Two routing lists are available. One for day-mode and one for Night-mode. The mode is set manually by the user, by a telephone call to the Mode Select system call number.

5 call numbers can be entered into each of the Incoming routing lists. An incoming call will try to connect to the first item of the list, if no conversation is encountered within the time-out period stated in the time-out box, the next call number will be tried out. When all call numbers of the list is used, the incoming call will be disconnected.
A typically time-out waiting for the extension to answer a call is 30 seconds.
Routing can be taken to a ringing group (refer to 8.5.13) or a new dial tone can be started at the end of a hunting period (refer to 8.5.9).

8.5.5.4 Outgoing calls

How outgoing calls are handled can be determined by altering the set-up items found on the Outgoing tab page.



8.5.5.4.1 Prefix dialing

The Prefix dialing window contains digits to be dialed just after the line hooks off.
If no number is entered here, the Prefix dialing is disabled.
If the "Wait for dial tone", is checked, the dialing will start as soon as a dial tone is detected from the line.
If the Audio mute is checked, audio from the line will be muted while the prefix dialing is in progress, so that the calling extension will not hear the dialing.
Prefix dialing can be used, as an example, if the line is a satellite line, and you wish to use a predefined land earth station. These digits can be used to dial the access code for this predetermined earth station.

8.5.5.4.2 B-answer mode

This window contains information of how to detect the outgoing B answer. This is a very important set-up, as the B answer is used a the start signal for time counting in turn used for charge calculation of outgoing calls.

3 modes can be selected:

No ringing tone.

> The call progress tone decoder of the line will be used to detect the ringing tone. This tone will, on most telephone systems, be heard when the B part telephone is ringing. The B answer is detected when this tone is no longer detected. Make sure to set the time-out to a sensible value, as this B-answer mode may fail due to line quality, tone quality, tone frequencies etc.

Polarity reversal.

> The line feed detector is used to check if a line feed reversal is detected. This is the most safe type of B answer mode, but requires that the line connected has the ability to reverse the line feed on B answer.
> If no line feed reversal is encountered, B answer will be declared based on the time-out.

Timed only.

> Checking this button will hand over the B answer detection to the time-out only.

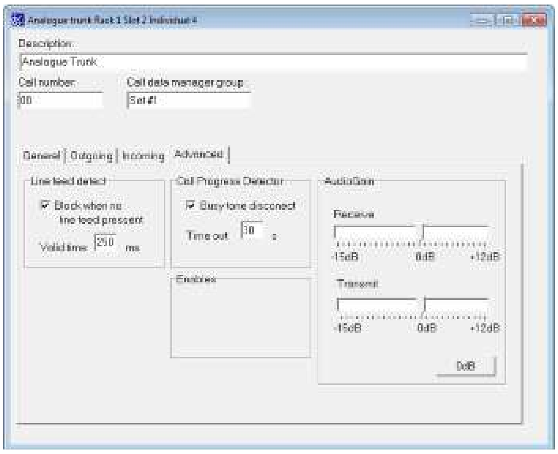
8.5.5.4.3 Release on no B number

When a call is made via an advanced trunk, digits to the line are send directly from the DTMF keyboard of the extension making the call. The audio tones from this A part telephone passes directly out onto the trunk line. In order to collect the digits dialed, the DTMF receiver of the FIO board is used. When the dial time-out expires, all the B digits collected will be stored as the destination B number, which in turn is used by the charge calculation.

Check the "Release on no B number" if the A extensions shall be disconnected if no digits are dialed at all.

8.5.5.5 Advanced options

A number of hardware set-up data are found on the Advanced tab page.



8.5.5.5.1 Line feed detector

The Line feed detect window is used to program the behavior of the line feed detector. The valid time states how long time a new line feed situation must be stable before it is really used for any actions.

Normally the line feed detect is used to set the line in a blocked state, preventing it from being used for any calls, whenever the line feed is not present.
If this feature is not desired, check the "No line feed block".

8.5.5.5.2 Call progress tone detector

The advanced trunk line makes use of the call progress detector (CPD). Check the "Busy tone disconnect", if the trunk line is to hang up when a busy tone is decoded for a period exceeding the Time out value.

This feature will prevent a hanging line if, for example, the A part extension making an outgoing call, fails to make a proper hook on after terminating a conversation.

8.5.5.5.3 Call DTMF tone lengths

The lengths of the DTMF tones generated by the FIO hardware can be set using the two boxes in the DTMF window.

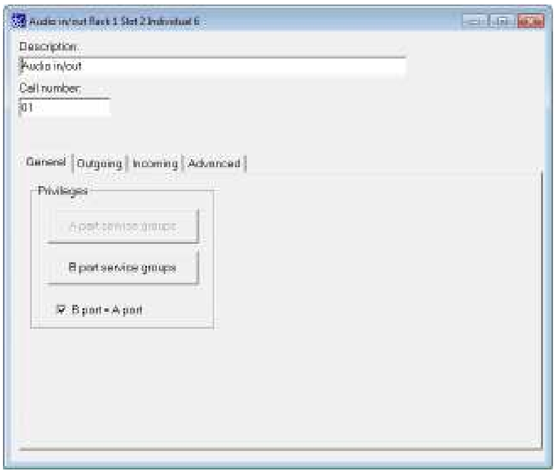
NOTE ! This does not change the lengths of the digits dialed directly from extension telephones, as these tones are generated by the DTMF transmitter of these telephones.

8.5.5.5.4 Audio gain

Using the sliders, the gain of the receive and transmit path can be adjusted.

8.5.6 Audio In/Out

The audio In / Out ports of the FIO4 are set up here

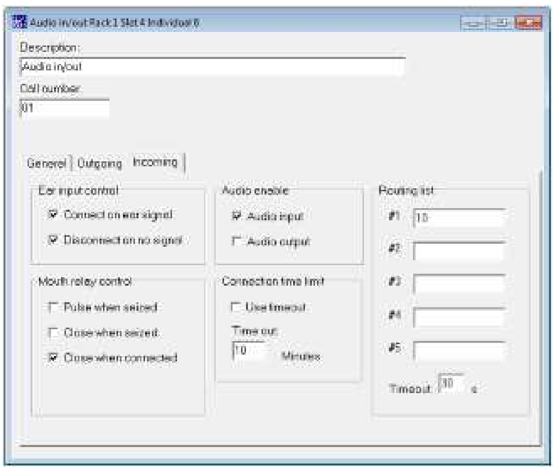


8.5.6.1 Call number / Description

These items are standard individual items found on all individuals

8.5.6.2 Incoming calls handling

Call from an Audio In/Out individual to another individual in the SeaCom system is considered as incoming calls. How the Audio In/Out handles incoming call can be set via the Incoming tab page.



8.5.6.2.1 Ear input control

Incoming calls can only be initiated via the digital input signal (Ear signal).

Clicking the “Connect on ear signal” makes the ear signal activate an incoming call.

Clicking the “Disconnect on no signal” makes the Audio In/Out terminate the call when the ear signal is no longer present. If this is not enabled, the conversation can only be terminated either by timeout or by the individual receiving the call.

8.5.6.2.2 Mouth relay control

The Audio In/Out can acknowledge actions by closing the relay output (mouth relay).

Clicking the “Pulse when seized” will make the Audio In/Out close the relay for a short time (approx 1 second) as soon as the incoming call is in progress.

Clicking “Close when seized” will make the Audio In/Out close the relay as soon as the incoming call is in progress. This overrides the above pulse setting.

Clicking “Close when connected” will make the Audio In/Out close the relay as soon as a conversation is established i.e. B-answer is obtained.

The relay will always open when the incoming call is terminated.

8.5.6.2.3 Audio enable

Use this box to enable audio incoming and outgoing depending on your need for the special application.

8.5.6.2.4 Connection time limit

Use this box if a time limit is to be set on incoming calls.

8.5.6.2.5 Routing list

This is a hunting list used when making incoming calls. When the Ear signal activates an incoming call, the topmost number will be dialed. The B-answer will be awaited fro the specified number of seconds, where after the next all number will be dialed.

8.5.6.3 Outgoing calls

Calls from an extension to the Audio In/Out individual is considered outgoing calls. How these calls are handled can be determined by altering the set-up items found on the Outgoing tab page.

8.5.6.3.1 Mouth relay control

How the mouth relay behaves is set by the check boxes of the Mouth relay control box.

Checking the “Pulse when seized” will make the Audio In/Out close the mouth relay for a short time (approx. 1 second) as soon as seized by any extension.

Checking the “Close when seized” will make the Audio In/Out close the relay as soon as seized. This will override the above setting.

Checking the “Close when connected” will make the Audio In/Out close the relay as soon as the conversation is established. The “hook of” signal is the ear input signal.

8.5.6.3.2 Ear input control

Checking the “Ignore ear input” makes the Audio In/Out connect directly when a seizure is made from an extension.

Checking the “Connect on ear signal” makes the Audio In/Out waiting for the ear signal to be present before connecting an outgoing call.

Checking the “Disconnect on no signal” makes the Audio In/Out terminate the outgoing conversation as soon as the ear signal is not present.
If this box is not clicked, the conversation can only be terminated by the extension calling the Audio In/Out.

8.5.6.3.3 Enable and multiple listeners

Use this box to enable audio incoming and outgoing depending on your need for the special application. Clicking the “Multiple listeners” check box will make the Audio In/Out accessible for more than one listener. The first extension dialing the Audio In/Out will be connected as if no multiple listening is enabled. The ear and mouth relay control is in use as selected. If another extension dials the Audio In/Out, this extension will be connected as listener only. The audio input to the Audio

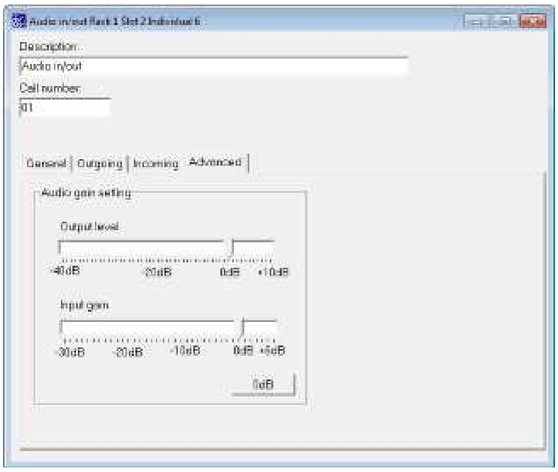
In/Out can then be heard by both extensions. There is no limit in the number of listeners. The Audio In/Out will disconnect the call when all listeners has terminated the listening. This feature is used for radio, music or VHF listening.

8.5.6.3.4 Connection time limit

Use this box if a time limit is to be set on outgoing calls.

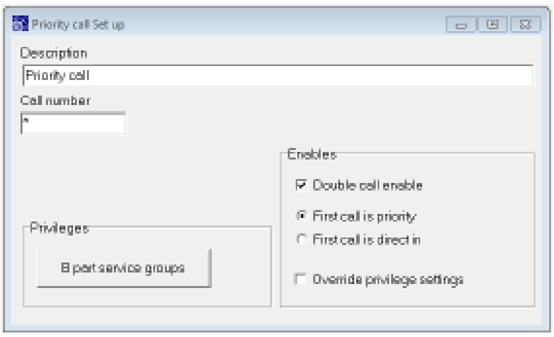
8.5.6.4 Advanced tab

Gains for the receive and transmit audio paths can be adjusted using the sliders on the advanced tab page.



8.5.7 Priority call

Installing a priority call system, gives the users of the system the possibility of making priority calls and making direct in calls to SC411, SC421 and SC211 stations. Priority calls or direct in calls are made by first dialing the call number of the priority call system, followed by the call number of the station to which priority or direct in is made.



8.5.7.1 Call number / Description

These items are standard individual items found on all individuals

8.5.7.2 Enables

Double call enable

Click this if both * and ** shall be valid dial numbers

First call is priority

When double call is enabled, clicking this will make dialing * only a priority call and dialing ** will be a direct in

First call is direct in

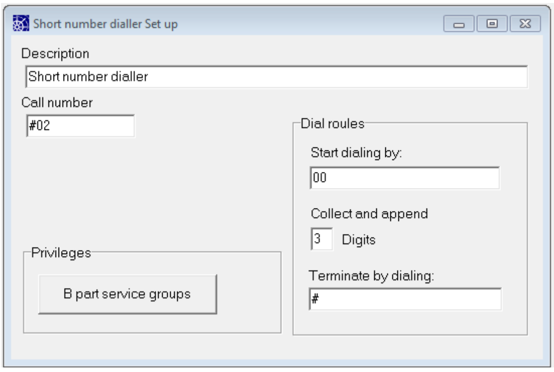
If this is clicked, * is dialed for direct in calls and ** dialed for priority

Override privileges settings

Click this if the normal access privileges shall be bypassed when a priority call is made.

8.5.8 Short number dialer

Global short number dialing can be implemented using the Short number dialer system. The short number dialer is a little intelligent, as the dialing can be divided into pre dialing, collect and append and terminate by. These features are used when dialing into VoIP systems.



8.5.8.1 Call number / Description

These items are standard individual items found on all individuals

8.5.8.2 Start dialing

This field contains the digits to be dialed by the short number dialer without any delay after calling. (could be a trunk line outgoing to a VoIP system)

8.5.8.3 Collect and append

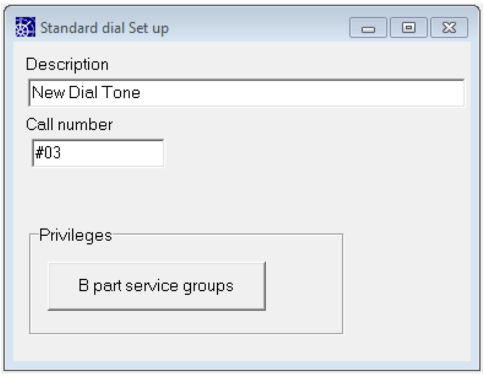
Set how many digits are then collected and appended to the dialing.

8.5.8.4 Terminate by dialing

After collecting this digit string is dialed. Typically used for the # key used by VoIP systems.

8.5.9 New dial tone

The New Dial Tone is used for making a new dial tone and starting a new dial sequence. This is most often used in conjunction with routing of incoming calls, where you need to give the caller a possibility of sub dialing.

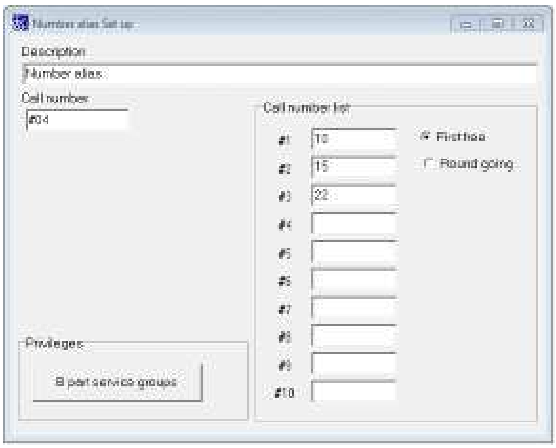


8.5.9.1 Call number / Description

These items are standard individual items found on all individuals

8.5.10 Number alias

The number alias system is used to convert a call number to another call number, or to group 10 call numbers into one call number with the possibility of converting to one out of 10 call numbers.



8.5.10.1 Call number / Description

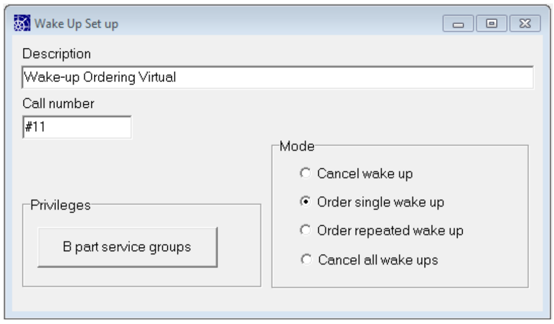
These items are standard individual items found on all individuals

8.5.10.2 Call number list

This window contains the 10 call numbers to convert to. Empty numbers are skipped.

8.5.11 Wake-up call ordering

The system includes a wake-up call system. In order to give the users the possibility of ordering a wake-up call, a Wake-up call ordering system must be installed.



8.5.11.1 Call number / Description

These items are standard individual items found on all individuals

8.5.11.2 Mode

This window contains selections for what a call to the function will do. The following is possible:

- Cancelling a wake-up call
- Ordering a single wake-up call
- Ordering a wake-up call repeated at the same time of the day for a number of days
- Cancel all wake-ups. This is a very power full mode, as invoking it will result in the wakeup ordered by all telephones

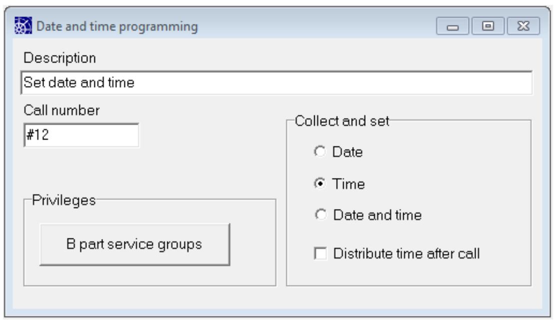
To implement all 4 functions it will be necessary to install 4 wake up call numbers, each taking care of one of the 4 functions.

The digit format of “order wake up” is HHMM

8.5.12 Set date and time system

The system includes the possibility of programming a call number to be the point of adjusting the date and time of the system. Date and time is used when logging call data, and when performing the wake up calls.

For the SeaCom 3000, the time must be set after each power up, unless an FIO4 board is present, as this board holds a Real Time Clock that has a 12 hours power backup, keeping the time running if no 24V DC power is present.



8.5.12.1 Call number / Description

These items are standard individual items found on all individuals

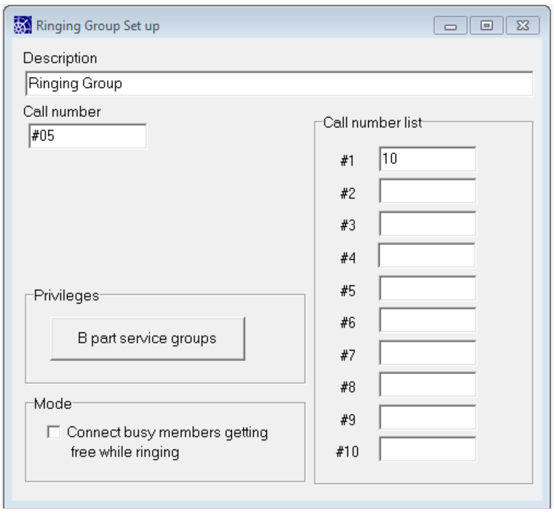
8.5.12.2 Collect and set

This pane is used to select what will happen when a call to the number is made. When the “Date” is checked the system will await the caller to enter YYYYMMDD.
When the “Time” is checked the system will await the user to enter HHMM.
When “Date and time” is checked, the system will await entry of form YYYYMMDDHHMM

The “Distribute time after call” shall be checked if the system has FSK receiving telephones displaying date and time. When the call to the Date and time programming is terminated successfully, the system will start distributing the time to all extensions, and free extensions will receive the time using an FSK transmission.

8.5.13 Ringing group

The ringing group is used when more than one telephone has to be ringing when dialing only one telephone number. This can be used, as an example, for incoming calls, where up to 10 extensions shall be able to answer the call. Another example: the work shop of the electrician has call number 11, the engine work shop has call number 12, and the engine control room has call number 12. These telephones can be grouped as “engine room” and call all together by dialing call number 10.



8.5.13.1 Call number / Description

These items are standard individual items found on all individuals

The call number field contains the call number to be used by the individual. Max 10 digits or characters can be entered.

The privileges window contains a button for opening the B part service groups.
Use the service groups to restrict the access to the ringing group.

8.5.13.2 Call number list

This window contains the 10 call numbers which are to be included in the ringing group. Only extensions can be a part of a ringing group.

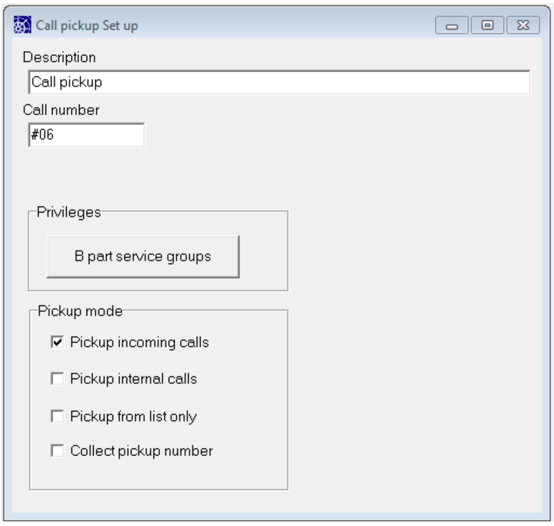
8.5.13.3 Connect busy members…

If an extension is busy when a ringing group is activated, the activation process will skip the extension. Ringing will only be send to non busy extensions. This is the case when the checkbox ‘connect busy members getting free while ringing’ is not checked.

When checking the checkbox ‘connect busy members getting free while ringing’, extensions, belonging to a ringing group which is currently active ringing, will receive ringing if it gets free during the group ringing, and can be the extension answering the ringing group.

8.5.14 Call pickup

Call pickup is used when a user want to take over a call to another extension. An example can be an incoming call which is directed to a cabin close by. The user can hear the telephone ringing. Then the call pickup number can be dialed, and the incoming call is connected as if the telephone making the call pickup was the ringing telephone. Ringing groups can be picked up.



8.5.14.1 Call number / Description

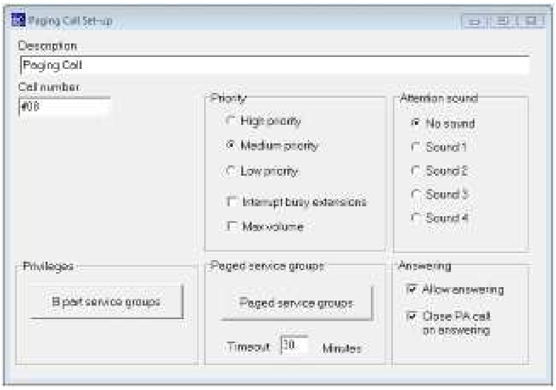
These items are standard individual items found on all individuals

8.5.14.2 Pickup mode

This pane is used for selecting types of calls to pick up. When using the “Pickup from list only”, pickup is only performed if the ringing extension or the extension or trunk line making the call is found in the Call number list. When choosing the “Collect pickup number”, the user has to enter the call number of the calling or the called party.

8.5.15 Paging call

Paging call is used when activating the SC211, SC411 and SC421 for an acoustic paging call. Many paging calls having different properties can be programmed, and several paging calls can be conducted simultaneously. Paging and alarm calls share the same group selection mechanism and shares the priority structure implemented.



8.5.15.1 Call number / Description

These items are standard individual items found on all individuals

8.5.15.2 Paged service groups

The service group system is used to create paging groups. The A part service groups of the Paging call system is used for assigning a group, and the B part service groups of the extensions going to participate must be set for the same service group.

In order for simplicity, one page of the service group editor is devoted paging and alarm calls service groups. Choosing this page for implementing the paging groups is recommended although not necessary.

A timeout can be set so a paging call will not hang for a long time.

8.5.15.3 Priority

The paging call can be assigned a priority level. When extensions participating in a paging call receives a paging or alarm call with a higher priority, the extension will switch over to the higher priority paging or alarm call.

It is possible to select “interrupt busy extensions”. This causes ongoing conversations to be temporarily broken by the paging call. When the paging call ends, the conversation is re-established.

Setting “Max volume” will cause the paging call to override possible volume settings in the bed side speaker panel. Speaker panels can be used for listening to music during which it can be desirable to reduce the volume. If a high priority paging call has the “Max volume” clicked, the attenuation will be disabled during the paging call.

8.5.15.4 Attention sound (Gong)

When making an acoustic paging, it can be convenient to call on attention before sending the spoken message. This can be done by choosing one of the “gong” sounds available. The gong will sound for 2 seconds.

8.5.15.5 Allow answering

When this check mark is set, the parties called by a PA call can make an answer to the initiator of the PA call by hooking off the telephone.

8.5.15.6 Close PA call on answering

When a PA call is answered by one of the called parties hooking off, the conversation will be distributed to all speakers sounding the PA call. This can be used to form a large intercom group. When another party makes hookoff, it will take over the conversation to the PA call initiator, and the full PA group will listen. This feature works like a wired VHF radio system, where everybody is listening, and one part only is speaking to all others.

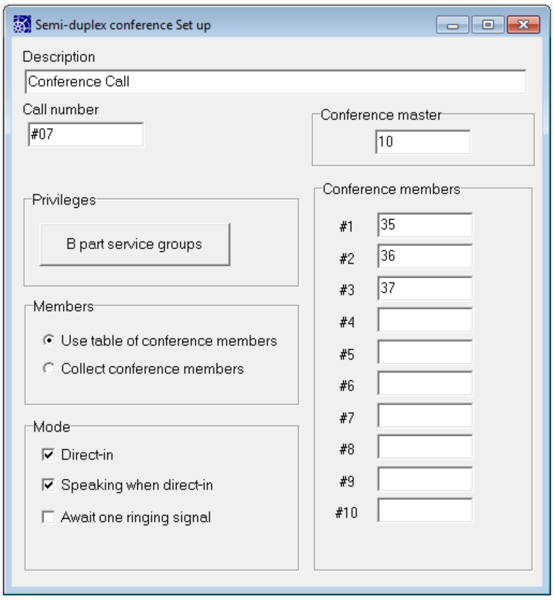
If the Close PA call on answering is checked this feature is disabled, and an answer to a PA call will only establish a two party conversation between the initiator and the answering party.

8.5.16 Semi-duplex conference

This system call number is used to implement a conference call between one master initiator and a group of up to 10 substations. The substations can be of any type, but the SC211 and the SC411/SC421s are the best choice for substations.

When using a SC411/SC421 as master station and the SC211 as slaves, a full talk-back functionality is obtained.
In this configuration a SC211 can be called, and opened for speaking, directly from a SC411 and the speach direction is controlled by the MIC button of the SC411. We call this a"talk-back" call or conversation.

The master control the speach direction by the MIC key, and the speaking substation can be selected by pressing the number keys 1..9.



8.5.16.1 Call number / Description

These items are standard individual items found on all individuals

8.5.16.2 Conference master

In a talk-back set up with a SC411 as master and one or more SC211 stations as slaves, the call number of the master station shall be entered in this field. This will be used when any of the slave station calls the semi-duplex conference number, in order to direct such a call to the master of the talk-back set up. The master station will be ringing, and when the master station answers the call, the system will disconnect and set up a talk-back call to all conference members, just as if the master has made the call to the conference.
This results in the speech direction to be controlled by the microphone (PTT) button of the SC411 master station.

The feature is also referred to as “reverse talk-back” calls.

8.5.16.3 Members

The conference can be set up with a fixed group of members by checking the “Use table of conference members”. In this case the call numbers of the members are entered in the list of conference members.

When the “Collect conference members” are checked, then the station initiating the conference must dial the call numbers of the conference members that are to participate. The list shall be terminated by dialing #.

8.5.16.4 Mode

Members of the conference can be called in two ways. Un-checking the “Direct-in” will make a conference call start by the conference members ringing, just like a normal telephone call. The ringing signal will be 3 short ringing tones, so that the conference members know that the caller tries to set up a conference. The members must answer the call to be participating.

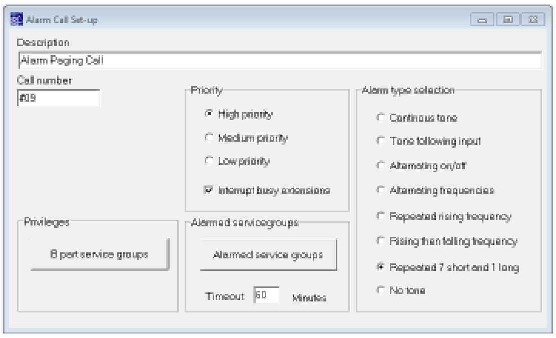
When checking the “Direct-in”, the members of the conference will enter the conference automatically without any hands on. This is used when implementing the classically talk-back function. When checking the “Direct-in”, the check box “Speaking when direct-in” becomes visible. This check box determines the speech direction initially set when the call is made. Check the “Speaking when direct-in” in order to force the speech direction to be from the initiator to the called conference members.

8.5.17 Alarm distribution call

An alarm call system call number is used when activating the SC211, the SC411 or the SC421 for an alarm paging call. Many alarm calls having different properties can be programmed, and several alarm calls can be conducted simultaneously.

NOTE ! There is only on alarm generator on each extension board.

Alarm calls and paging calls share the same group selection mechanism and shares the priority structure implemented.



8.5.17.1 Call number / Description

These items are standard individual items found on all individuals

8.5.17.2 Alarmed service groups

The service group system is used to create alarm groups, selecting the extension lines which are participating in the alarm call. The A part service groups of the alarm call system is used for assigning a group, and the B part service groups of the extensions going to participate must be set for the same service group.

In order for simplicity, one page of the service group editor is devoted paging and alarm calls service groups. Choosing this page for implementing the paging groups is recommended although not necessary.

A timeout can be set so an alarm call will not hang forever. Set to zero if no timeout is wanted.

8.5.17.3 Alarm type selection

The alarm call can be chosen to send one out of 7 alarm signals:

Alarm type	Technical details
Continuous tone	700 Hz
Tone following input	700 Hz
Alternating on/off	700 Hz 500 ms tone 500 ms pause
Alternating frequencies	700 Hz sounding 500 ms 1200 Hz sounding 500 ms
Repeated rising frequency	Lowest frequency 600 Hz, Highest frequency 2400 Hz Period 1s
Rising then falling	Lowest frequency 600 Hz, Highest frequency 2400 Hz Period 3s
Repeated 7 short and one long	500 Hz Short tone/pause 500 ms Long tone 3s Long pause 6s

8.5.17.4 Activating alarm by relay

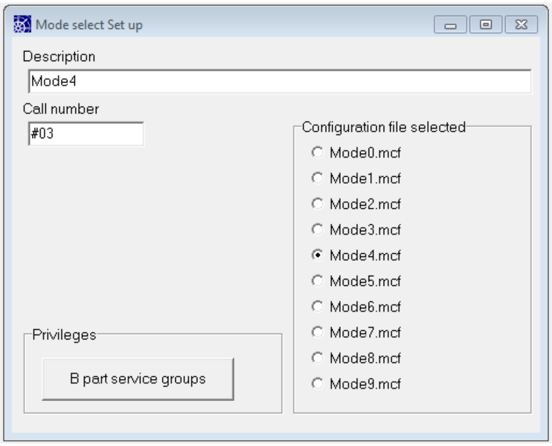
The alarm type “Tone follow input” is used when an external equipment has to control the tone pattern. The 2 wires of the extension line used as alarm generator is connected to the dry relay contacts of the master alarm system. When the relay is closed, the extension must be set up to call the alarm number representing the “Tone follow input” alarm call. The alarm call will be set up, and tone is send to the alarm group as long as the relay stays closed. When the relay is opened, the tone is stopped. If the relay stays open for more than the timeout period of 5 seconds, the alarm call be stopped.

8.5.17.5 Priority

The alarm call can be assigned a priority level. When extensions participating in an alarm call receives a paging or alarm call with a higher priority, the extension will switch over to the higher priority paging or alarm call.

8.5.18 Mode select system call

The mode of the system will be selected by a user calling a mode select system call number.



Selecting the mode of the system is a very power-full thing to do. When a user calls the mode select system call, the chosen mode configuration file will be copied into the operating mx.mcf file. After the call, the system waits for all conversations and connections to be terminated. When no connections are active, the system opens the selected mode file and checks that the content will match the operating mx.mcf file. This means that exactly the same boards, individuals and system calls must be found in both files and in the same order. In praxis this means that the mode files must be created based on copies of a full featured mx.mcf file. If the contents are matching, all the selected items of the mode file will be copied into the operating mc.mcf file. The system will hereafter reinitialize all boards and system calls to make the modified mc.mcf file properly operating.

8.5.18.1 Call number / Description

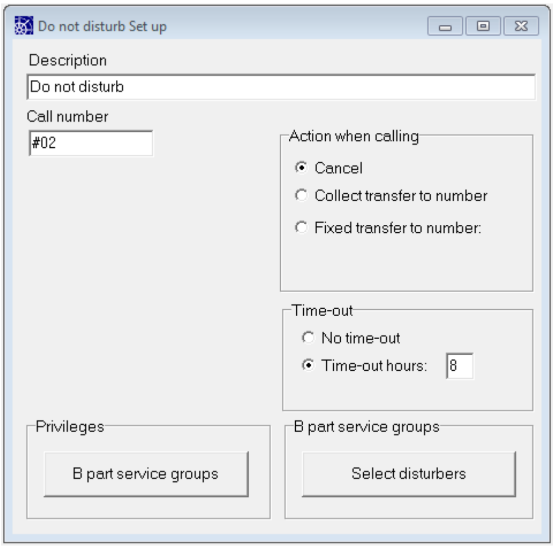
These items are standard individual items found on all individuals

8.5.18.2 Configuration file selected

The mode files must have the name: Mode0.mcf, Mode1.mcf …. Mode9.mcf. Choose the file to be activated by selecting the name of the file in this box.
In praxis You have to assign names like “Day mode”, “Night mode” etc. to each file.

8.5.19 'Do not disturb' system call

An extension can enter a 'do not disturb' mode, where incoming calls are refused or transferred to another call number. By making a call to a do not disturb system call.



8.5.19.1 Call number / Description

These items are standard individual items found on all individuals

8.5.19.2 Action when calling

The system call can be used for 3 purposes:

Cancel	Used when extension wants to exit the 'do not disturb' state.
Collect	When this button is checked, the caller must follow the call by dialing a valid call number to which calls will be transferred during the 'do not disturb' condition. If the call number is valid, the call will be terminated by the user hearing a dial tone. If the call number is invalid, the caller will hear a busy tone.
Fixed	When checked a valid transfer to number must be entered. Calls to the extension in 'do not disturb' state will be forwarded to this number.

8.5.19.3 Time-out

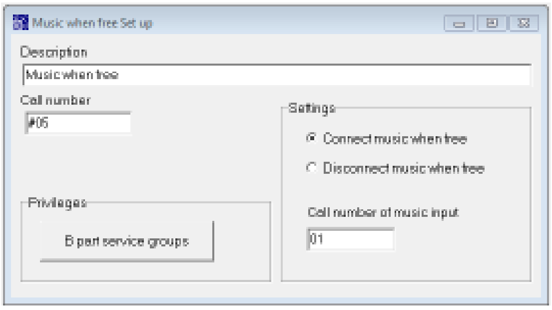
Enable and set the timeout I the call shall start a timeout period for the 'do not disturb' condition. This is used to prevent telephones from being permanently in 'do not disturb' in the case that the user forgets to cancel the 'do not disturb' ordered.
Valid times are 1.. 23 hours.

8.5.19.4 Select disturbers

Push this button to open a service group editor for selecting access for disturbers. If no groups are selected, no-one can disturb unless using a priority call.
This is used for allowing selected extensions or trunk lines to override the 'do not disturb' situation.

8.5.20 Music when free

Stations having a speaker can request listening to music from and FIO audio in channel.
To request music when free and to stop music when free, the Music when free call number must be installed.



8.5.20.1 Call number / Description

These items are standard individual items found on all individuals

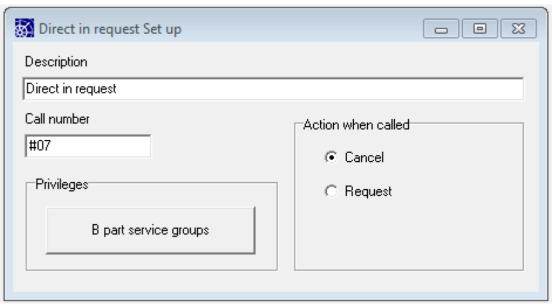
8.5.20.2 Settings

A system call number of this type can be used for either requesting music or for stopping music. The purpose is determioned by the choice in settings:

If Connect music when free is checked, then the Call number of music channel must be filled with a valid call number music input.

8.5.21 Direct-in request

Stations having the possibility of operating in hands-free can make use of this system call number to order hands-free hook of to conversation when a call comes in.
The SC411 and SC421 stations and the SC325 do have this ability.



8.5.21.1 Call number / Description

These items are standard individual items found on all individuals

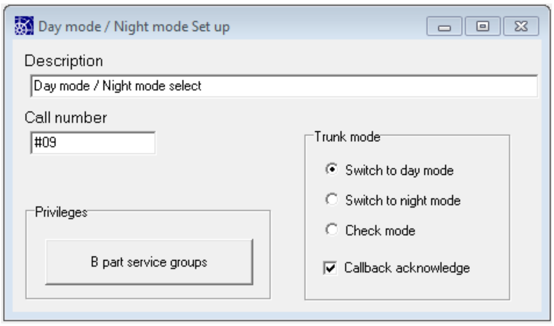
8.5.21.2 Action when calling

If request is checked, the following calls will open the station direct into hands free.

If Cancel is checked, then the following calls to the station will make it ring as normal

8.5.22 Day mode / Night mode

This system call number gives the user a possibility of manually selecting day or night mode for all trunk lines. This is used for incomming calls routing. When a call to a day mode selection is made, all trunk lines will choose the daymode routing list, until a call to the night mode selection is performed, whereafter all trunk lines will use the night mode routing list.



If a telephone with display is used, a return acknowledge ringing will be made, so the user can see which mode is currently selected.n

These items are standard individual items found on all individuals

8.5.22.1 Action when calling

The system call can be used for 3 purposes:

- Switch to day mode
- Switch to night mode
- Check mode

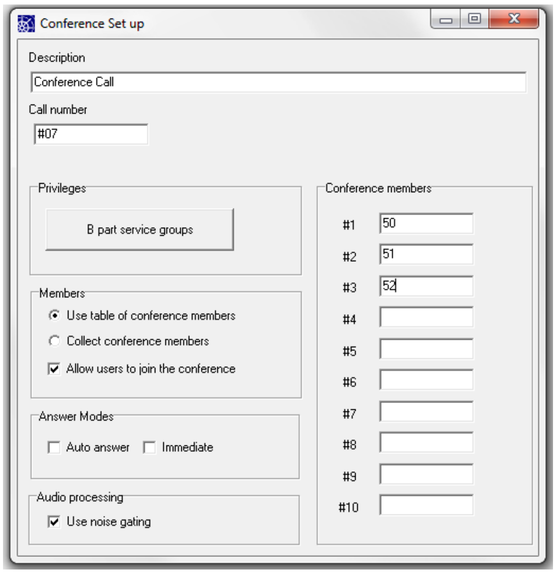
8.5.23 Conference

The conference call is used for setting up a conference of 1 initiator and up to 10 substations. In a conference call, all stations can listen to the conference, and all stations can speak to the conference, all at the same time.

The conference set-up form is shown in the figure below.

8.5.23.1 Call number / Description

These items are standard individual items found on all individuals. But for the conference call, the text in the description box will be shown on the display of the substations called. So if extension number 10 makes a conference call to 50 and 51, these stations will see the "Conference call" in their display, and not the "Extension 10" as for normal calls.



8.5.23.2 Members

The conference can be set up with a fixed group of members by checking the “Use table of conference members”. In this case the call numbers of the members are entered in the list of conference members.

When the “Collect conference members” are checked, then the station initiating the conference must dial the call numbers of the conference members that are to participate. The list shall be terminated by dialing #.

If the "Allow users to join the conference" is checked, the it is possible to join an ongoing conference by calling the "#07" call number.

8.5.23.3 Answer Modes

Members of the conference can be called in two ways. Normal ringing or “Direct-in”. When no answer modes are checked, then parties called into a conference will just ring like a normal telephone call, and participating in conference starts when the ringing is answered by hook off. The ringing signal will be 3 short ringing tones, so that the conference members know that the caller tries to set up a conference. For stations with display, the name of the conference will be displayed. Defaults to "Conference call".

When checking the “Direct-in”, the members of the conference will enter the conference automatically without any hands on.
For the SC411 and SC421 there are two ways of direct-in. When checking the "Immediate" check box, the call will open the SC411 and SC421 before any ringing, and no display will be in use. When un-checking the "Immediate", the SC411 and SC421 will make on ring, and the display will show the name of the conference before hooking off into hands free.

8.5.23.4 Audio processing

When many telephones are in conference, the system will add the audio coming from all microphones into one common audio signal send to the speakers of all participants. This can lead to very much noise added.
When checking the "Use noise gating", a noise gate is put into the signal path from each of the participating microphones. The noise gate reduces the signal from those telephones not speaking, and it amplifies the signal from those telephones speaking.

9. LSP details

This chapter is for advanced technicians only. The LSP runs Linux. This chapter describes how the operating system and its files are organized. The explanations follows the figure in the bottom of this page.

9.1 Folders

As seen from the user. the system has the following main folders:

- Operating (/home/pi)
- Configuration (/home/pi/Configuration)
- Manual (/home/pi/Manual)

The following will describe their use and how the folders are shared.

9.1.1 Operating

This folder is the place where the system executes from when running.

It is a write protected place, and it is shared as read only, so users can watch and copy, but are not entitled to change anything in this folder.

9.1.2 Manual

This is a folder for default software, and good to have tools, for example the VNC-Viewer. And it is the place for a copy of this manual.

The folder is shared read only.

9.1.3 Configuration

This folder is used to hold the **Configuration.exe** tool software and the **configuration_file**.

The folder is shared read/write, and the user can execute the **Configuration.exe** from this folder, when connected from a Windows computer.

When an edited configuration file is saved, it will be saved into the **Configuration** folder directly to the SD card.

9.1.4 USB

The USB disk is a removable disk. It is auto mounted when inserted, and is fully read/write. It is not shared.

9.2 Write protecting the SD card

In order to protect the system from accidentally corrupting, and to reduce tear and wear on the SD card, an overlay system is in use.

The overlay is working like a transparent folio on an art- painting. When laid out, you can see the painting, and You can paint on it, but when removing the folio, the original art-painting is there again.

This is also the case with the file overlay. When Linux boots, it lays out the overlay. After that, any writing to the SD card will not go to the card itself, but go to the overlay (which is in RAM).

The overlay covers the Linux operating system, the Manual folder and the Operating folder, but it does not cover the Configuration folder.

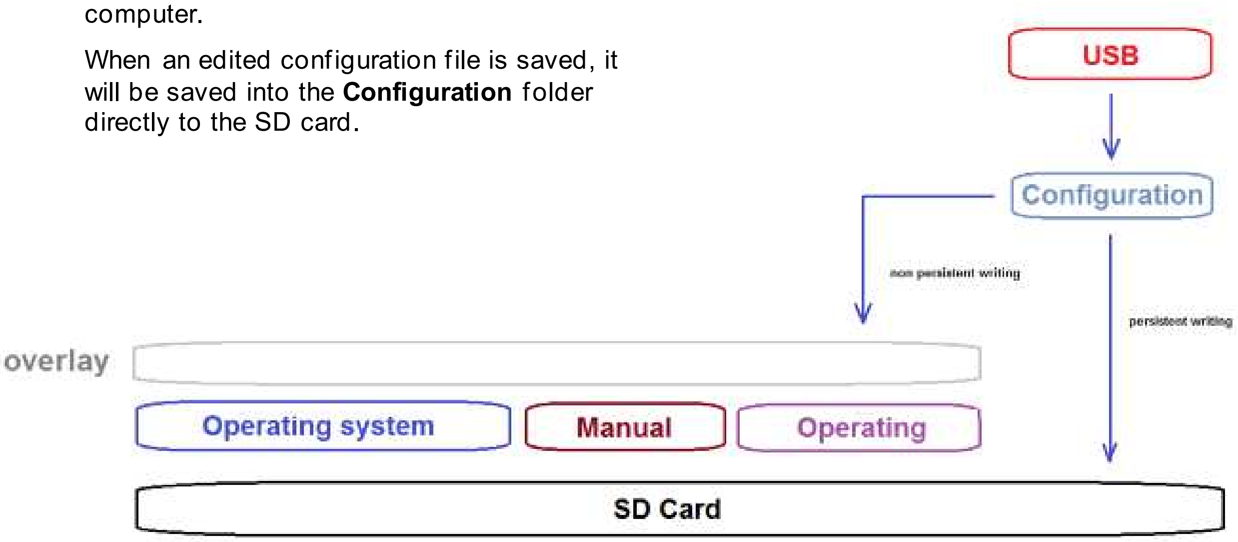
9.3 System start up

When the system boots, the first thing is to lay out the overlay, so nothing can really be written to the SD card.

Now the Linux operating system is started.

Then a copy function copies everything found in the **Configuration** folder to the **Operating** folder. This includes the **Proces.exe** and the **configuration_file**. The copies are put into the overlay. If no files are copied, the system will use the files stored in the **Operating** folder below the overlay.

After copying, the **Proces.exe** is started running in the **Operating** folder.



9.4 File change monitoring

The system is set up with some file change monitoring in force. This is in order to make a use full file flow when altering the **configuration_file.**

9.4.1 Operating folder

When the CP software finds that a **configuration_file_new** is being placed in the **Operating** folder, it initiates a close down-and-reboot sequence.
During reboot, the **configuration_file_new** is copied to **configuration_file,** and the **configuration_file_new** is deleted.
On a running system, this is all done in the overlay.

9.4.2 Configuration folder

The system is set up with a file change monitor script supervising the **Configuration** folder. When a **configuration_file_new** file is being placed in this folder, the file change monitor script will make a copy of it to the **Operating** folder, and then delete it.

9.4.3 USB stick

Inserting an USB stick in the USB port will trigger the USB monitor. The USB disk will be mounted on the Linux file system, and a check to find a **configuration_file_new** file on the disk is performed.

If a **configuration_file_new** is found, then this file is copied into the Configuration folder as both **configuration_file** and **configuration_file_new**, and the file is deleted from the USB stick.

If not the **configuration_file_new** is found, then thes USB monitor will make a copy of two files: **configuration_file** and the **Configuration.exe** both taken from the Operating folder.
This leaves on the USB stich to the user both the running **configuration_file** and the tool to edit it

9.5 Ethernet settings

The Ethernet port is set up to be a port for connecting a laptop computer for programming and service of the system.

The system is given the fixed IP address:

192.168.0.53

and the computer name

phonesystem

the user name is

admin

and password

1017

9.5.1 DHCP server

In order to make access from a laptop connected directly to the Ethernet port, a DHCP server is running on the system.

This means that connecting to a full network should be avoided or at least done with care.

9.6 Making the SD card read/write

If it is needed to make some changes to the operating system, it is possible to change the SD card to read/write and back to read only.

This is done by taking the SD card out of the LSP, and inserting it into a computer that can read the card. File system is FAT32, so Windows computers can also read it.

Open a text editor and open the file **cmdline.txt**.

The first line of this file will read:

boot=overlay dwc_otg.................

Remove the *boot=overlay* in order to make the SD card read write, and reinsert the text to make the SD card read only.

On the next boot of the system, the SD card will be protected or not protected by the overlay depending on the presence or not presence of the *boot=overlay* text.

10. List of parts

This chapter holds stock number of parts.

Item description	Order nmbr
SeaCom 3000	10-092-0305
AEXT16-8	10-110-2020
AEXT16-16	10-110-2021
AEAXT16-24	10-110-2022
FIO4-2	10-110-1402
FIO4-4	10-110-1404
CTU24-8	10-110-1208
CTU24	10-110-1224
Ribbon cable 120mm	20-121-0120
Ribbon cable 350mm	20-121-0350
Plastic card guide	10-130-0000
Board compartment set	10-400-1105
PIM	20-110-0070
LSP	20-110-0301
SD card with system software	20-132-1100
SC211	10-102-0211
SC220	10-102-0220
SC411	10-102-0411
SC421	10-102-0421
TX325	10-400-0050
SC325	10-400-0055
Handset for SC220 and SC421	10-400-1020
Handset mounting kit	10-400-1025
Handset flush mount for SC411	10-400-1050
Front cover door	10-400-1040
10W horn speaker 8ohm	10-501-0100
Headset with 10m cable	10-400-0205
Beacon sounder blue	10-400-0404
Beacon sounder white	10-400-0405