SeaCom System Manual

The manual for the SeaCom 2100 maritime intercom system

- Covers SeaCom 2100 telephone system
- Covers SC211, SC220, SC411 and SC421 stations
- Covers the Windows based configuration tool
- Valid for the SeaCom 19" systems as well

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

DNV

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.

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Form code: TA 251   Revision: 2021-03   www.dnv.com   Page 1 of 4

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

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.

DNV

Job Id: **262.1-011116-4**
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 as Your supplier of wired on board communication.

This manual will introduce You to our maritime communication system, and take You through all parts of the process from evaluating the needs for communication through designing and building a system towards commissioning and delivery to the end user.

As well as being a reference for finding technical data, we have added a handful of chapters giving good to know information, and explaining good installation praxis which have been proven to work in real life.

We hope You will enjoy working with our system.

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, SeaCom 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.

2. 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 2100 is enclosed by a heavy steel box. Handle these with care and mount the properly fastened, so they do not drop of their brackets.

- 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.

3. System description

This chapter is meant for first time users of the SeaCom 2100 communication system.
It will give an overview of what You can do with the system, the features You will find, and overview of the system components involved.

3.1 Purpose of the system

The purpose of the SeaCom 2100 maritime communication system is to give the 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.

3.2 Operational features

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

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

3.2.1 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.

3.2.2 Distribute satcom

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.

3.2.3 Talk back

Our communication station SeaCom 211 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.

3.2.4 Conference

The system allows for conference calls with one initiator and up to 10 participants. Users can set up a fixed conference, or the conference can be dynamic created, left and joined.

3.2.5 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.

3.2.6 Time distrib 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, or the SeaCom communication system can receive the time from a central clock system using serial NMEA communication.

Based on the system time, wake up calls can be ordered from any telephone connected to the system.

3.2.7 Call Data Management

A ship owner that wants to register calls made via the satellite equipment connected can make use of our Call Data Management system.

The seamen are given accounts and PIN codes which are used to open the access to the satellite lines. When conversations are conducted, call data records are stored on the system disk. Based on these stored data, the responsible officer can print invoices covering the telephone calls made.

3.3 Main system components

A SeaCom installation consist of the following main components:

- The exchange
- Communication stations
- Telephones

3.3.1 Exchange

This is the central component of the system. All telephone lines are taken out of this box from its extension line boards, the satellite lines are connected to its trunk boards, and this box also distributes power to the communication stations via the signal cables.

SeaCom 2100

Maritime telephone system with up till 256 telephone and communication station positions.

SeaCom 19"

Maritime telephone system with up till 800 telephone and communication station positions.

3.3.2 Exchange circuit boards

In order to build exchanges that suits the customers needs, a range of circuit boards are available for inserting into the basic exchange systems. Each board adding features as needed.

AEXT8

A circuit board adding 8 analogue extension lines

3.3.3

3.3.4 AEXT16

A circuit board with 8, 16 or 24 lines

FIO2

A circuit board adding 2 trunk lines (satellite) channels to the system.
This board can also be reconfigured for other IO/ purposes such as audio in and out.

FIO4

A circuit board adding 2 or 4 trunk lines (satellite) channels to the system.
This board also holds audio I/O and a lot of relay functions.

PDU

Power distribution. Used for distributing the 24V DC power to stations.

CTU2

Cable termination unit. Used for terminating ship cabling into the system.

3.3.5 Communication stations

The communication stations are advanced telephones mounted at each location that need communication. Typically locations like bridge, Engine Control Room and Steering Gear Room where a simple telephone will not do.

The following communication stations are available:

SeaCom 211

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

CTU24

Terminating 24 telephone lines.

SeaCom 411

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

SeaCom 421

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

SeaCom 220

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

3.3.6 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.

3.3.7 Power supply

The SeaCom communication system is powered by 24V DC only. Two inputs are available:

- Main input (from AC/DC source)
- Battery (from ships radio battery)

3.4 Typical installation example

The figure shows an example of how the
SeaCom communication system can be
architectured.

3.5
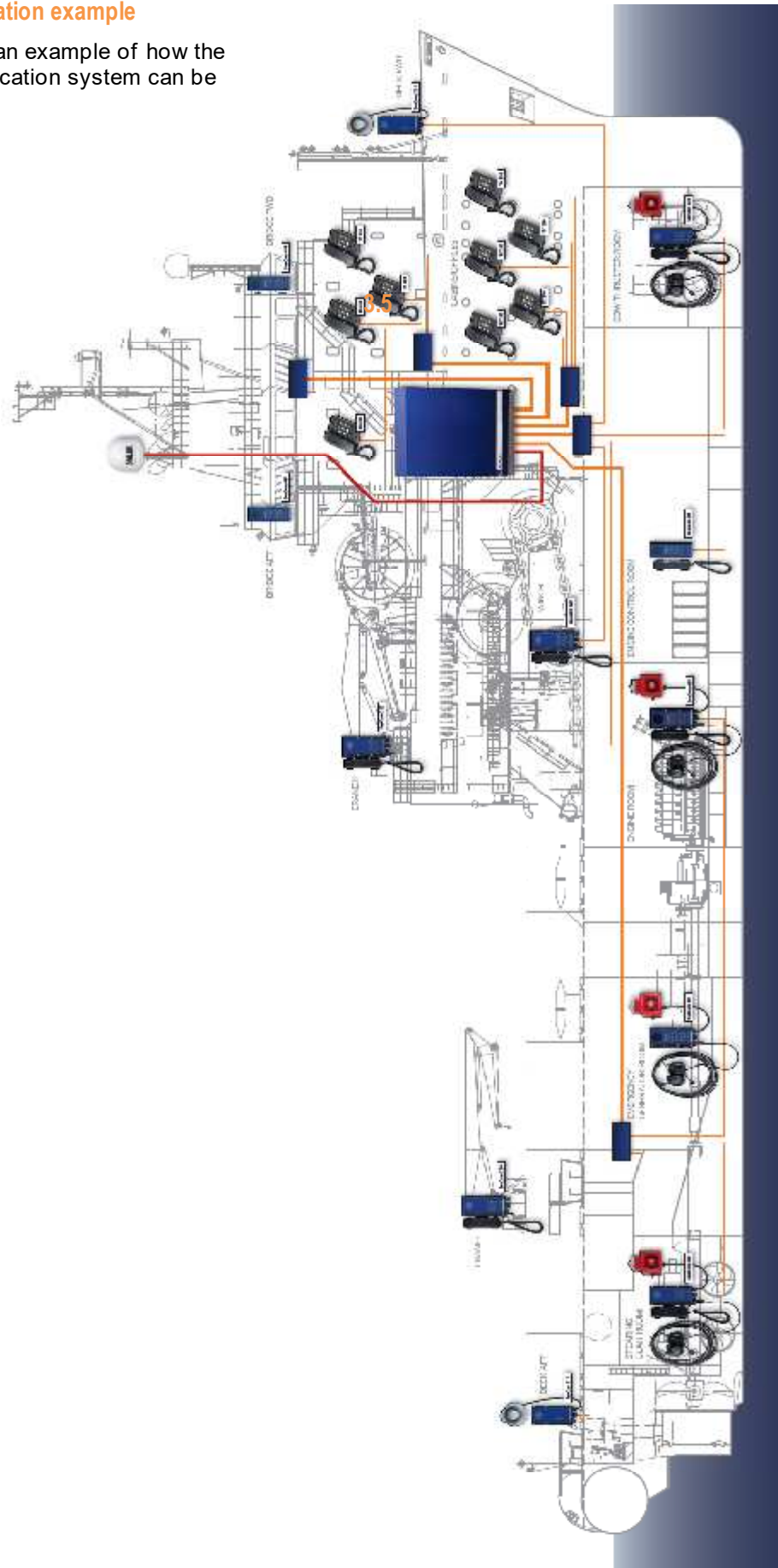
4. Designing a system

The intentions of this chapter is to be a guide to how to approach the task of designing 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 the components.

4.1 Analyzing

All ships have different needs. This is what makes design and installation of on board communication challenging. The SeaCom 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 location 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.

4.1.1 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.

4.1.2 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.

4.1.3 Type of communication

On each location it must be chosen which kind of communication is most convenient. 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.

4.1.4 Satellite lines

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

4.1.5 Audio I/O

Count how many I/O channels are used for radio music and output to external amplifier units etc.

4.2 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 maritime communication system.
The below describes which type of station to be used in each type of environment.

4.2.1 Analogue telephones

Plain analogue telephones are to be used in protected areas only. The TX-325, has the below operating range:

Hands free up to 65 dB noise

Handset conversations up to 75dB of noise.

4.2.2 SeaCom 411

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.

4.2.3   SeaCom 211

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.

4.2.4   SeaCom 421

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.

4.2.5   SeaCom 220

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.

4.3   Selecting the exchange system

Based on the analysis , the basic exchange system must be selected, Either the SeaCom 3000 (refer to own manual), the SeaCom 2100, or our 19'' systems.
And the circuit boards to be added to this basic system must be chosen.
The choice depends on how many extension boards and trunk boards that are needed, and the number of CTU'2 (cable termination units) and PDU's (power distribution units) needed

to actually terminate the ship cables for the purpose.

4.3.1   Choosing AEXTxx-xx

The total number of extension lines including lines for the stations SeaCom 411, SeaCom 211 and SeaCom 421 shall be counted.
Add 8, 16 or 24 lines AEXTxx-xx cards to meet the requirement.

4.3.2   Trunk and audio I/O (FIOx

Next count the number of 2 wire trunk lines needed for connecting satellite communication equipments, and the number of audio in/out needed for connecting music and external amplifiers to the system.
Add the required number of FIO2 or FIO4 to meet the requirement.

4.3.3   Cable terminations (CTU)

Now all the AEXT cards and FIO card need cable termination units. The CTU2 terminates 16 telephone lines and 2 trunk lines, whereas the AEXT24 terminates 24 telephone lines only.

 If audio I/O and relay functions of the FIO cards are in use, add CTU's to support this as well.

4.3.4   Power distribution (PDU)

The number of stations requiring 24V DC power must be counted. These are the stations SeaCom 411, SeaCom 211and SeaCom 421. If the number is 4 or less, the auxillary output of the power input module (PIM) found internal to the exchange systems can be used for powering these stations. If the number is 5 or more, a PDU (power distribution unit) must be added for each 32 stations.

4.3.5   Conclusion

Finally it must be checked that the number of cards and CTU's will fit within the chosen SeaCom system. For the SeaCom 2100 and 19'', a maximum of 18 card spaces are available, one of which must be the CP card. And the maximum number of CTU's and PDU's are 16.

4.4 Assembling the exchange

The exchanges must be assembled according to the customer requirements.
This paragraph takes you through the assembly process.

4.4.1 Card guides

In order to insert circuit boards into the board magazine, plastic card guides must be used.

These are mounted in "click in" fashion at the positions requiring a circuit board.

4.4.2 Inserting boards

Then slide the boards into position and make sure that they are properly pushed into the backplane connector.

Be sure that the FIO2 boards are jumped according to the intended use. Refer to chapter 10.3.5

4.4.2.1 Board positions

The figure below shows the boards and their typical locations. PSU, CP2 and the FIO2Master cannot be moved. Other boards can be located at any free position.

E x t e n s l o n s   CP trunk   Power

4.4.3 Mounting CTU's

The CTU's are then mounted using 4 3x6mm screws.
Use labeling for documentation.
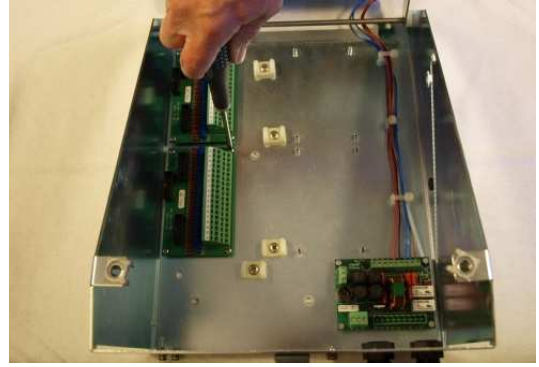
4.4.4 Mounting the ribbon cables

The boards of the boards magazine are connected to the CTU's using 16 wire ribbon cables. These cables are taken over the steel rod over which the rack frame swings, and secured using strips.

Extension boards are wired to the A1 or A2 connectors of the CTUs, as well as are FIO2 boards which are not entirely used as trunk interfaces.
FIO2 boards configured as two analogue trunk lines can be terminated at the T connector of a CTU board, disregarding the use of the A1 and A2 connector.

It is the responsibility of the installer to make a proper documentation for the internal wiring.

4.5 Power

The exchange system and the communication stations need 24V DC power for its operating, and it is strongly recommended that this power is backed up by a 24V DC battery, typically the radio battery of the ship.
Therefore the PIM module has two power inputs: M+ - and B + -

A PDU (power distribution unit) is used to distribute 24V DC to the stations.

4.5.1 Mounting PDU's

The PDU's are mounted, and the power input connected to the auxiliary power output of the PIM module, for maximum 10 stations, or directly to the 24V DC power supply, if larger number of stations are to be powered.
A rule of thumb is that 10 stations can be powered via the PIM module auxiliary output.
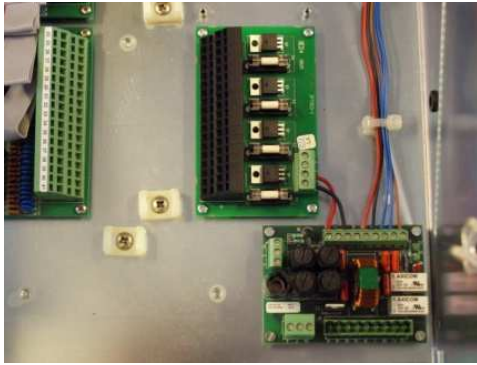
The picture shows the PDU and the wiring to the auxiliary output of the PIM module.

4.5.2 Fusing

IT IS THE RESPONSIBILITY OF THE INSTALLER TO MAKE PROPER FUSING

When assembling a system, the proper fuses must be inserted into the F1 and F2 fuse holder of the PIM module (max 6.3A) , and into the 4 fuse holders of the PDU module (max 4A).
The wiring of the power supply must be made with wires of appropriate dimension to ensure that the fuses blow before the wires burns, and the power supply used to power the system must be designed to be able to deliver a current capable of blowing the fuses. More ever a good design includes an external fuse and breaker inserted in the 24V DC system.

4.5.3 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.

4.5.4 Power consumption - stations

Except for the analogue telephones and the SeaCom 220, 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:

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

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

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

4.6 The assembled exchange system

The figure below shows the assembled system before mounting the card fixing rails.
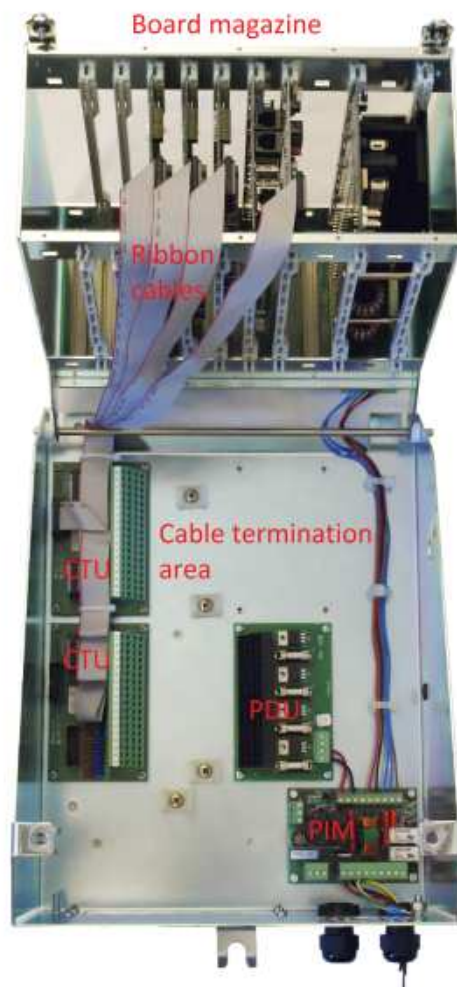
At the lowest right corner of the cable termination area, is the PIM (power input module) which holds the two 24V DC inputs and fuses.

4.7 First time power up

When the exchange is assembled, it is time to power up. Connect 24V DC to the PIM module, and check that the green indicators are lit. Also a green indictor on PSU in the board magazine is on. This means that 24V DC power is present.

The red indicator, shows that one of the powers main or battery is missing. Link the two power inputs to make it turn off. If only single power is used, then link the two inputs, and the red indicator turns off.

Next push the ON/OFF switch down and watch the system powering up.

Indicators of the PSU shows green, and the system boots the WinXPE operating system, and the CP application starts. After this the circuit boards are auto detected and configured. When the Red/yellow indicator of the FIO2 master board starts flashing, the system is up operating, and all extension lines can be tested for having a dial tone.

4.7.1 Burn-in

It is good practice to test all extension lines, and to make a two or tree days burn- in of the system with a renewed test before making the delivery.

On/Off switch

Power indicators

Red/Yellow flash when the sytem is running
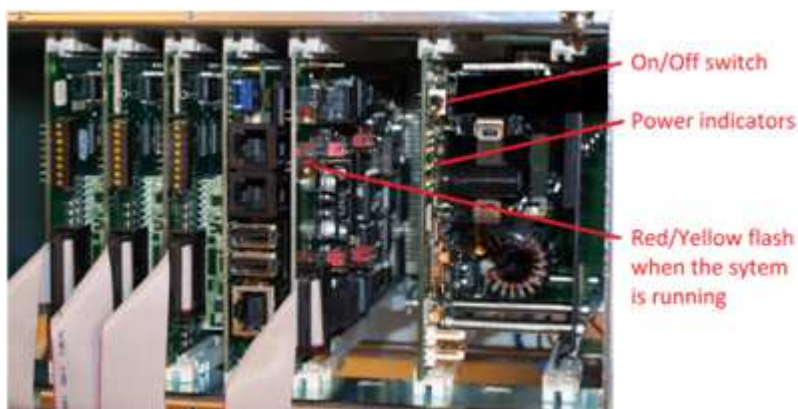
5. Installation

This chapter describes how to install the SeaCom maritime communication system, considerations to be taken, and good practice to follow.

5.1 Pre-installation considerations

The exchange must be installed in a location on board giving the lowest length of cable to be used.

It must be located in a dry ventilated space with a maximum ambient temperature of 55 dgs, the cooler the longer lifetime shall be expected.

5.2 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.

5.3 Wiring diagram

It is the responsibility of the installer to make sure that a full system diagram exist for proper documentation of the full installation. This is most often made in cooperation with the shipyard.

5.4 Mounting the exchange

The exchange must be mounted on a bulkhead in a vertical style to ensure that the intended air flow can be realized.
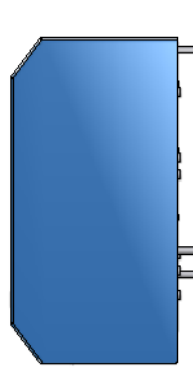
Mounting the exchange – note the 20mm air gap between the exchange and the bulkhead.

Do not mount the exchange horisontal.
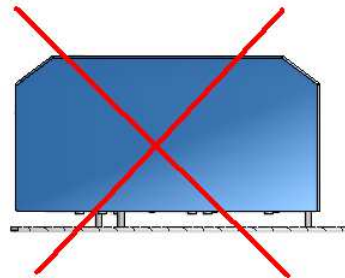
Do not mount the exchange so the air gap behind is closed

5.5 Mounting the stations

Mounting stations are quite straight forward, but if superb hands-free performance shall be obtained with the SeaCom 411 the acoustic design must be taken into account. Refer to chapter 11.3.

5.5.1 Bulkhead

The communication stations SeaCom 211, SeaCom 421 and the SeaCom 220 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

5.5.2 Flush mount

SeaCom 411 is meant for console flush mount.
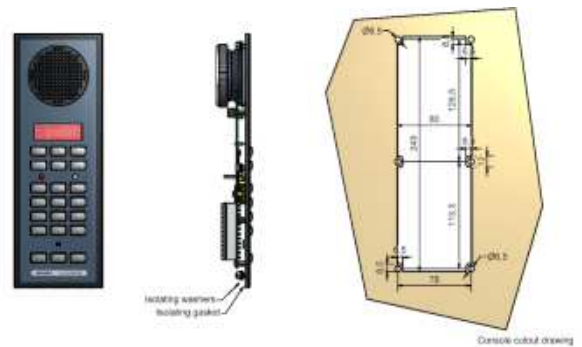
It is delivered with an isolating gasket and isolating washers, so that it is possible to make a galvanic isolated mounting, preventing the stations frame from being connected to the console.
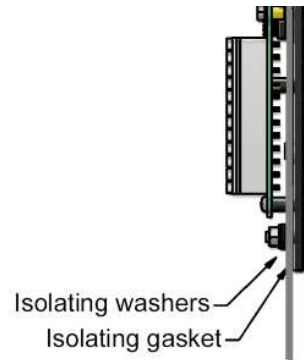
5.6 Power considerations

Power supply – 24V DC – is taken to the exchange unit. Typically an AC/DC converter is used, which is connected to the main power input, and the 24V DC from the shops radio battery is connected to the battery input. Or a power supply with build in switchover can be used (for example the Sailor PS4655)

5.6.1 Galvanic connection to battery

Note that this creates galvanic connection between the power ground of the SeaCom maritime communication system and the battery.

This connection is allowed because the communication and telephone system is isolated entirely from ship's hull.

5.6.2 Fuses

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.

5.6.3 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 SeaCom 421 and SeaCom 411 but it is NOT recommended.

5.7 Shielding and protective earth

When installing the SeaCom 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.

5.7.1 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.

5.7.2 Wiring and junction boxes

The system shall preferably be wired so that all cables are taken from the central exchange unit to the stations in one cable with one shield taken unbroken all the way from exchange to the stations.
If a multi core cable is used out of the exchange, conducting several extension lines to a junction box where it is split into a number of cables to each communication location, this will be allowed. In this case the shields of the multi core cable and the shields of the single core cables must be connected in the junction box, without being connected to anything else.

5.7.3 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, only to shield of cables.
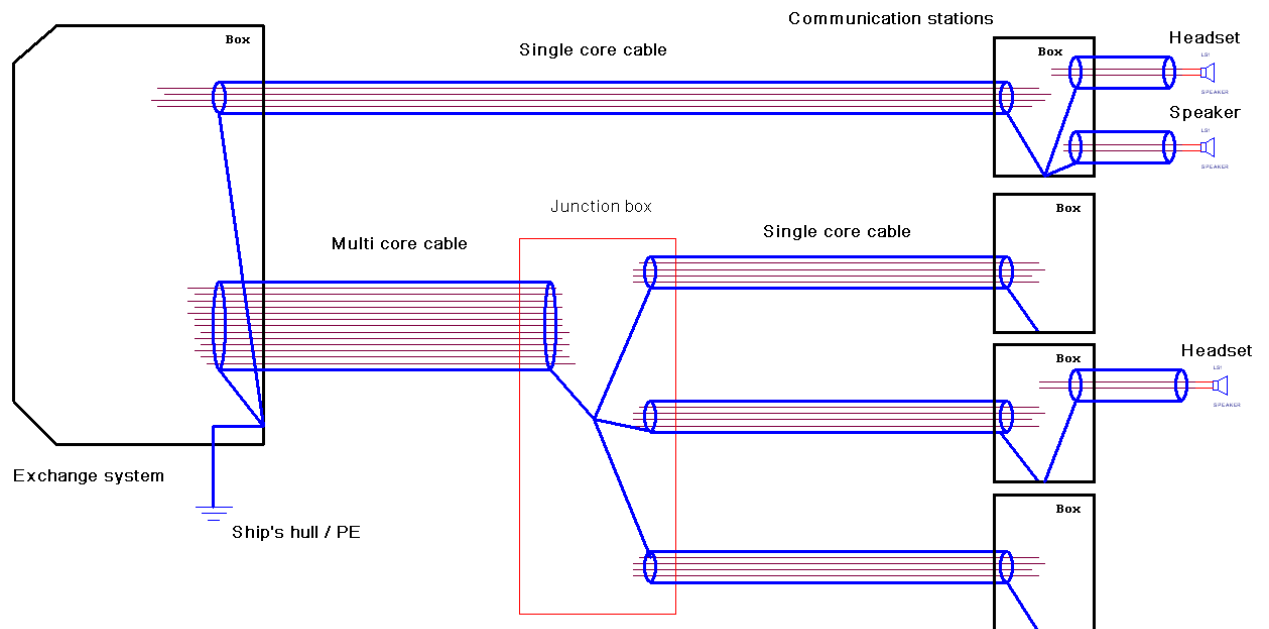
This figure shows schematically the way the shields should be connected.
Note that there is only one point of connection to ship's hull / PE.

6. Commissioning

When the system is properly installed on board, time has come for commissioning the system. If installation and wiring is made by an installer company, it is the responsibility of the person commissioning the system to verify that the installation is done according to the recommendations.

Commissioning includes the below disciplines:

- Powering on for the first time
- Making sure that the system programming, call numbers etc. are correctly set up
- Testing all extensions and communication stations
- Making sure that an end user manual is available.

6.1 First time power on

It is recommended that a step by step power up sequence is followed:

Remove the cover of the exchange and swing up the board magazine.
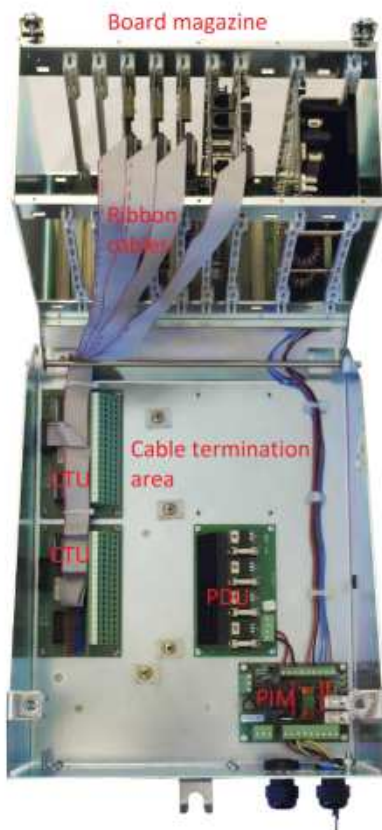
Remove all fuses of the PDU's and remove fuse F1 and F2 of the PIM module.
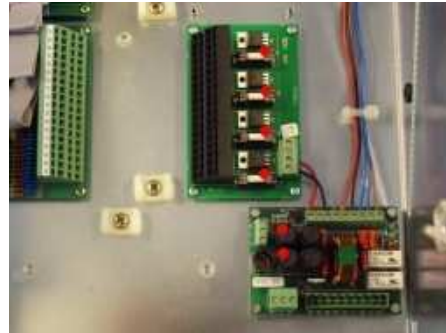
Apply 24V DC and flip the ON/OFF switch down and view the system start up. This takes approximately 2 minutes. After this, dial tones shall be heard in all the analogue telephones connected.

Now it is time to power up the communication stations one by one by successively inserting the fuses F1 and F2 first, and then one by one the fuses of the PDU. Successively verifying that the stations are alive.

6.2 System programming

The SeaCom system is extremely flexible, all depending on the system programming and configuration chosen by the installer. For instructions in how to use the configuration tool software, refer to chapter 12.

There are 4 main disciplines:

- Setting equipment types
- Call number plan
- Access control
- System call numbers

6.3 Equipment type

For each extension line, the equipment hanging on that line must be set. By default all lines are set to basic analogue telephone with no display, but for the communication stations, it is important for the full set of features to be revealed, that the system programming begins with setting the equipment type of each analogue extension line.
If an extension line is connected to a SeaCom 411 station, the line must be set up to be of such type for example the direct in calls to work properly.

6.4 Number plan

The number plan of the installation must be determined. Call numbers of all extensions, all communication stations, all trunk lines, all audio in/out must be programmed as well as the text description of them.

Be aware of number conflicts and double call numbers.

6.5   Access privileges

SeaCom maritime communication system, gives a wide range of possibilities for restricting the use of services to selected extensions only.
For example will it be possible to open the Captains telephone for access to outgoing calls via a satellite service, whereas a crew telephone will only be able to receive calls on the same service.
This is done using service groups or access privilege settings.

Refer to chapter 12.

6.6   System call numbers

The number plan must also include system call numbers. These call numbers are representing functions like wake-up ordering, set-date-and-time, priority call, direct in and so on.

The below list shows the possible system calls to install:

- Priority
- Direct in calls
- Set-date-and-time
- Wake-up
- Ringing groups
- Short numbers
- PA calls
- Call pickup
- Conference calls (talk-back)
- Do not disturb

6.7   Testing

When all stations are powered up, it is time for testing. Testing requires two persons, one in the exchange room, and the other visiting the locations with telephones armed with a VHF radio. All positions having either an analogue telephone or a communication station must be visited.

A good tool is to connect a test telephone in the exchange room, and open the VNCviewer with which it is possible to check the call number of the last active extension.

6.7.1   Extensions

Telephones are checked for having dial tone, and for having the desired call number and name. A call must be set up from each telephone to the test telephone in the exchange room, and vice versa.

6.7.2   Stations

Commissioning the communication stations is a little more complex, as the stations all have settings to take care of.
Refer to the chapters describing each stations for details.

The below paragraphs describes what as a minimum have to be checked.

6.7.2.1   *SeaCom 211 the talk-back stations*

The SeaCom 211 station have gains to be set for adjusting gains of microphones and speakers. These must be adjusted properly.

And the 3 push buttons must be programmed to address the right call number, typically bridge or ECR

6.7.2.2   *SeaCom 220 industrial telephone*

The SeaCom 220 will normally operate with the factory settings only, but gains can also be adjusted for this telephone also.

6.7.2.3   *SeaCom 421 and SeaCom 411*

These stations can be installed in hands free only, or with handset and/or headset. All this equipment must be checked and especially the microphone gains be checked.

If the stations are operated in hands free, then the hands free gain margin setting must be checked, to make sure that the margin is set as low as possible without acoustical feedback. Especially if external speaker is used, this parameter is important, and factory setting cannot be relied on, as the acoustical return signal cannot be foreseen.

6.7.3   Trunk lines

Systems with trunk lines connected to satellite equipment, shall have these connections tested during commissioning. Always test both outgoing and incoming calls. It must be verified that incoming calls are really handled and that the routing is really as desired.

6.7.4   I/O

If music in is used or if an audio out channel is used, these shall be tested.

6.7.5   Testing system call numbers

All system call numbers must be checked.

6.7.6 Access privileges

When testing extensions, it is necessary to verify that access to or blocking of access to features like satellite services are really installed.

6.8 End user manual

It is the responsibility of the company commissioning the system to create an end user manual. This manual is to be used by the crew members using the SeaCom system.

Take chapter 7 as a starting point for this manual.

SeaCom System Manual which You are reading now is not meant as a manual for crew.

7. Operation

This chapter s 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.

7.1 Making and transferring a call

Making calls are made as with any telephone system. Lift the handset of a telephone and dial the number 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.

7.2 Ringing signals

The SeaCom 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)

___ ___ ___   ___ ___ ___

7.3 Tone signals

The SeaCom 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 dialled
(intermittent tone)

___   ___   ___   ___

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

__   __   __

7.4 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 dialling 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.

7.5 Direct in

The Direct-in feature can be used with SeaCom 211, SeaCom 421 and SeaCom 411 only.
It is made by dialling 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 SeaCom 411 and SeaCom 421 enters hands-free mode and conversations or listening is effective.
Direct in to the SeaCom 211station is to be considered a single station PA call, as it is only possible to speak from the caller to the SeaCom 211.

7.6 Direct in to ringing group

A special feature is direct in calls to a ringing group.

When a bridge is equipped with more than one station with the direct-in call possibility (typically SeaCom 411's), 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 dialling **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 dialling the bridge group call number.

7.7   Setting system time

The system time of the SeaCom 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.

7.8   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.

7.9   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.

7.10   Using the call data manager

On systems where the CDM system is installed. (Call Data Manager), dialling out via

satellite services are made by the following dial sequence:

Dial: #10   to access the CDM service

Dial: AA   Two digits account number

Dial: PPPP   Four digits pin code

Dial: 004586296297#

Wait for answer and speak.

When the conversation is ended, a call data record is stored on the system disk of the CP board.

7.11   Paging calls

Paging calls can be received by the communication stations with speakers - SeaCom 211, SeaCom 421 and SeaCom 411 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.

7.12   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.

7.13   Alarm calls

Alarm calls can be received by the communication stations with speakers - SeaCom 211, SeaCom 421 and SeaCom 411 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.

7.14 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.

7.15 Music when free

If an audio in channel is configured, the SeaCom 411 and SeaCom 421 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.

7.16 Talk-back command calls

This is the classical "talk-back" call.
The call number #07 is used to make a talk-back command call:

 If the call is made from a SeaCom 411 to one or more SeaCom 211's, the speech direction of the call can be controlled only by the microphone key of the SeaCom 411 initiating the call.

The command call can be set up as a fixed group or as a dynamic group, where the first has the SeaCom 211's 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 SeaCom 211's 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 SeaCom 211 not being the primary slaves will just listen to the conversation, and can hear both the master and the primary slave.

7.17 Reverse talk-back

A Semiduplex-conference call can be initiated from one of the SeaCom 211 substations. In such case a conference master of type SeaCom 411 must be defined in the system programming. When the SeaCom 211 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.

7.18 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).

7.19 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.

7.20　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.

7.21　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.

8. Maintenance

When the system is properly installed there is no daily maintenance, but some components are worn out and need regular maintenance or replacement.

Refer to the list of parts chapter 1.

8.1 Headsets

The part most often worn out is the headsets of communication stations located in workshops and other areas with dirty environment.

In order to make an easy replacement, it is recommended always to use the headset with water tight connector set.

8.2 Batteries of analogue telephones

If the TX-325 telephones are used, the hands-free function and the display function depends on batteries.
These batteries typically last 6 months, less when frequently used.

Check and replace batteries of all telephone every 6 months.
In locations where this is not practically to be done, use an analogue telephone without display and hands-free.

9. SeaCom 2100

The SeaCom 2100 is our largest boxed and type approved maritime communication system

9.2 Exploded view

Below figure shows the interior.

9.1 Specification

- 8 to 256 analogue extensions
- 2 to 10 2 wire trunk lines
- 18 slots
- 16 CTU / PDU positions
- 24V DC 20-100W
- N.C. alarm relay output

9.3 Mechanical outline
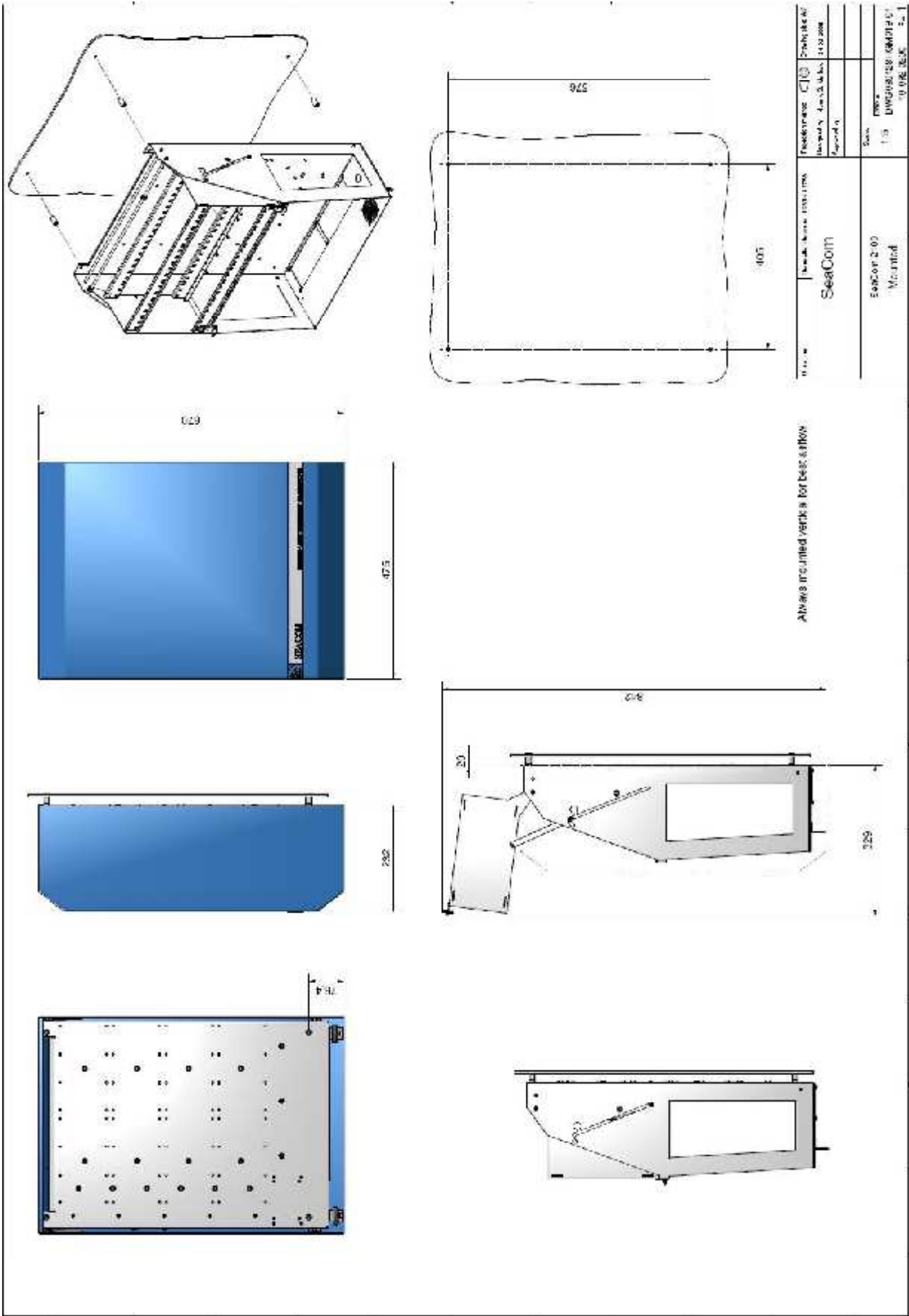
9.4　The system inside

This chapter describes common parts that are inside the systems.

9.4.1　Board magazine

The top of the systems holds the board magazine, carrying the circuit boards of 100x220mm wired in a bus system by the backplane.

This board magazine is exactly like the SeaCom 19'' system.

The board magazine is mounted in a swing up frame to compact the system when in operation, and facilitate access to the cable termination are when the system is under installation.

The board magazine carries circuit boards:

- Backplane
- PSU (power supply unit)
- CP (Central processor)
- FIO2 Master (2 x Trunk line and master clock generator)
- AEXT8 (8 x Extension lines)
- FIO2 Slaves (2 x Trunk line)

These boards are described in details chapter 10.

9.4.2　Cable termination area

The backside of the cabinet is the cable termination area. In this area the below boards can be mounted:

- CTU (cable termination unit)
- PDU (power distribution unit)

These modules connects the ship cabling in a push and insert manner, and the CTU holds the EMC protection components.

For details refer to chapter 10.

9.4.3　Power input module - PIM

The power input module is used for connecting the 24V DC power, and for providing the alarm relay contacts, and for providing an optional fan 24V DC drive. Included on the PIM module are power indicators and fuses.

Although described in chapter 10, a short description is found here.
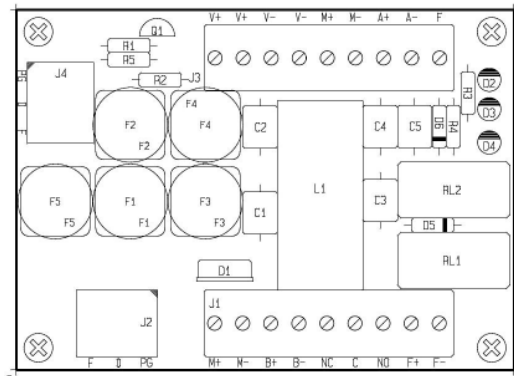
9.4.4　J1 power input

The use of J1 is

Terminal	Use
M+ and M-	24V DC main power
B+ and B-	24V DC battery backup
NC, C, NO	Contacts of alarm relay
F+, F-	Fan drive output 24V 1A

9.4.5　J3 power output

The use of J3 is

Terminal	Use
V+ and V-	Auxiliary 24V DC output fused by F1, F2

9.4.6　Fuses

The fuses are

Terminal	Use
F1, F2	Fuses for Auxiliary 24V DC output
F3, F4	Fuses for power to board magazine

9.4.7　Indicators

The PIM has 3 indicators:

Terminal	Use
D2	24V to board magazine OK green – on when power OK
D3	Auxiliary power out OK green – on when power OK
D4	System alarm active red – on when alarm or error

The red alarm indicator is on if one of the power supplies is missing. If single supply is use, then link the two inputs.

10. Circuit boards.

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

10.1 PSU power supply unit

The PSU is found in the rightmost position of the board magazine.

Its purpose is to convert 24V DC into the internal used system voltages +5V , -5V, -48V DC and 80V AC 50 Hz.

All inputs and outputs are carried through the backplane connector.

Beside this, the power supply holds the ON/OFF switch, and it has the overall system watchdog which is kicked by commands from the CP.

10.1.1 Technical data

- Dual input with switchover
- Input voltage 18 to 32V DC
- Output 5VD 5A
- Output 5VA 750mA
- Output -5VA 750mA
- Output -48V 4A
- Output 80V AC 50Hz 250mA
- N.O. alarm relay contact
- Watchdog
- Temperature supervision

10.1.2 ON/OFF switch

The ON/OFF has 3 positions:

Mid position is OFF

Down position is ON

Up is used for fast turn off.

Switching from OFF to ON starts a power up sequence where the power supply turns powers on and tests if these are OK.
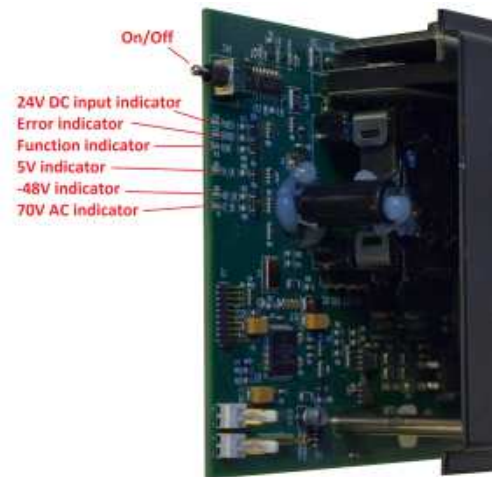
Switching from ON to OFF starts the power down sequence, which takes 10 seconds.

When in ON and the Up position is used for a short time, the power supply will skip the power down sequence and turn of immediately.

10.1.3 Indicators

The PSU has 6 indicators:

Indicator	Use
Input power	Green - On when input is OK
Function	Yellow - Flashing when power on or power of process is running. Occulting when on battery operation
Error indicator	Red - On when fatal error Flashing when temperature high
5V indicator	Green - On when 5V is OK
-48V indicator	Green - On when -48V is OK
70V AC	Green - On when 70V AC is turned on (Only the case when any telephone is active)

10.1.4 Alarm relay

The power supply has a N.C. alarm relay contact. This relay is driving the alarm relay of the PIM module, so it is normally not accessed directly by the installer.

The alarm relay is normally closed, but will open on temperature errors or power supply internal errors.

10.1.5 Fan drive

An open collector output is used to drive a fan relay located on the PIM module. The fan relay is activated based on temperature measurements.

10.1.6 Input power supervision

If the source supply voltage is below 18V the power supply will switch off and stay as such until the input voltage exceeds 20V. During power low, the power indicator will flash.

10.1.7 Temperature protection

The power supply has a temperature protection mechanism. The scheme below shows its actions:

Temperature	Action
< 35°C	Fan off Alarm relay closed
> 45°C	Fan on Alarm relay closed
> 70°C	Error indicator on Alarm relay open
> 85°C	Shut down Alarm relay open

Note that when the alarm is set, the alarm cannot be reset unless the system is turned off and back on. The on/off switch will not do this.

10.1.8 Layout

10.2 CP2 central processor
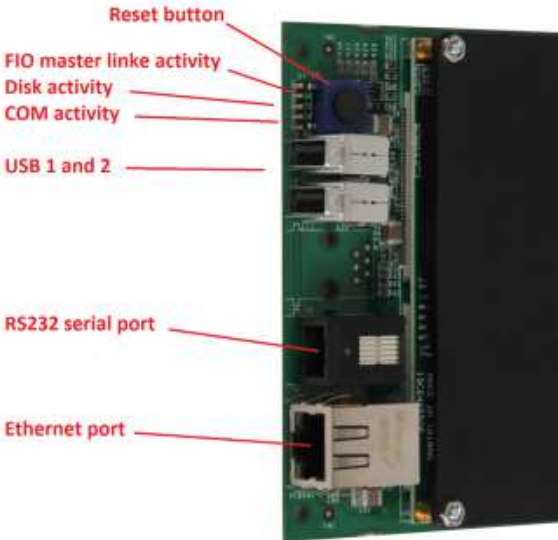
The purpose of the CP2 board is to execute the CP software which is the very brain of the exchange system, and to hold the configuration file which stores call numbers, privilege settings etc.

The CP2 board also holds the data base used for storing call data when the Call Data Manager is installed.

The CP2 board is located anywhere in the board magazine.

The CP2 runs a Windows XP embedded operating system, which is stored in a write protected part of the compact flash card.

10.2.1 Technical data

- Geode LX800 processor
- 512Mb RAM
- Disk 2Gb Compact flash
- 10/100 Mbit Ethernet
- 2xUSB Serial port
- 1xRS232 serial port
- Windows Embedded Standard
- Power 5W
- DHCP server (20-131_0100 or later)

10.2.2 Serial port

J4 is the RS232 serial port used for receiving NMEA signals. The port is a 6 pole modular receptacle and has the below pin-out.

Number	Use
1 (closest to SW1)	TX (output)
2	RX (Input)
3	GND
4	GND
5	Leave open
6	Leave open

NOTE: The serial port does not provide galvanic isolation. This is to be handled externally if needed.

10.2.3 Advantec SOM 4455

The primary part of the board is the Advantec SOM4455 ETX standard industrial computer board, which is covered by the black heat sink.
This board, have a flash disk connector where the compact flash card holding the operating system and data is mounted.

10.2.4 Indicators

The CP2 board has 5 indicators.

Indicator	Use
Disk activity	Green - Flashing when disk access is going on
Com	Green and red - for receive and red for transmit Flashing when communicating
FIO2 link	Green and red - flashing when link is operating

10.2.5 Reset button

The blue button is for reset. This can only be used for software development purposes.

10.2.6 Layout
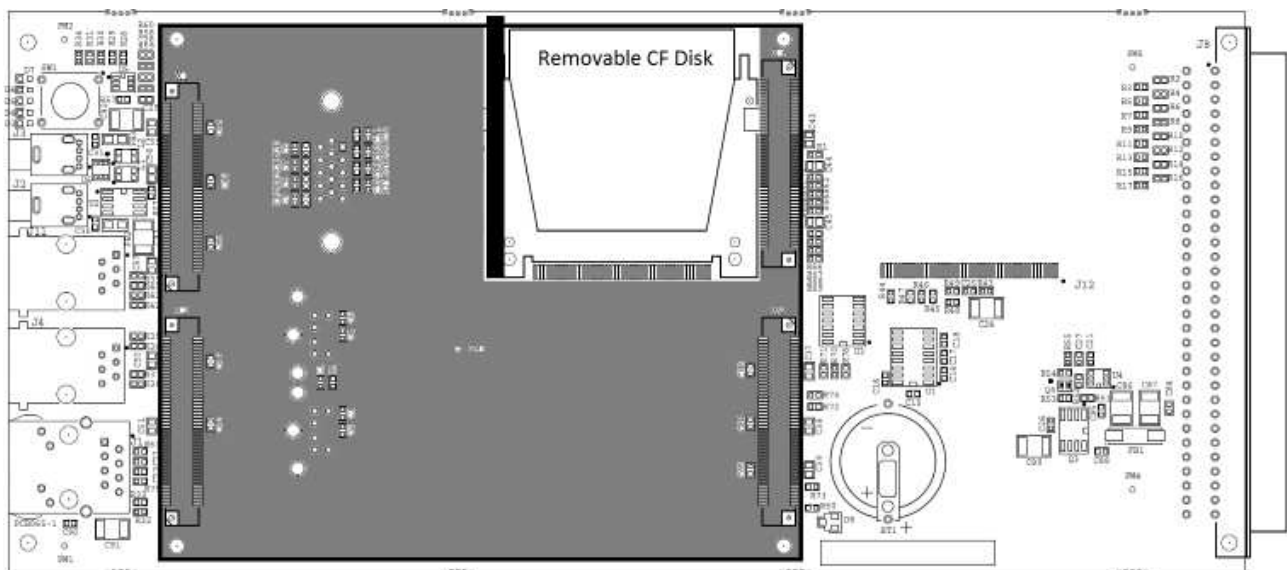
10.3 FIO2 master and slave

The FIO2 board (flexible I/O) holds two channels which can either be configured as 2 wire analogue trunks or as general purpose audio in/out channels.

The FIO2 board comes in two versions, the master and the slave.

One and only one FIO2 master board must be found in all exchange systems, so when more than two trunk lines or audio in/outs are needed, additional channels are added using the FIO2 slave board.

10.3.1 Features of the master

The difference between the two board types is that the FIO2 master holds a master clock generator and the HDLC master communication function. Both these are needed for all the circuit boards to communicate via the backplane and to pass voice as PCM samples between boards engaged in conversation.
The FIO2 master also creates the dial tone used by all cards in a system.
The FIO2 master board is connected to the CP board via a bidirectional serial communication link. Via this link, the CP is commanding the behaviour of the exchange

system. Activity on the link is indicated by red and green indicators

This figure points out the extra circuits of the master board.

The flashing of the link indicators is used for debugging purposes.

10.3.2 Technical data

- 2 channels
- trunk and audio in/out galvanic isolated (1.0kV)
- 600 ohm trunk line
- 600 ohm audio in/out
- Line voltage and polarity detector
- Line loop current and polarity detector
- Ringing signal detector
- DTMF receiver
- DTMF transmitter
- Call Progress Tone Detector
- +6dB to -9dB gain adjustment receive and transmit
- Relay 24V DC 1A
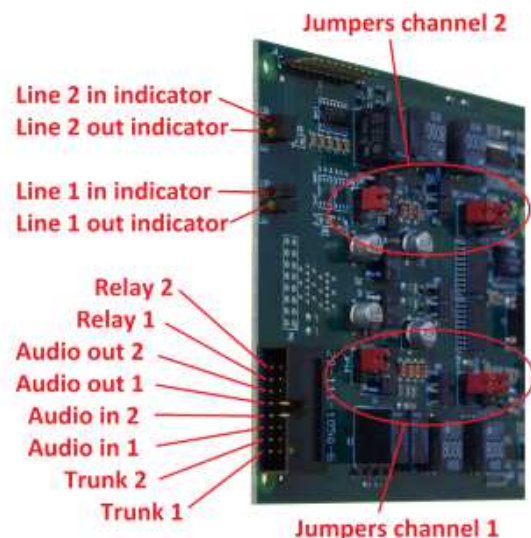
10.3.3 Connector

Trunk lines and audio are connected via the 16 pin front connector. All connections are in pairs of two pins placed horizontal to each other. The below table shows the connector layout, where the lowest pin is number 1 and topmost number 8.

Number	Use
8	Relay contacts 2
7	Relay contacts 1
6	Audio out 2
5	Audio out 1
4	Audio in 2
3	Audio in 1
2	Trunk line 2
1	Trunk line 1

10.3.4   Line activity indicators

The FIO2 board has 4 line activity indicators, two for each channel

Indicator	Use
Outgoing call indicator	Yellow - off: line is idle - flash 1 Hz: wait for B answer - flash 4 Hz: dialing - on: conversation out
Incoming call indicator	Red - off: line is idle - flash 1 Hz: wait for B answer - flash 4 Hz: dialing - on: conversation in

When both the red indicator and the yellow indicator of one of the lines flash simultaneously, the line is blocked due to missing detection of line feed.

10.3.5   Selecting trunk or audio I/O

Each channel has 3 jumpers, which are used to select the hardware function of a line to either the 2 wire analogue trunk function or the audio in/out function.

When choosing a hardware configuration using the jumpers, it is important to choose the right equipment type for that channel in the system configuration programming. Refer to chapter 12.

The use of the jumpers are explained below:
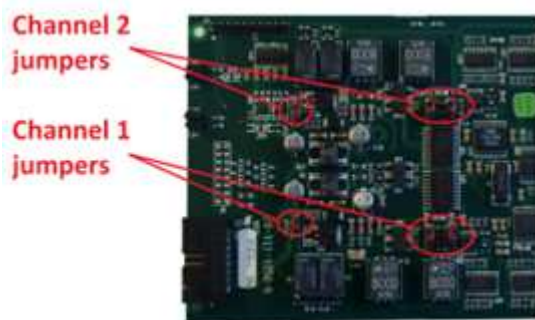
Analogue trunk line:
Set all 3 jumpers of the channel in position 1+2 (upwards as shown on picture). This is the default setting.

Analogue in/out:
Set all 3 jumpers of channel in position 2+3 (downwards).

10.3.6   Trunk line circuits

The 2 wire analogue trunk line is used for interfacing to most types of satellite equipment, which has a 2 wire PSTN output.

The features of the trunk line circuits are:

- Galvanic isolation 1kV
- 600 ohm line impedance
- AC ringing detection when ringing > 45V AC 50Hz
- Line feed detection when line voltage > 24V DC
- Polarity reversal detection

10.3.7   Audio I/O circuits

The audio I/O is a general purpose interface which is used to interface PA amplifiers and input music from CD players or input voice from a VHF radio.

The features are:

- Galvanic isolation 1kV
- 600 ohm transformer coupled in and out
- Relay 24V DC 1A

10.3.8 Layout
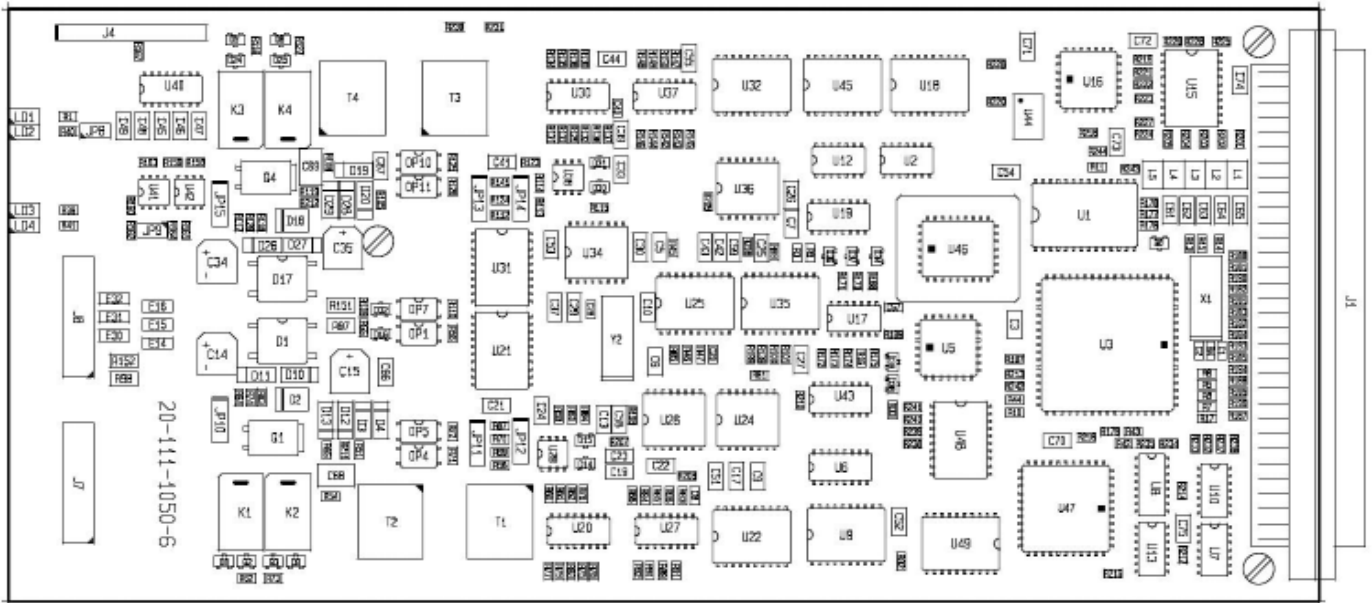
10.4 AEXT8

The AEXT8 is an 8 line analogue extension board. It is used for connecting analogue telephone sets as well as connecting communication stations.

The extension board is responsible for creating the gong sound when an extension is making a PA call and it is responsible for generating the alarm tones when an alarm call is made.

The board has two DTMF receivers shared among the 8 lines when dialling, and it has one 16 kHz signalling generator shared among lines for sending the 16 kHz communication to stations.

10.4.1 Technical data

- 8 channels
- 48V DC line feed
- 25mA DC linefeed when hook off
- DTMF receiver
- Loop disconnect dialling
- R button detect
- Drives 3 analogue telephones in parallel when ringing
- Per line activity indicators
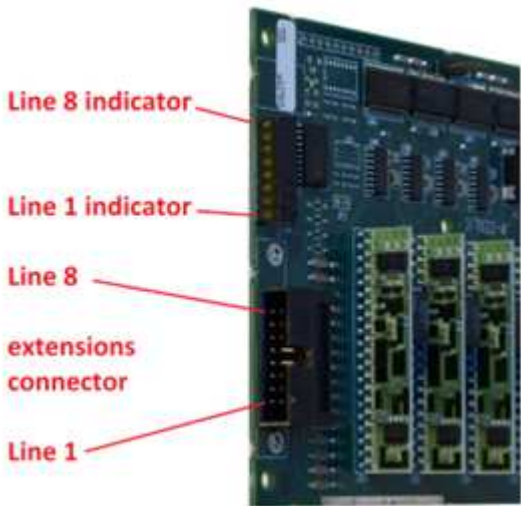
10.4.2 Front connector

Extensions are connected via the 16 pin front connector. All connections are in pairs of two pins (tip and ring) placed horizontal to each other.

The below table shows the connector layout, where the lowest pin is number 1 and topmost number 8.

Number	Use
8	Line 8
7	Line 7
6	Line 6
5	Line 5
4	Line 4
3	Line 3
2	Line 2
1	Line 1

10.4.3 Line activity indicators

The AEXT8 board has 8 line activity indicators, one for each channel

Indicator	Use
Activity	Yellow - off: line is idle - flash 1 Hz: wait for B answer or line terminated off hook - flash 4 Hz: dialing - flash 4 Hz with intervals 5s: Extension is ringing - on: conversation out

10.4.4 Layout

AEXTB 20-111-1041-6
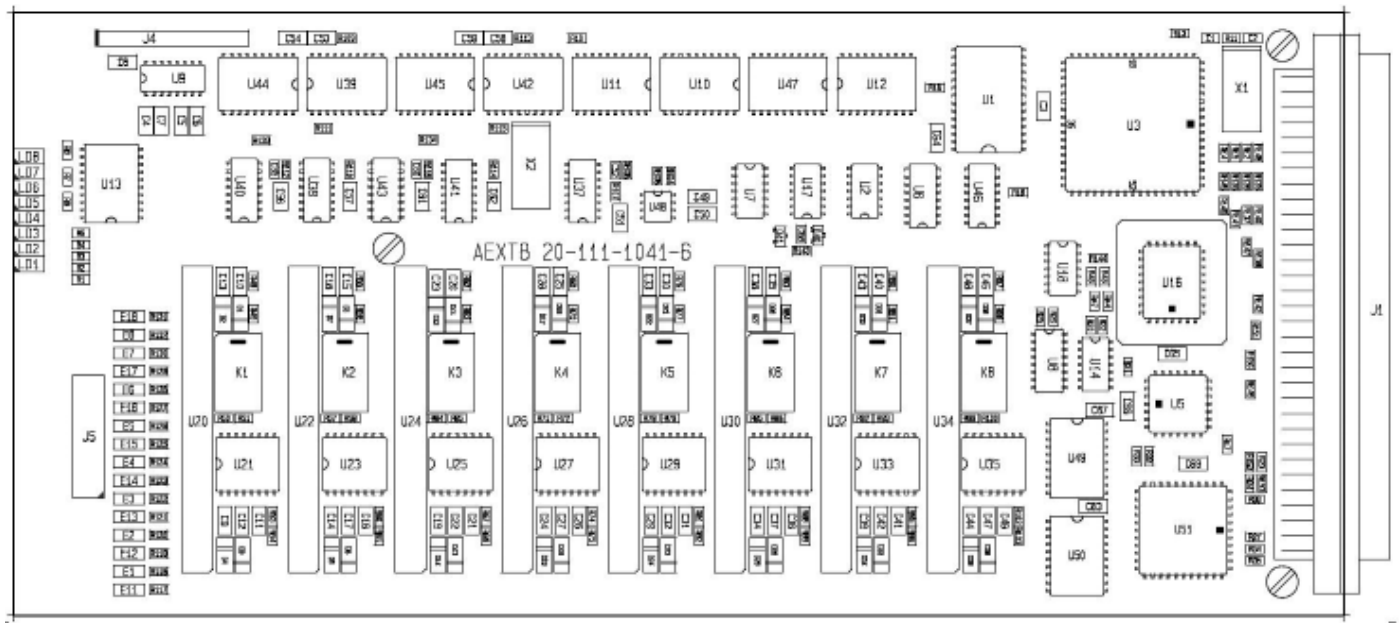
10.5 The AEXT16

The picture below shows the **A**nalog **Ext**ension card AEXT16:
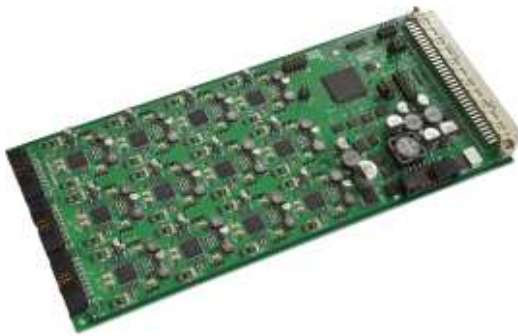
The AEXT16 is the telephone line card of the system. The AEXT that comes with the SeaCom 2100 has 8 lines, but the AEXT16 line card comes in 8, 16 or 24 lines versions as well, in order to more closely adapt the need of the customer.

1,2 or 3 lines connectors are found. These have the 2 wire analogue telephone lines.

The 64 pole DIN connector is used to connect the card to the backplane. This connector carries 24V DC power, PCM data streams, reset signals and inter board communication.

10.5.1 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

10.5.2 Jumpers

This paragraph is of importance only if You plan to use the AEXT16 as system master card.

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

One, and only one, board of a SeaCom 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	0-5V level Used with LSP
5,6 7,8	RS232 level Used with CP2

10.5.3 Board layout

The board layout is shown below:
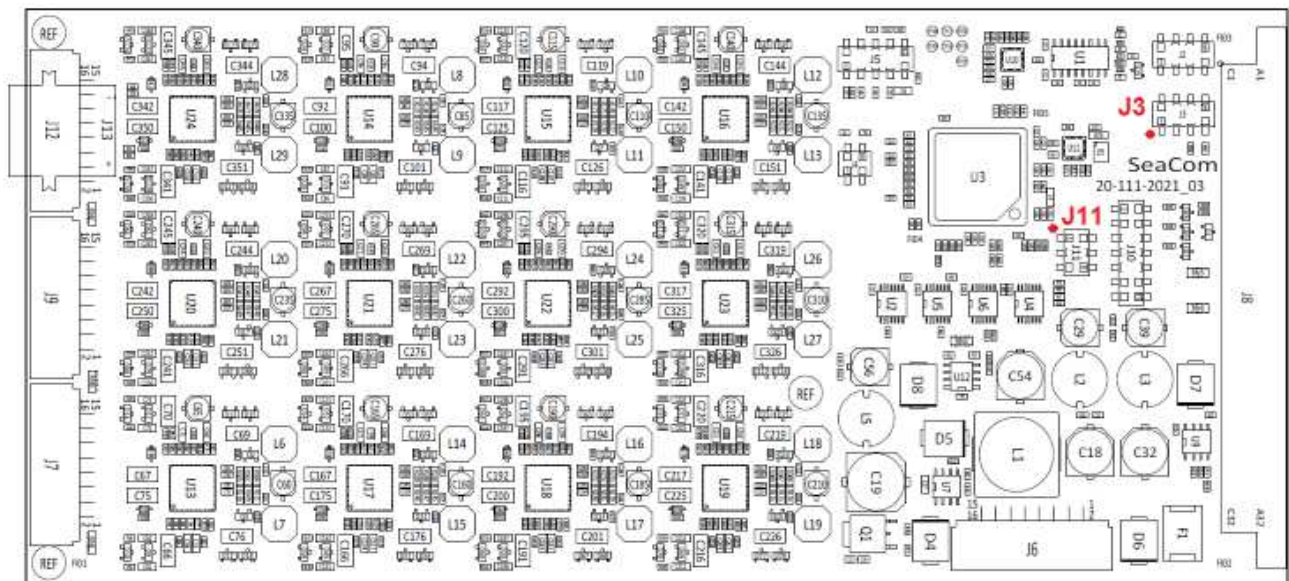
10.6 The FIO4

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

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

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.

10.6.1 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.

10.6.2 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

10.6.3 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

10.6.4 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

10.6.5 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

10.6.6 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

10.6.7 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.

10.6.8 Jumpers

This paragraph is of importance only if You plan to use the FIO4 as system master card.

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	0-5V level Used with LSP
5,6 7,8	RS232 level Used with CP2

10.6.9 Board layout

The board layout is shown below:
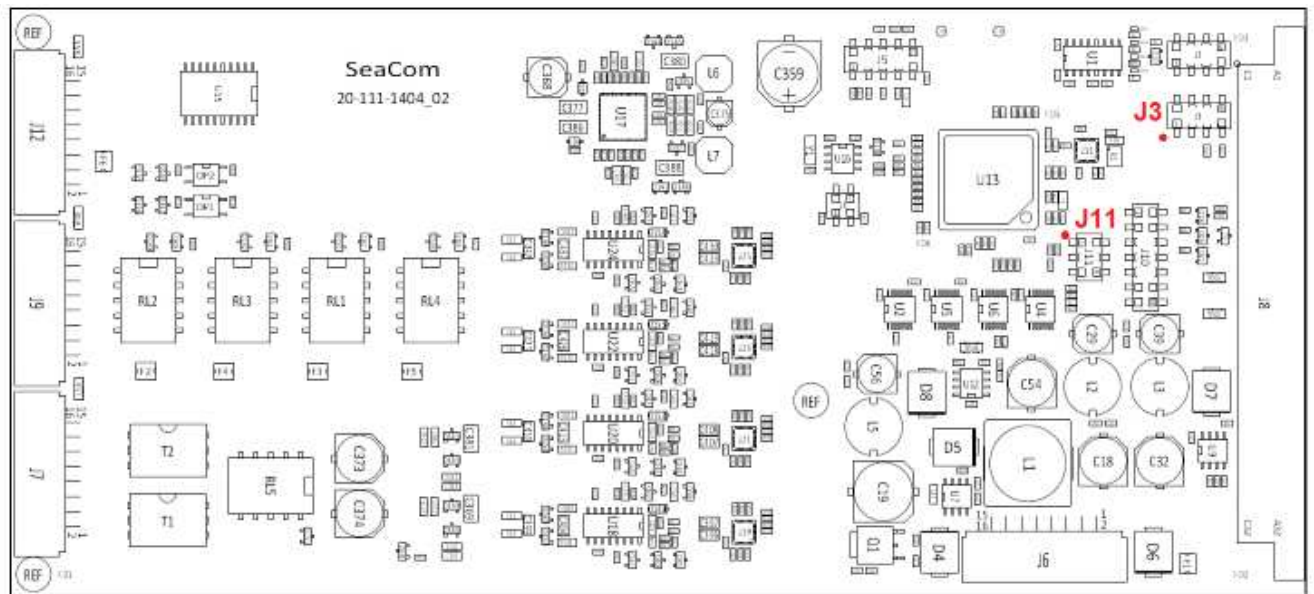
10.7 CTU2

The CTU2 (cable termination unit) is used for making the connection between the robust cables used on ships, and the tiny connectors used on circuit boards.
The CTU2 also makes up the EMC barrier by having T- filters and a 2kA transient absorber.

The CTU2 is mounted in the cable termination area at the back of the exchange systems using 4 pcs 3x6mm screws.
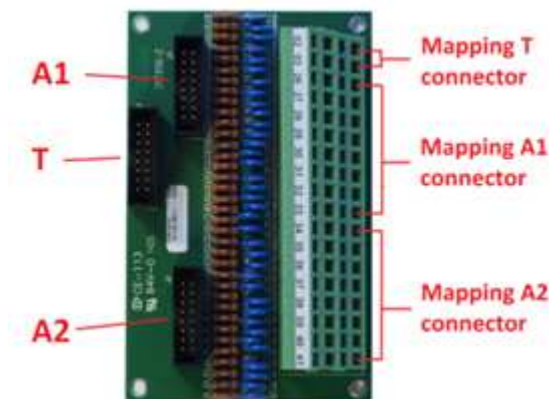
10.7.1 Technical data

- Terminates 18 pairs
- Max 1.5mm^2
- 16 extensions and 2 trunk lines

10.7.2 Using the CTU2

The SeaCom system makes use of prefabricated 16 wire ribbon cables between boards of the board magazine and the termination units. These cables comes in a selection of lengths to suit the needs.

The CTU2 has two 16 pin connectors, giving the possibility of terminating 2 AEXT8 boards with 8 telephone lines.
The one AEXT8 board will be connected to the A1 connector, and the other to the A2 connector. The CTU2 maps the connections via the T-filter and transient absorbers to the green or black ship cable connector in a 1:1 fashion.

As a supplement to the two A connectors, the CTU2 has a T connector which is also a 16 pin connector, but out of which only the topmost 4 pins are in use. This connector is used for terminating a FIO2 board which is entirely used for analogue trunk lines. The two trunk lines will be mapped into the topmost two pairs on the ship cable connector.

If one of the two channels of a FIO2 board is used for something else than 2 wire trunk, then the FIO2 board has to be connected to A1 or A2 in order map all the pins from the FIO2 board onto the ship cable side of the CTU2.

10.7.3 Mapping

Circuit boards mounted in the board magazine will have channel 1 as the lowest pin on the front connector. This applies for both AEXT8 and FIO2. The lowest pin will be mapped into the topmost pin on the CTU2 ship cable terminals.

As seen on the picture above, the trunk lines and the extension numbers are numbered from the top.

10.7.4 Layout
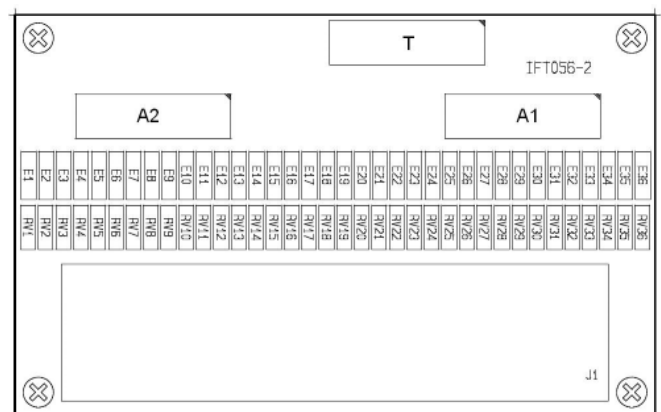
10.8 CTU24

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.

10.8.1 Technical data

- Terminates 24 pairs
- Max 1.5mm^2
- 16 extensions and 2 trunk lines

10.8.2 Using the CTU24

The SeaCom system makes use of prefabricated 16 wire ribbon cables between boards of the board magazine and the termination units. These cables comes in a selection of lengths to suit the needs.

The CTU24 has 3 16 pin connectors, giving the possibility of terminating 3 AEXT8 boards with 8 telephone lines or one AEXT16-24 with 24 lines.

10.8.3 Mapping

The CTU24 mapping is easy 1:1

10.8.4 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
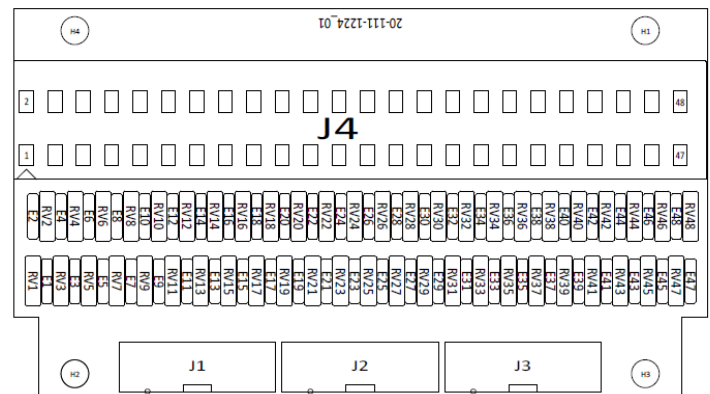
10.9　PDU

The PDU (power distribution unit) has the purpose of distributing 24V DC power to stations that need power, and provide the proper fusing needed.

The PDU is mounted in the cable termination area at the back of the exchange systems using 4 pcs 3x6mm screws.

10.9.1　Technical data

- 36 outputs in 4 fused groups
- 4A max per group
- 1A max per output
- 24V DC operation (max 40V)
- 4 x 4A fuses
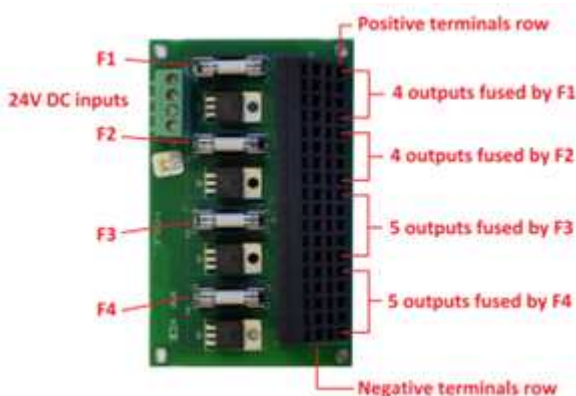
10.9.2　Using the PDU

T24V DC is applied to the green power input connector. This power is distributed via the 4 fuses to the black ship cable connector, where 36 sets of positive and negative power outputs are found.

The row of terminals closest to the card edge is the positive outputs and the row on the top of the ship cable connector is the negative outputs. Note that all negative inputs and outputs are all tied together.

There are 4 fused groups, F1 .. F4.
F1 covers the topmost 4 outputs, F2 the next 4 outputs, whereas F3 and F4 each covers the lowest two groups of 5 outputs.

10.9.3　Power design

Designing the power supply will depend very much on the actual system to be installed. Stations use less than 150mA in stand by, but can use as much as 1.5A when active with a PA call at full volume.

Power to the stations can be taken from the auxiliary DC output of the PIM module. This output has a maximum of 6.3A, which makes feed of 40 stations possible in stand by, but only 5 stations can be feed when engaged in PA calls at full volume.

As a rule of thumb, 10 stations can be feed via the auxiliary DC out of the PIM, when care is taken not to operate at full volume for all stations in PA.

When the power output capabilities of the PIM module is not adequate, the PIM modules can be powered directly from the 24V supply, but in this case external fusing must be used.

IT IS THE RESPONSIBILITY OF THE INSTALLER TO MAKE PROPER FUSING !

10.9.4 Layout
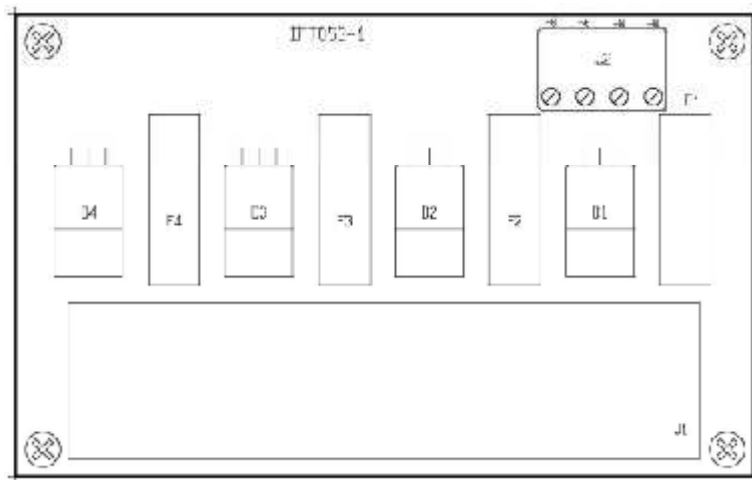
10.10   PIM

The purpose of the PIM (power input module) is to make a common power entry point for main DC supply and battery DC supply for the exchange system as well as for small number of stations.

The PIM includes proper fusing, and there is a relay output for connecting to alarm systems.

There is a 24V DC drive for a fan, controlled by the temperature supervision feature of the PSU.

And the PIM has a single fuse to be used when an AC/DC converter is mounted inside the exchange system.

10.10.1   Technical data

- 24V DC main input
- 24V DC battery input
- Board magazine 24V DC output
  Fused max 2A
- Auxiliary 24V DC output
  Fused max 6.3A
- N.O. and N.C. alarm relay output 24V DC 1A
- Power indicators
- Error indicator
- Fan drive output
  24V DC 1A
- Optional AC fuse

10.10.2   J1 power input

The use of J1 is

Pin	Use
M+ and M-	24V DC main power
B+ and B-	24V DC battery backup
NC, C, NO	Contacts of alarm relay
F+, F-	Fan drive output 24V 1A

10.10.3   J3 power output

The use of J3 is

Pin	Use
V+ and V-	Auxiliary 24V DC output fused by F1, F2
Outputs to board magazine	All connections to the board magazine, including 24V DC power

10.10.4   Fuses

The use of the fuses are

Fuse	Use
F1, F2	Fuses for Auxiliary 24V DC output
F3, F4	Fuses for power to board magazine
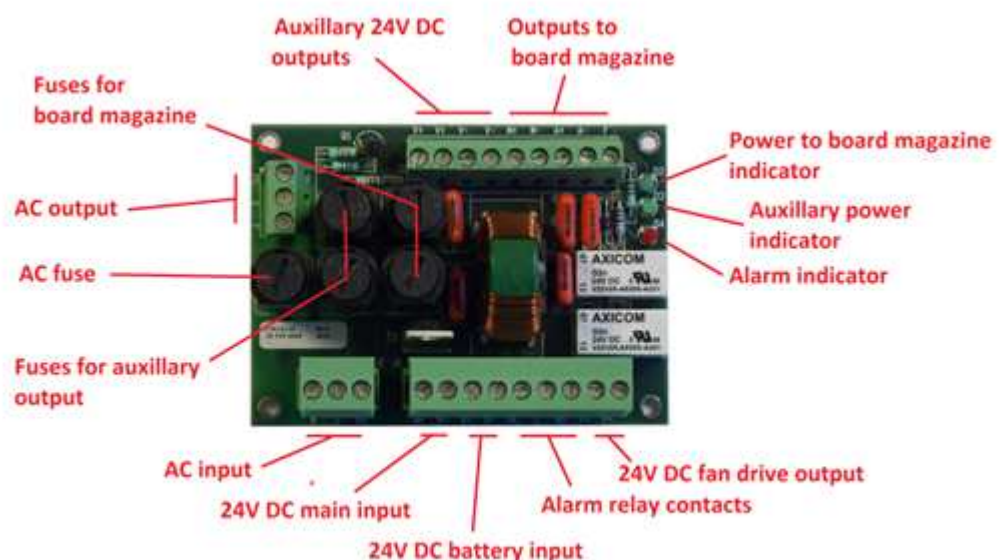
10.10.5 Indicators

The PIM has 3 indicators:

Terminal	Use
D2	24V to board magazine OK green – on when power OK
D3	Auxiliary power out OK green – on when power OK
D4	System alarm active red – on when alarm or error

The red alarm indicator is on if one of the power supplies is missing. If only single supply is used, then connect the two inputs in parallel.

10.10.6 Alarm relay

The alarm relay is activated when one of the following conditions are set:

- One of the power supplies is missing (connect power inputs in parallel for single supply)
- The PSU detects over temperature
- The PSU detects error on supply voltages

10.10.7 Layout
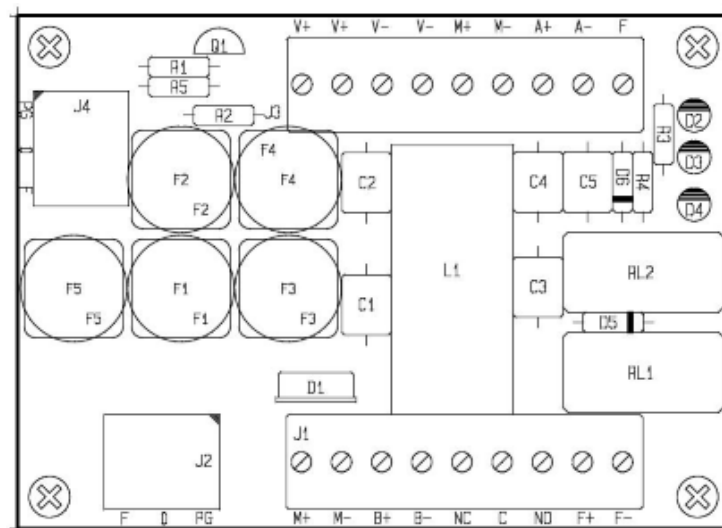
10.10.8 AC connectors

As an option, the PIM module has an AC in (J2) and an AC out (J4) connector with a fuse in between (F5). These are used if an AC/DC converter shall be mounted inside the exchange system.

When using this feature, the system contains hazardous voltages inside.
It is the responsibility of the installer clearly to indicate this by adhering a label on the outside the enclosure.

Warning !
High voltage inside

10.11 Backplane

The backplane connects all circuit board of the board magazine to form an exchange system.

It transfers power from the PSU to all cards, includes a common reset signal, and holds the serial telecom busses used for passing voice and data between the boards of the exchange.

This backplane is used for the SeaCom 2100.

10.11.1 Technical data

- Dual 24V DC power input
- Alarm relay N.C. relay contact 24V DC 1A
- Fan drive output
- 18 board slots + PSU slot
- Sum 24V output 1A

10.11.2 Using the backplane

The backplane is an integrated part of the SeaCom 2100 and the SeaCom 19'' systems. The black ribbon cable connectors are used when paralleling 19'' systems for building large exchange systems (up till 800 lines).

10.11.3 J22 – the power connector

The use of J22 is

Pin	Use
Battery+ and-	24V DC battery backup
Supply+ and -	24V DC main power
Fan	Open collector fan drive Max 40V, 100mA
Relay	Contacts of alarm relay
+24V	Sum output of 24V DC 1A
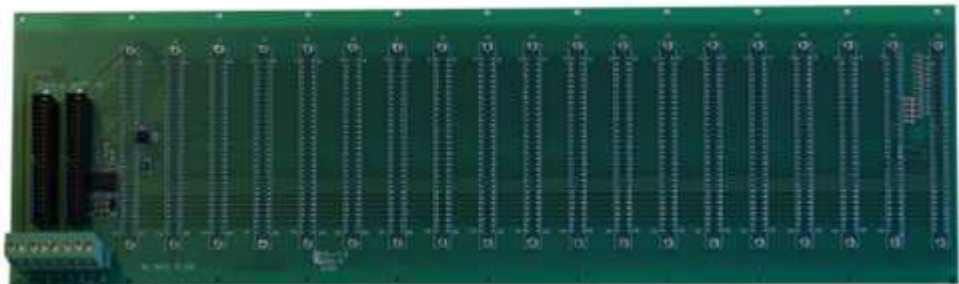
J22 power connector
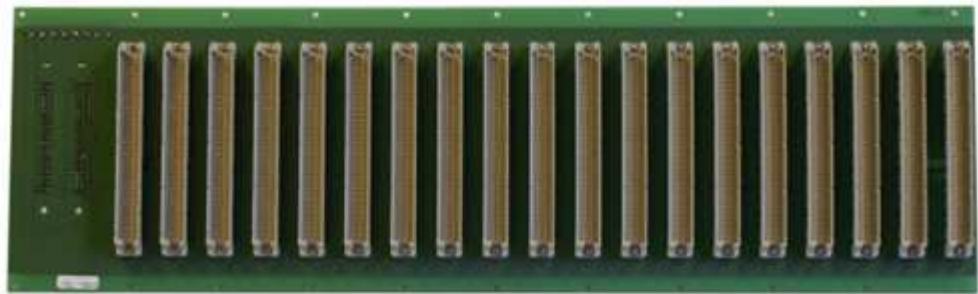
10.11.4 Layout

10.11.5
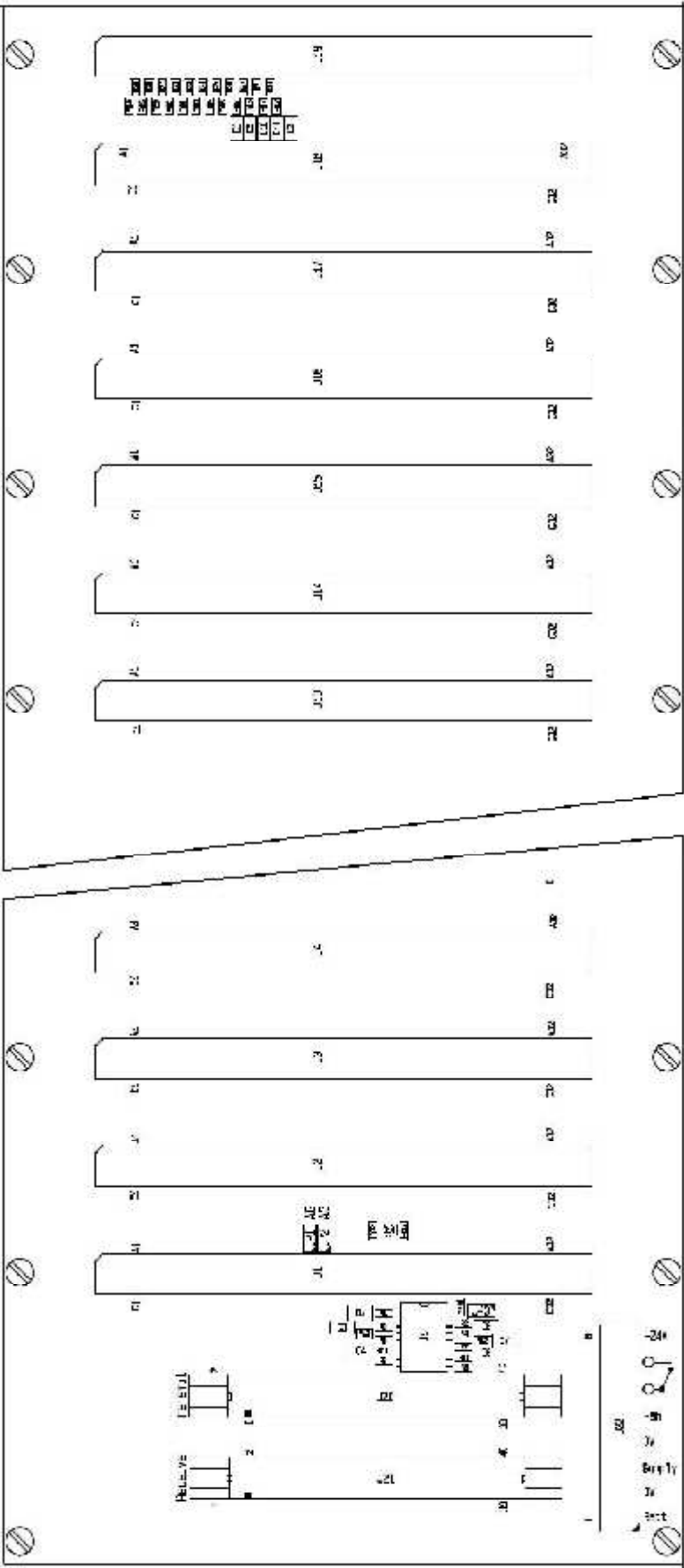
11. Stations

The SeaCom 211, SeaCom 220, SeaCom 421 and the SeaCom 411 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 SeaCom 211, SeaCom 220, SeaCom 421 and the SeaCom 411 flush mount stationis found in this paragraph,

11.1 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.

11.1.1 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.
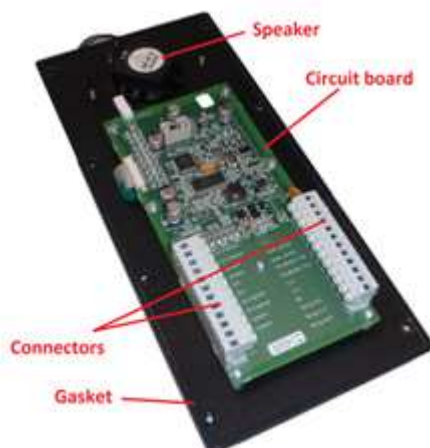
The picture show the connectors without the cable part to which the ship cables are connected.

11.1.2 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.

11.1.3 Gore vent

In order to make a pressure relief when temperature changes, a Gore vent is used. This vent is essential for keeping the interior 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.

11.1.4 Mounting brackets

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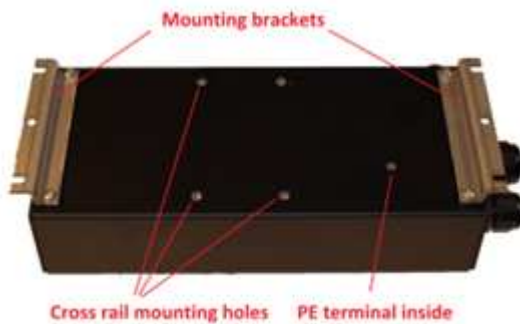
These are used for hanging the station on a bulkhead. In order to make it legal to terminate the cable shield to the PE terminal of the encapsulation, the two stainless mounting brackets are isolated from the aluminum box by isolating washers as seen on the below picture.

Do not corrupt the isolation.

11.1.5 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.
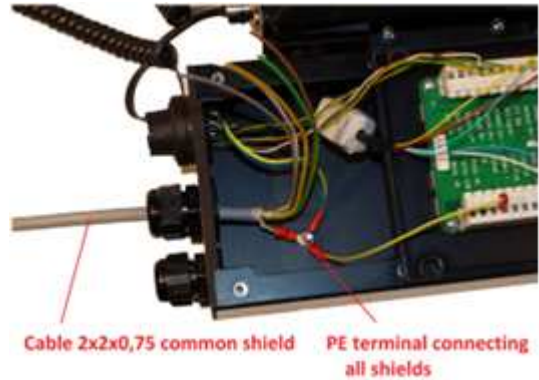
Refer to chapter 5.7 discussing shielding.

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

11.1.6 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.

11.1.7 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.

11.1.7.1 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.

Cable gland O-ring

Make fast the handset to the cross rail, using the 4 screws and the nylon washers.
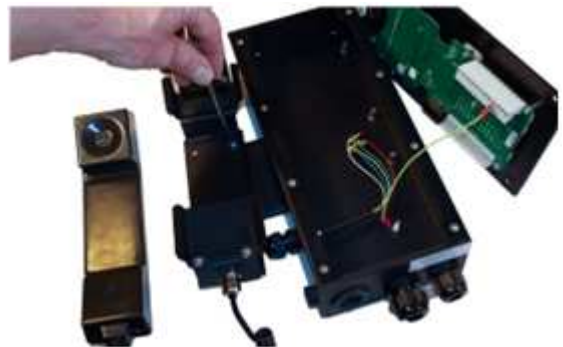
Tighten the PG9 cable gland so it forms a water tight cable pass through, and put on the ferrite delivered with the handset.

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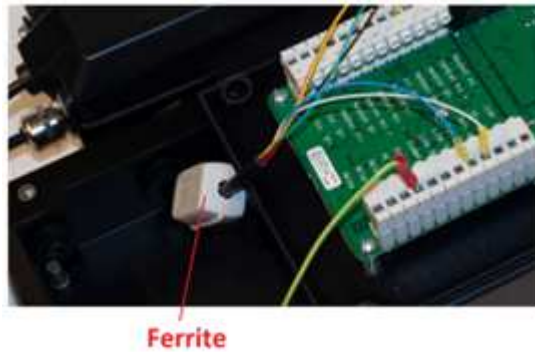
Connect the handset to the connectors of the circuit board. The use of the wires are:

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

11.1.8 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.

11.1.8.1 Mounting headset connector

The figure below shows the connector set:

Remove the grommet covering the headset connector hole.

Hole for water tight headset connector
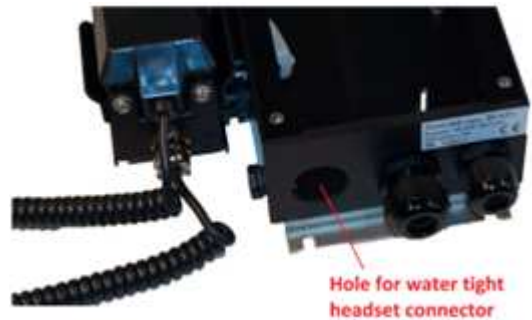
Pass the white connector through the hole, and make sure that the gasket is in place.

Connector

Gasket

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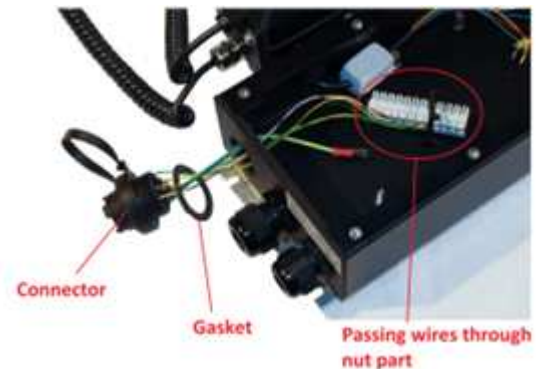
Tighten the nut, connect the PE terminal and connect the PCB connector part to the PCB.

Cap

Headset connector

Gasket

PE terminal

Microphone, spealer, PTT

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

11.1.9 Exposed equipment

The SeaCom 211, SeaCom 220, SeaCom 421 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 SeaCom 421 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.

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.

11.1.9.1 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.

11.1.10 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

11.1.11 Handset mechanical outline

Dynamic microphone 300 ohm   Red - Black
Earpiece speaker 300 ohm   Yellow - Green
Hook switch max. 50mA   Blue - White

Handset with cradle
10-400-1020

SeaCom

DWG090918CJ01_05
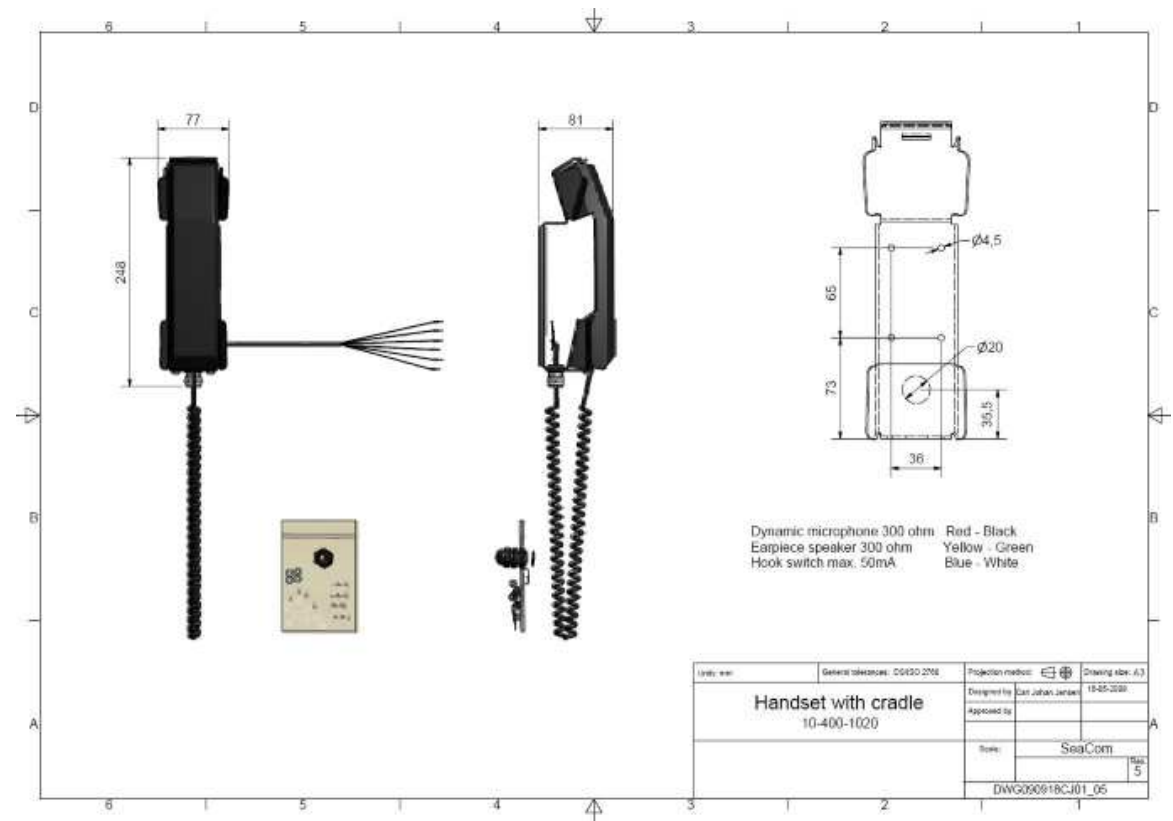
11.1.12 Cross rail drawing

Back Cross Rail
20-103-0020

Alu. ENAW-5754H22 (ALMG3), Sort-
eloxeret. Press stag stainles

SeaCom

DWG090611CJ08_02
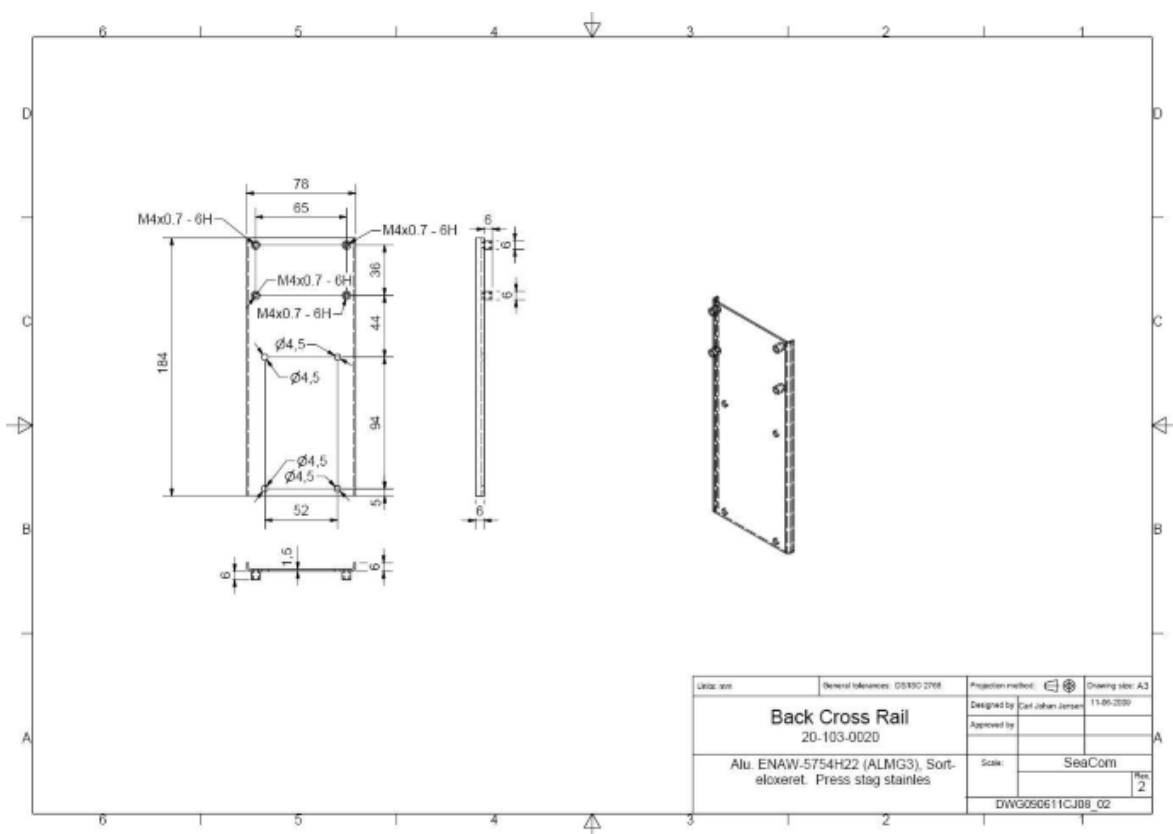
11.2 SeaCom 211

The picture below shows the SeaCom 211 in a full equipped configuration with headset and loudhailer.

This station is also called the talk-back station.

11.2.1 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 SeaCom 411 or SeaCom 421, 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 SeaCom 211 end. Such a call can be made to a group of up till 10 of the SeaCom 211 stations, forming a command group call where all stations are listening when the initiator is speaking.

11.2.2 Specification

The SeaCom 211 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

11.2.3 Inside
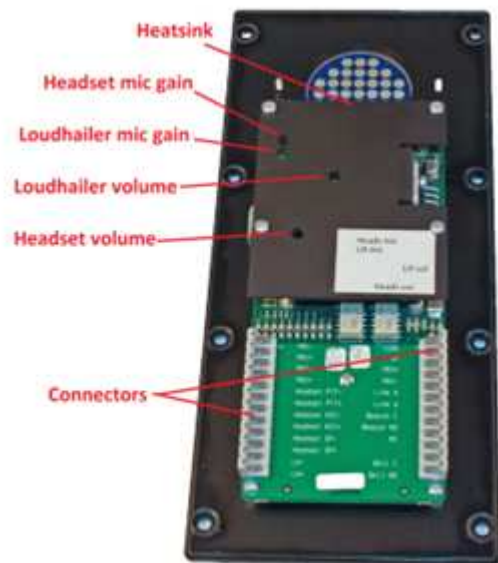
11.2.4 Electrical connections

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.

11.2.5 Jumper field

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

Jumper field

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	

11.2.6 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

11.2.7 Operating

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

11.2.7.1 Making calls

Press one of the buttons of the SeaCom 211 or the PTT button of the headset. This will cause the SeaCom 211 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.

11.2.7.2 Receiving calls

When anyone calls the SeaCom 211 station, 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 SeaCom 211 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.

11.2.7.3 Conducting calls

The SeaCom 211 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.

11.2.7.4 Terminating a call

The SeaCom 211 will terminate the call automatically when the far end terminates.

11.2.7.5 Receiving paging calls

Public address calls are received by the SeaCom 211. 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.

11.2.7.6 Command group (talk-back)

The SeaCom 211 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 SeaCom 211 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 chapter 12 for information on how to configure and set up a talk-back call.

11.2.8 Mechanical outline

SC211 - P211 Outline and mounting

Mounting will ensure isolation between ship hull and the aluminium box of the communication station.
Cable shields are to be terminated at the PE terminal inside station.
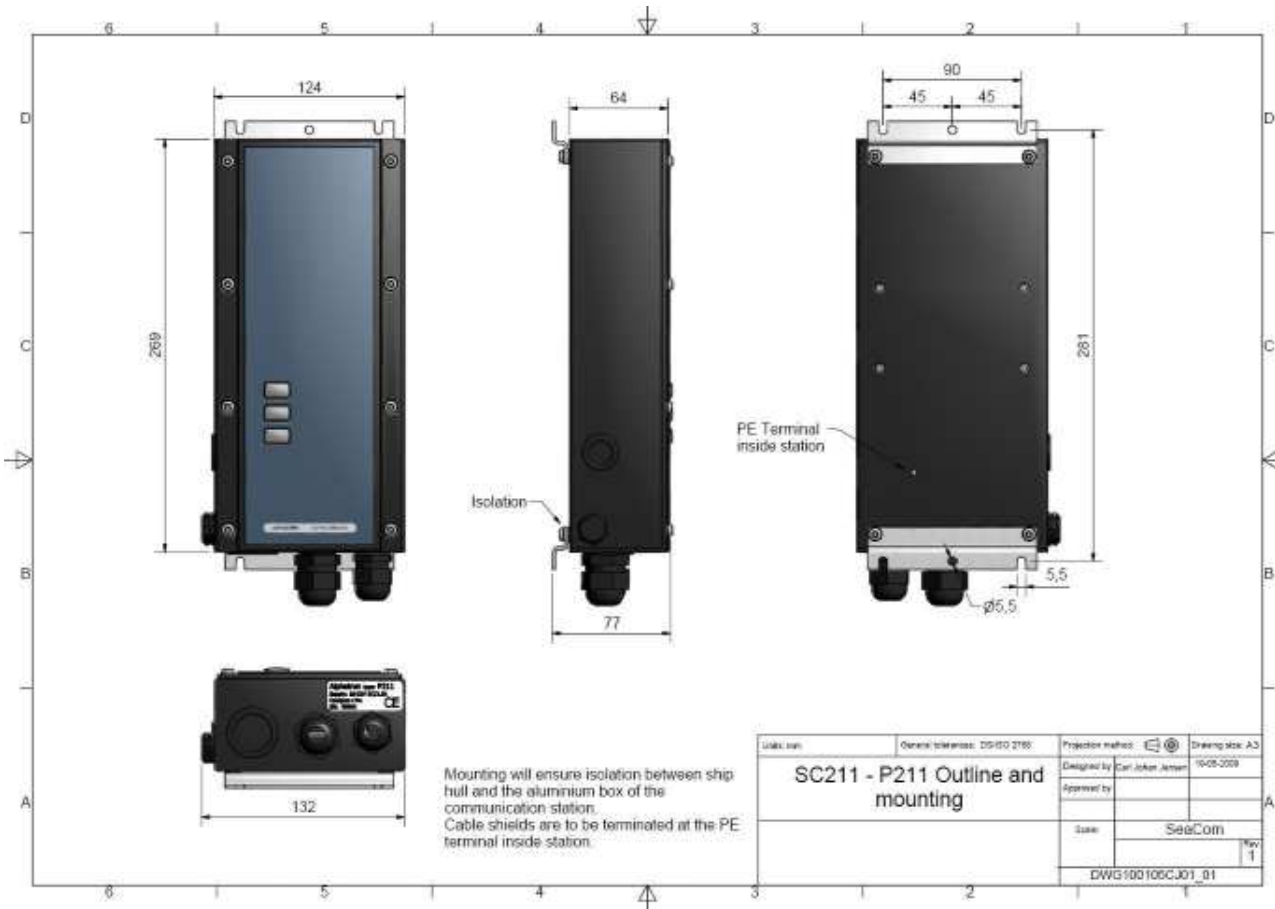
11.2.9 PCB layout
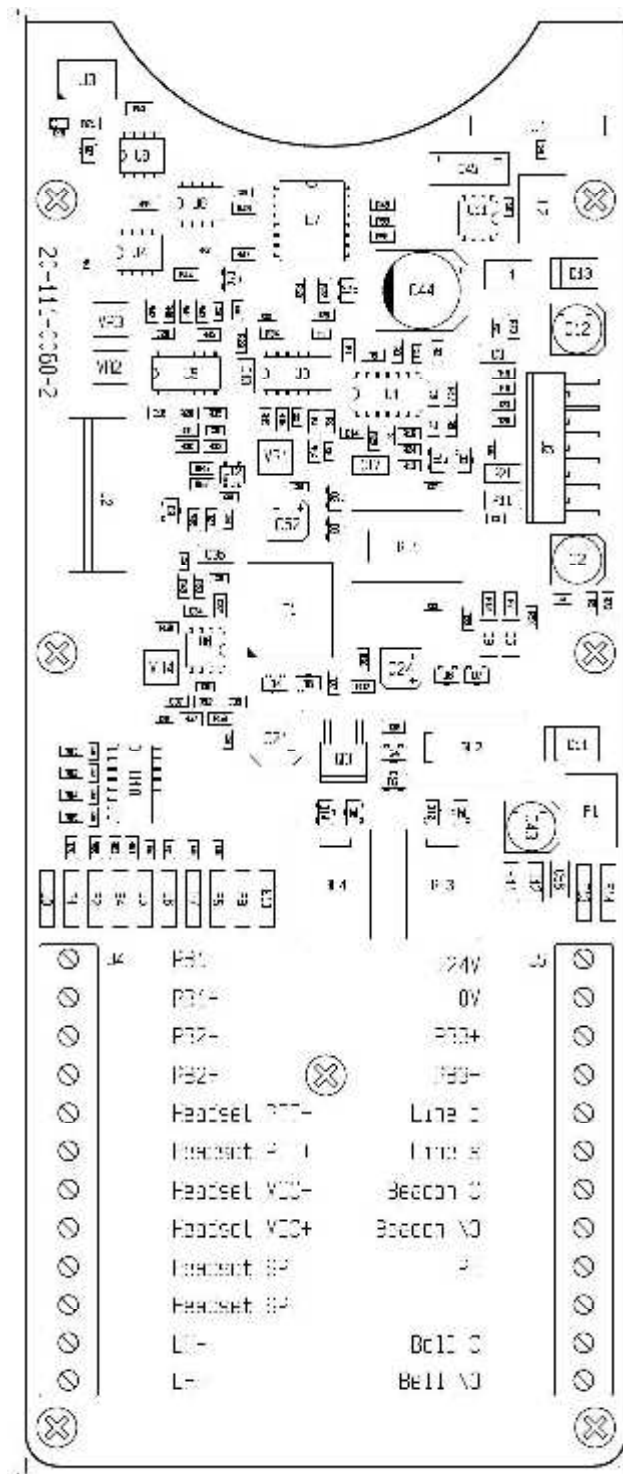
11.2.10 Schematic
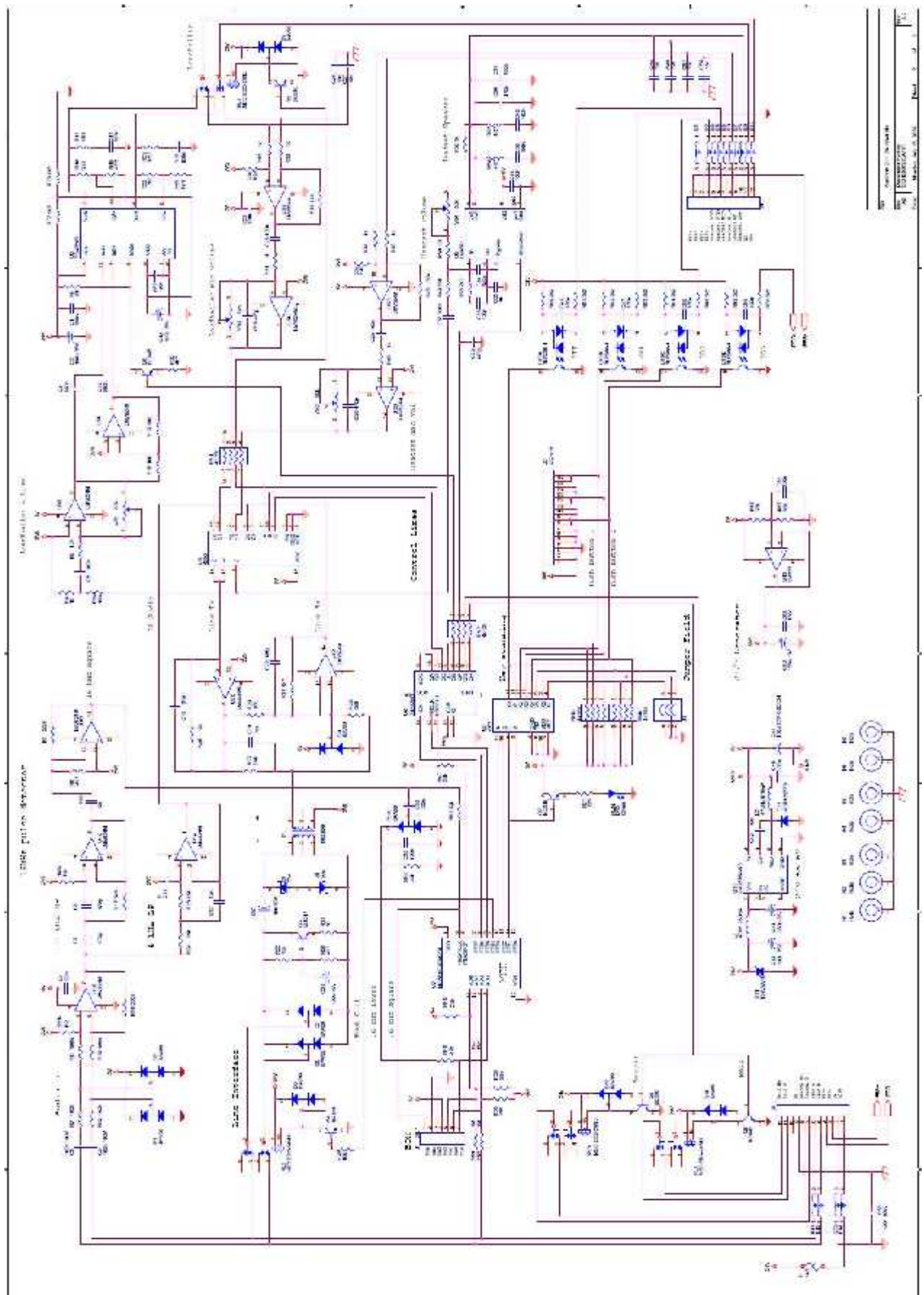
11.3 SeaCom 220

The picture below shows the SeaCom 220 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.

11.3.1 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 SeaCom 421 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.

11.3.2 Specification

The SeaCom 220 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

11.3.3 On the front

The front of the SeaCom 220 has the below functions:

11.3.4 Inside
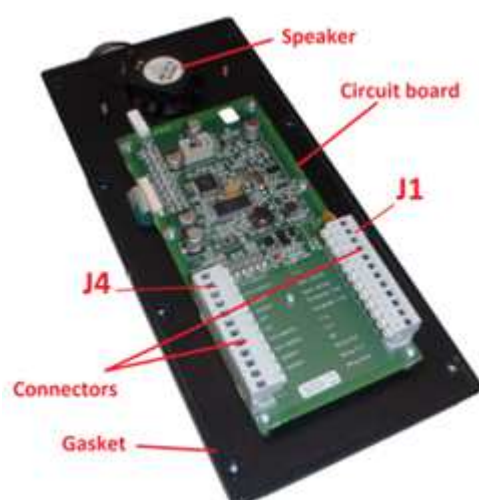
11.3.5 Electrical connections

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.

11.3.6 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

11.3.7 Operating

This paragraph describes how to use the SeaCom 220 station as seen from the end users view.

11.3.7.1 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.

11.3.7.2 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.

11.3.7.3 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.

11.3.7.4 Receiving calls

When anyone calls the SeaCom 220 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.

11.3.7.5 Timed termination

The SeaCom 220 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.

11.3.7.6 Busy tone termination

The SeaCom 220 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.

11.3.7.7 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.

11.3.8   Mechanical outline

11.3.9 PCB layout

22 Handset microphone	N.U. 10
23 Handset microphone	N.U. 11
24 Handset speaker	Hook Switch 12
25 Handset speaker	Hook Switch 13
26 Headset PTT	Telephone line 14
27 Headset PTT	Telephone Line 15
28 Headset microphone	N.U. 16
29 Headset microphone	N.U. 17
30 Headset speaker	PE 18
31 Headset speaker	Relay N.C. 19
32 N.U.	Relay COM 20
33 N.U.	Relay N.O. 21
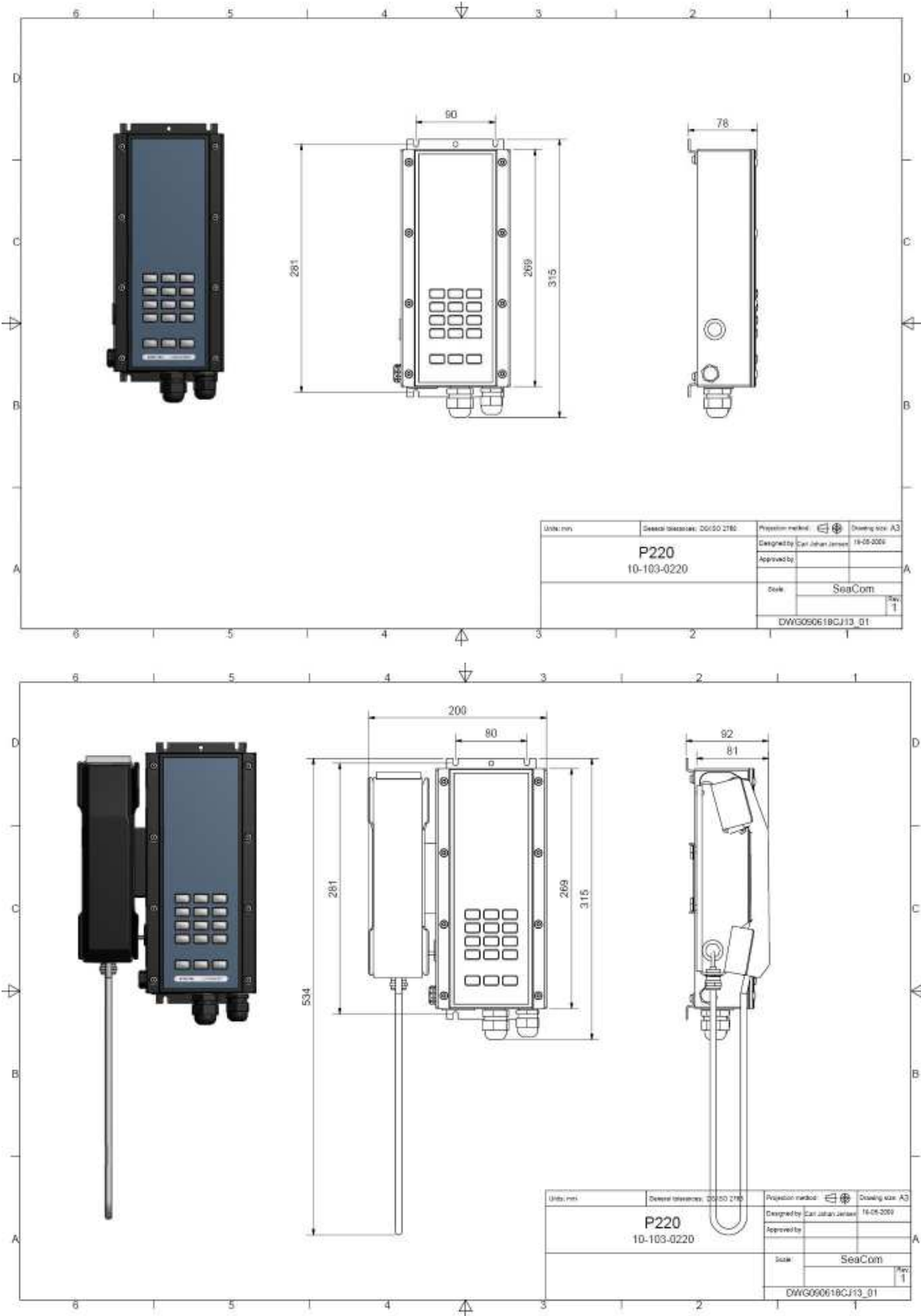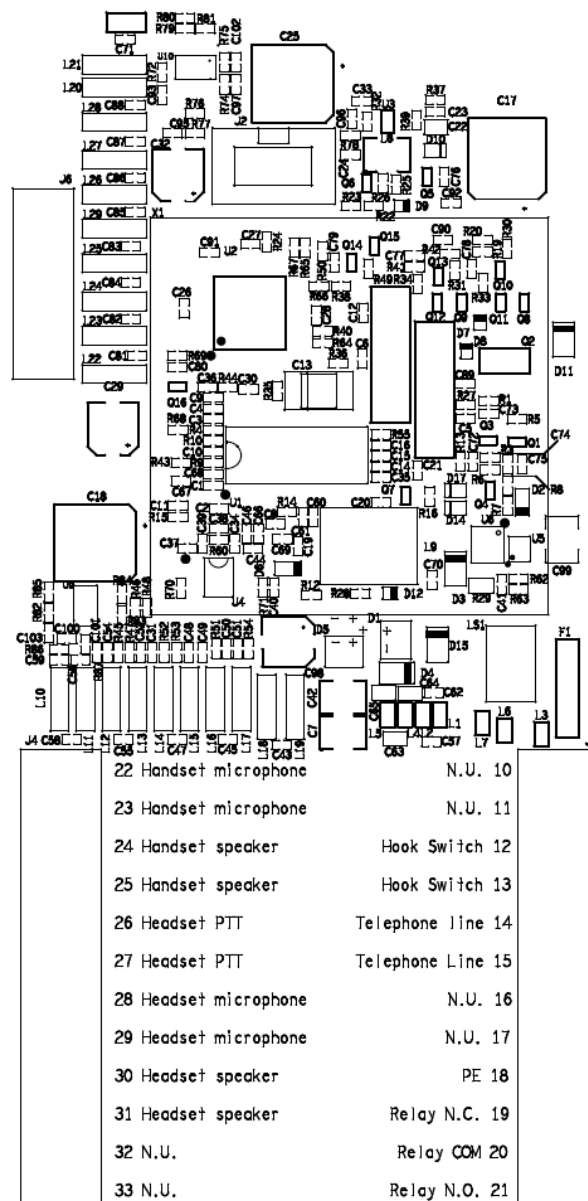
11.3.10 Schematic
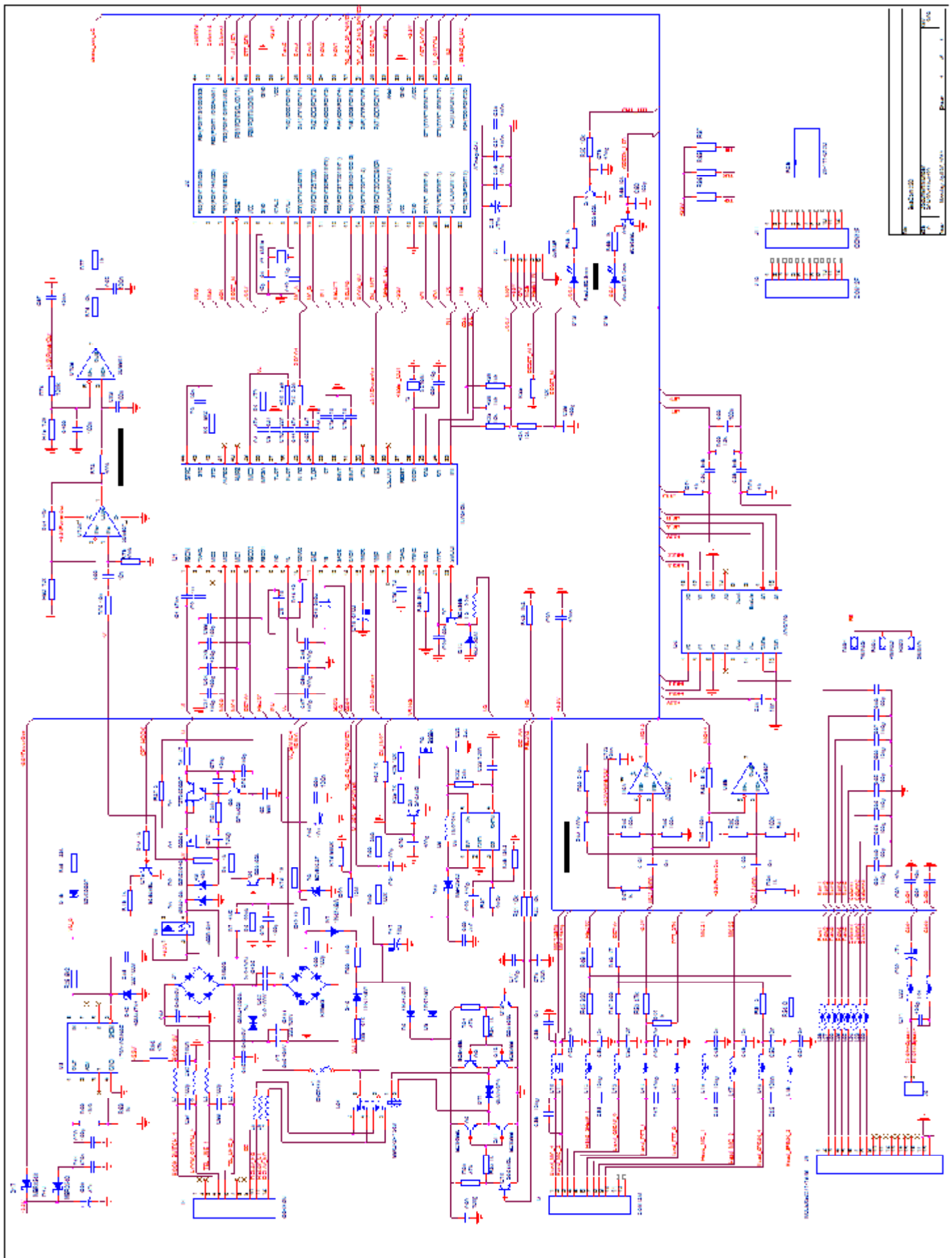
11.4 SeaCom 411 / SeaCom 421

The SeaCom 411 and the SeaCom 421 are our topmost stations will all features.

The picture below shows the SeaCom 411 in a basic version without any accessories.

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

The picture below shows the SeaCom 421 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 SeaCom 411is a flush mounted IP22 station with an extra high quality speaker, whereas the SeaCom 421 is an IP65 enclosed station looking very much like the stations described in the above chapters.

11.4.1 Description

The SeaCom 411 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 SeaCom 411 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 SeaCom 211 talk back stations are controlled. The speech direction will be controlled by the microphone key of the SeaCom 411 or the SeaCom 421 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 SeaCom 421 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 SeaCom 421 can 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.

11.4.2 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 ^{o}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

11.4.3   On the front

The front of the SeaCom 411 and the SeaCom 421 is shown below:

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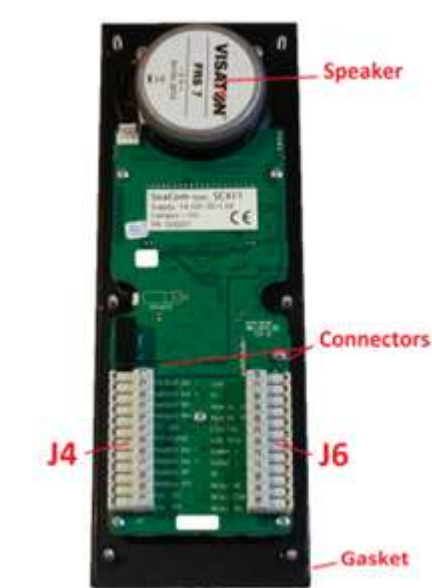
The picture shows the station without the shielding mounted, so that the PCB can be seen.

11.4.5   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.

11.4.6   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 SeaCom 411 and the SeaCom 421 station

J2:

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

11.4.7 Keys

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

11.4.7.1 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.

11.4.7.2 Up down keys

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

11.4.7.3 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.

11.4.7.4 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.

11.4.7.5 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 SeaCom 411 to one or more SeaCom 211 stations.

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

11.4.7.6 R key (transfer)

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

11.4.8 Display

The SeaCom 411 and the SeaCom 421 have a display which shows incoming calls, shows dialled digits and holds some status and information fields.

10/27 14:08 S	Time / status row
35	Digits row
HANDS FREE	Information row

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.

11.4.8.1 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

11.4.9 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)

11.4.10 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.

11.4.11 Mounting the SeaCom 411

The SeaCom 411 is differing from the other stations by beeing a flush mount station. This chapter describes the installation. The SeaCom 411 is and is meant to be mounted without any screws visible from the top. Refer to the console cut out drawings chapter 11.4.14

It is recommended that the station is isolated from the console frame mechanics. This is

accomplished as the SeaCom 411 has an isolating gasket at the rim of the aluminum frame

Isolating gasket

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

Isolating gasket

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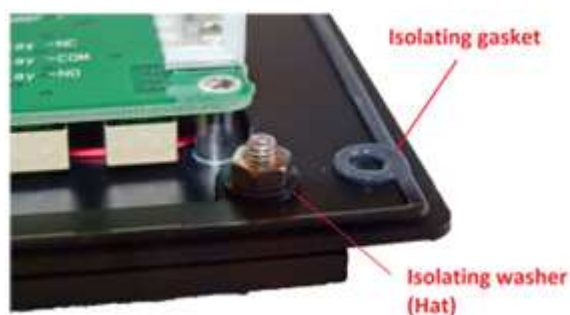
The cable coming from the exchange is a shielded 2x2x0,75 with common shield. The shield must be terminated at the PE terminal of the station.

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.

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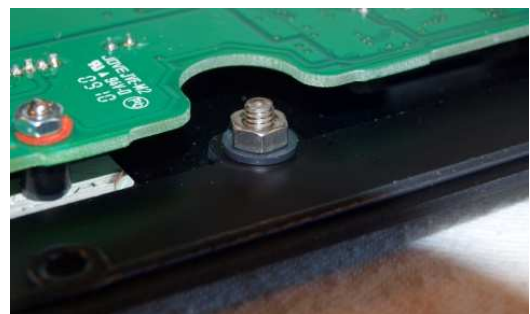
And its mounting

11.4.12 Operating

This paragraph describes how to use the stations.

11.4.12.1 Modes of conversation

The SeaCom 411 and the SeaCom 421 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)

11.4.12.2 Dialing

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

11.4.12.2.1 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.

11.4.12.2.2 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 11.4.13.2.15.

11.4.12.2.3 Last number re-dial

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

11.4.12.2.4 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.

11.4.12.3 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.

11.4.12.4 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.

11.4.12.5 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.

11.4.12.6 Terminating a call

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

11.4.12.7 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.

11.4.12.8 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 SeaCom 411 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.

11.4.12.9 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.

11.4.13   The menu system

The SeaCom 411 and the SeaCom 421 has a comprehensive menu system which is described in this chapter.

11.4.13.1   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.

11.4.13.2   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.

11.4.13.2.1   Speaker volume

Used to preset the speaker volume.

11.4.13.2.2   Backlight

Adjust level of backlight.

11.4.13.2.3   Ringer volume

Set the level of the ringing sound.

11.4.13.2.4   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.

11.4.13.2.5   Bell signal

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

A sample is played when the type is altered.

11.4.13.2.6   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.

11.4.13.2.7   Call in

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

11.4.13.2.8   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.

11.4.13.2.9   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.

11.4.13.2.10   Direct in

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

11.4.13.2.11   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.

11.4.13.2.12   External speaker

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

11.4.13.2.13   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.

11.4.13.2.14   Speed dial

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

11.4.13.2.15   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.

11.4.13.2.16   Dial signal

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

11.4.13.2.17   Loop dial type

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

11.4.13.2.18   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 SeaCom 211 station.

11.4.13.2.19   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 SeaCom 211 station.

11.4.13.2.20   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 SeaCom 411 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 SeaCom 411 itself.
If hands free communication is wanted, a short push will turn the SeaCom 411 into hands free.

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

11.4.13.2.21   Headset microphone type

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

11.4.13.2.22   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.

11.4.13.2.23   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.

11.4.13.2.24   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.

11.4.13.2.25   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.

11.4.13.2.26   External microphone gain

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

11.4.13.2.27   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)

11.4.13.2.28   Relay hold off

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

11.4.13.2.29   Contrast

Adjusting the contrast of the display.

11.4.13.2.30   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.

11.4.13.2.31   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.

11.4.14 Mechanical outline

11.4.15 PCB layout
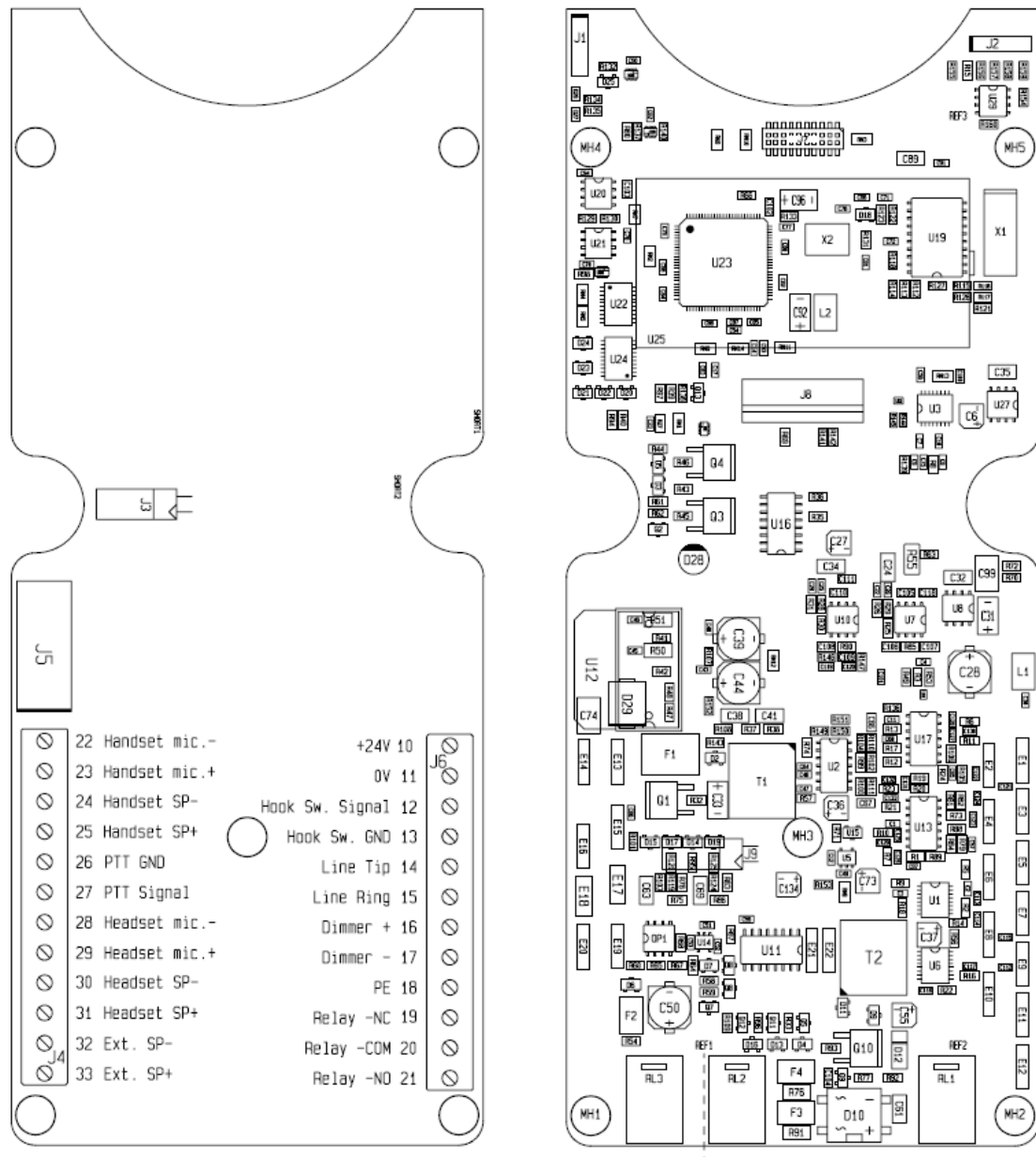
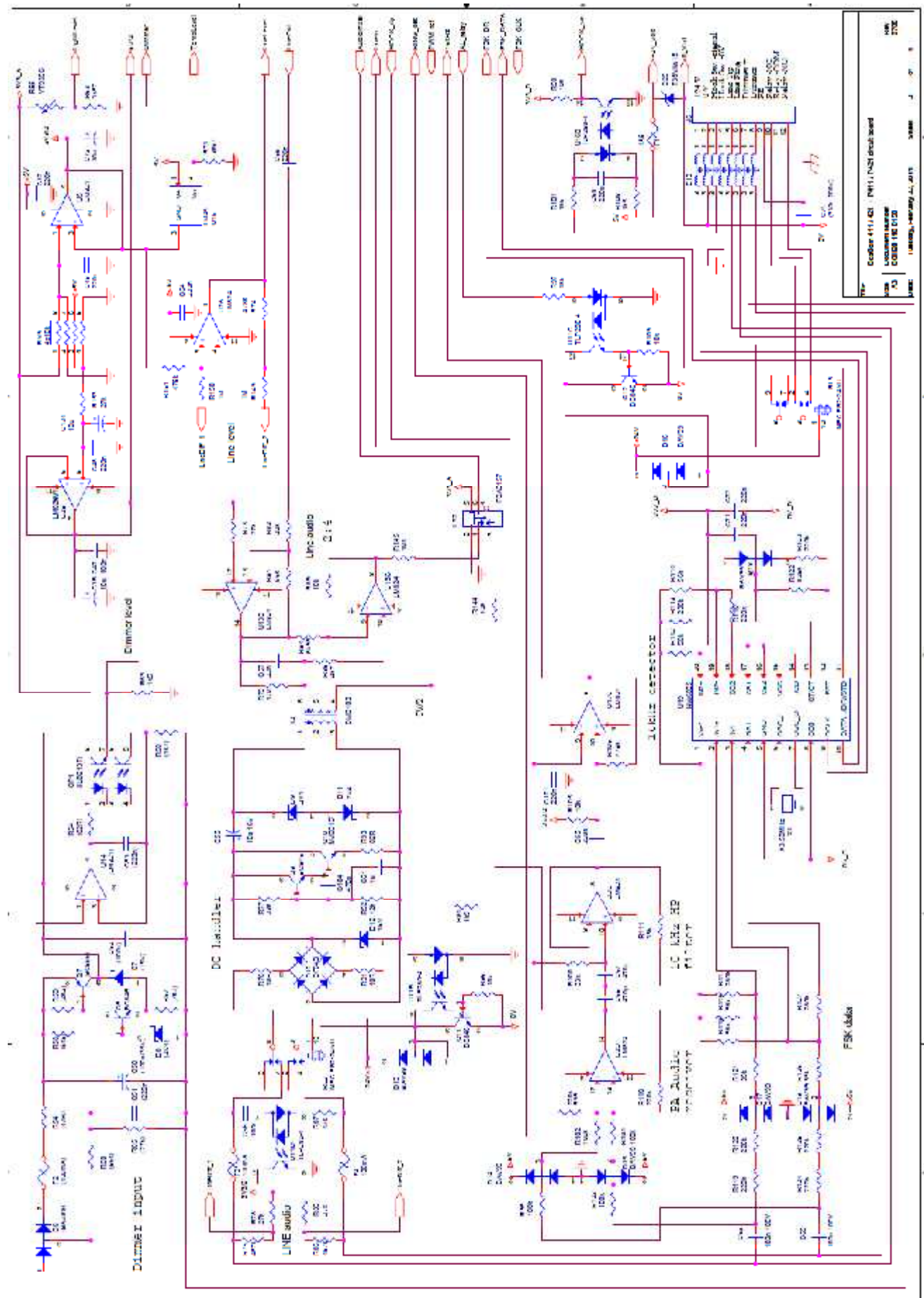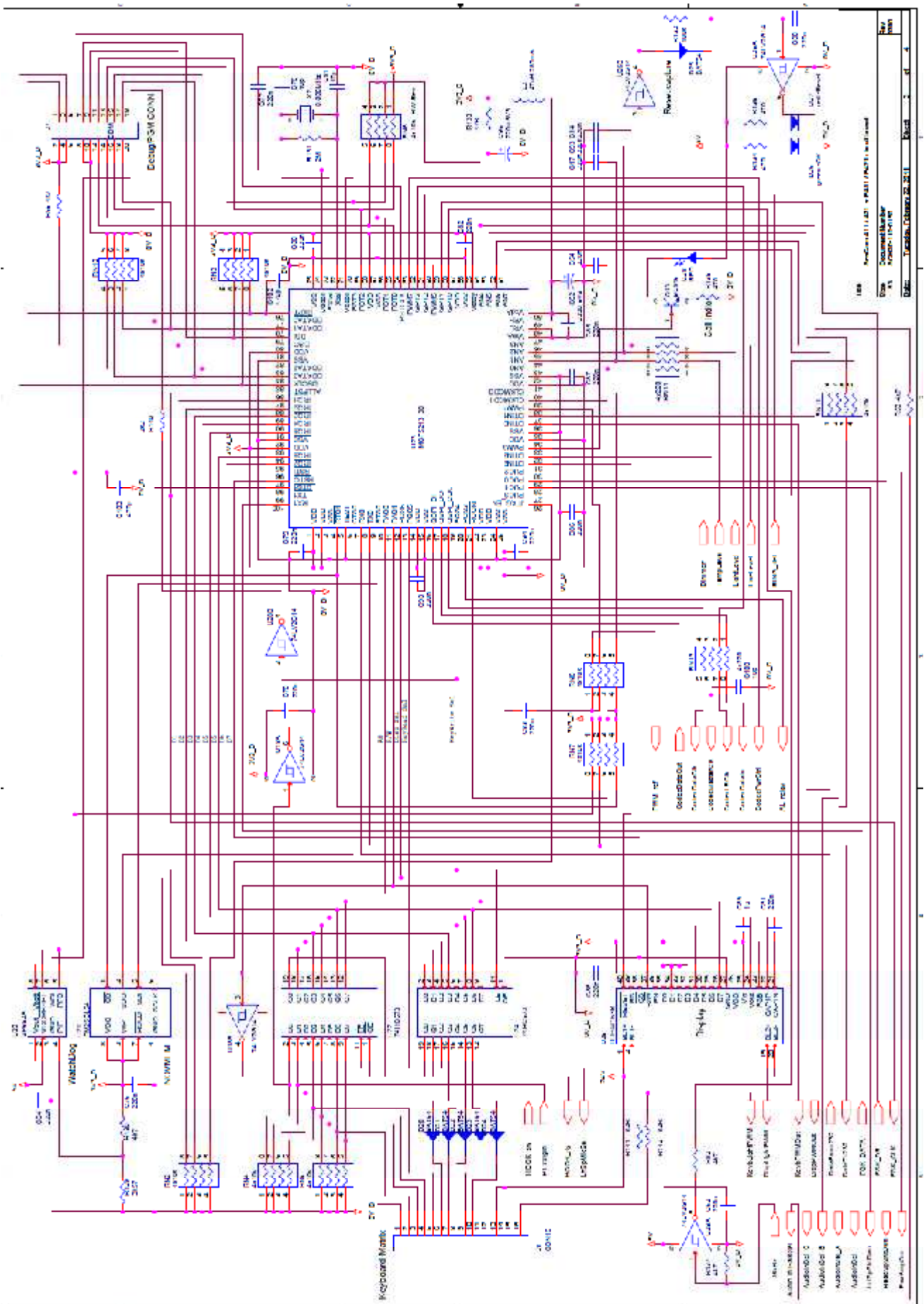|			
22	Handset mic.-	+24V	10
23	Handset mic.+	0V	11
24	Handset SP-	Hook Sw. Signal	12
25	Handset SP+	Hook Sw. GND	13
26	PTT GND	Line Tip	14
27	PTT Signal	Line Ring	15
28	Headset mic.-	Dimmer +	16
29	Headset mic.+	Dimmer -	17
30	Headset SP-	PE	18
31	Headset SP+	Relay -NC	19
32	Ext. SP-	Relay -COM	20
33	Ext. SP+	Relay -NO	21
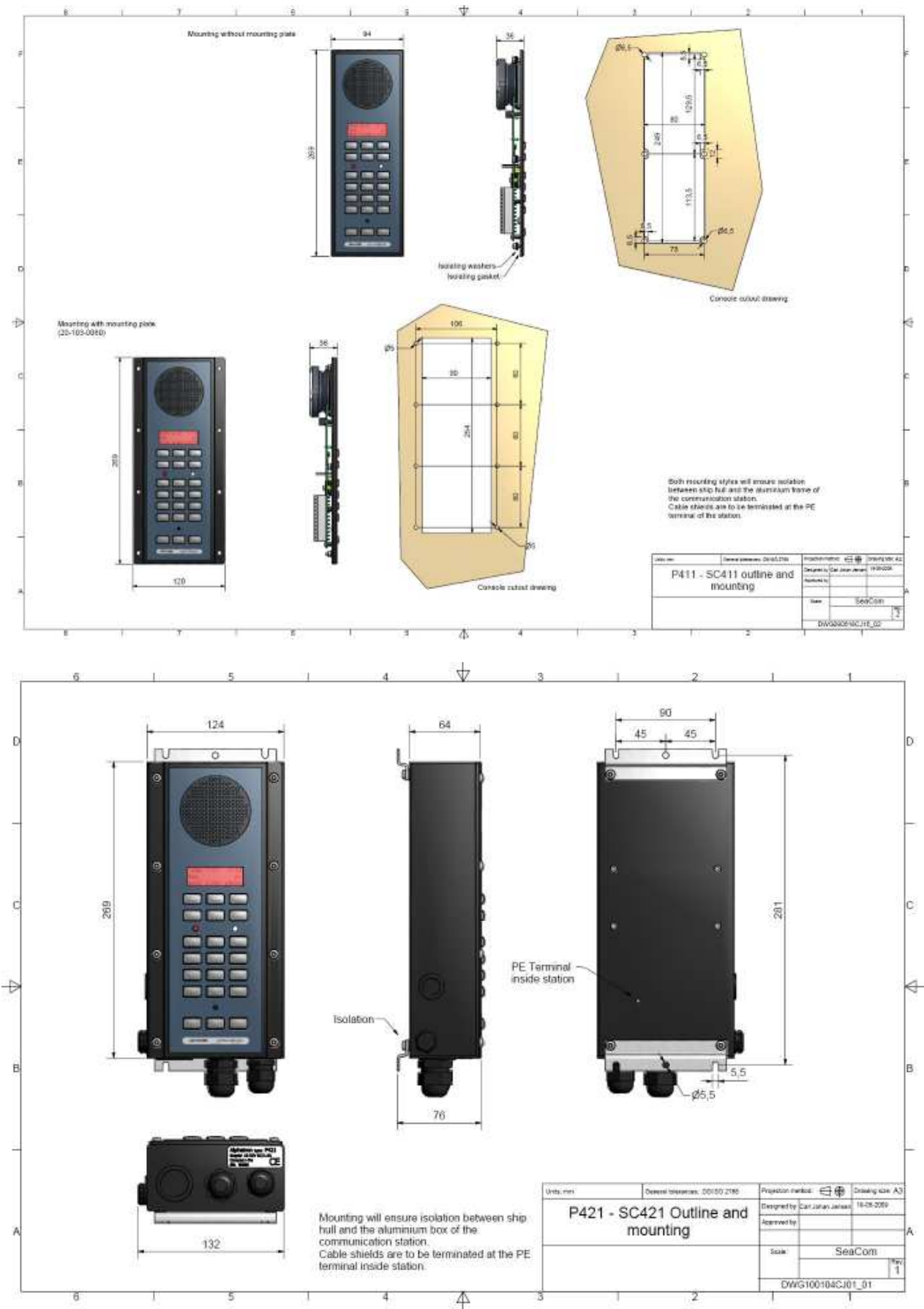
11.4.16 Schematic
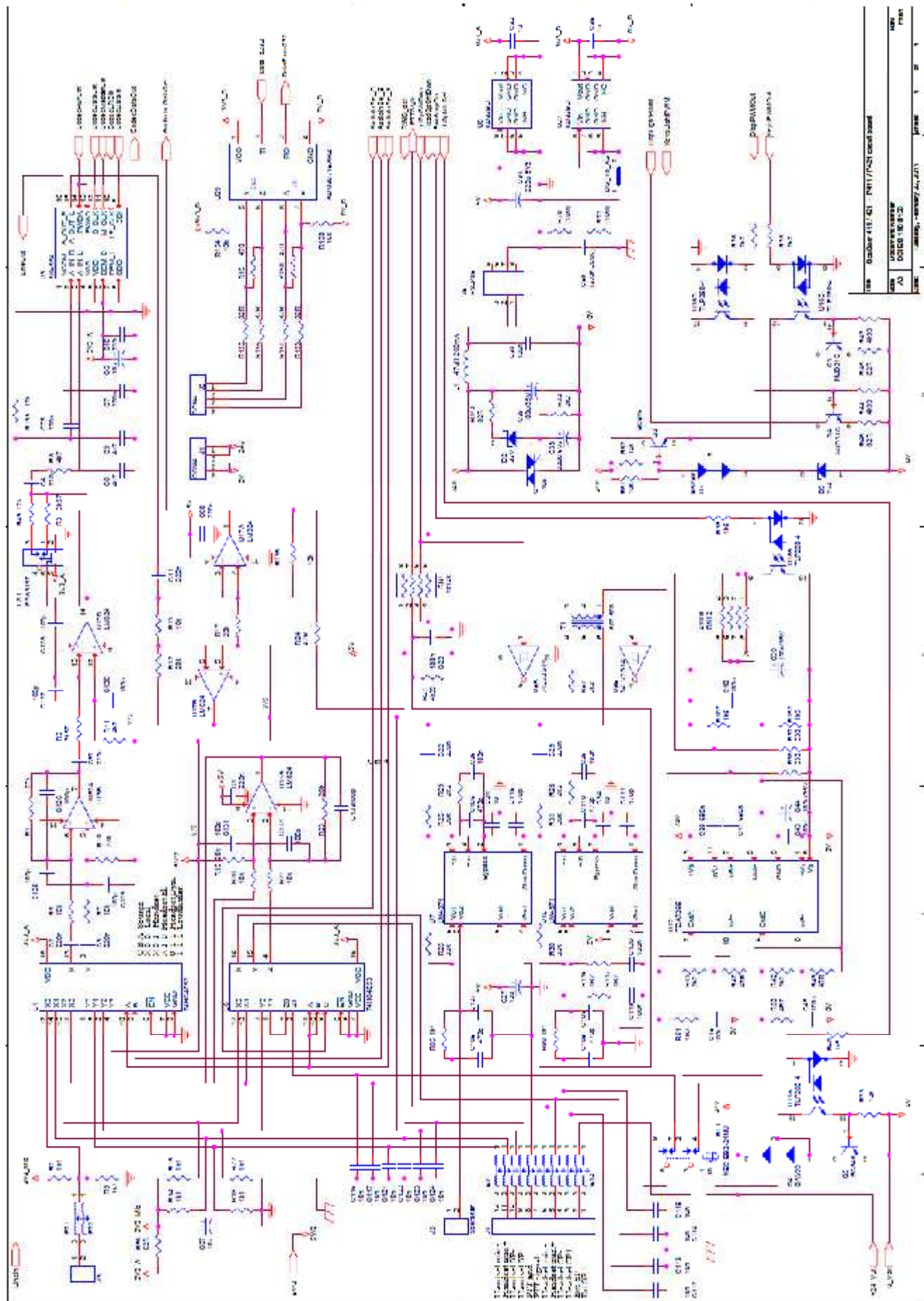
12. System programming

The SeaCom system 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 our configuration file editing tool software:

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

12.1   Getting access to the system

There are two ways of accessing the configuration file:

- USB access
- Network access

The USB access is used for small adjustments to an existing installation, whereas the network access gives the full opportunities for editing and debugging.

12.1.1   USB access to the config. file

In order to get USB access to the configuration file, a SeaCom USB key must be used.

Insert the key into the lowest USB port of the powered on and operating SeaCom system. Wait 15 seconds for the red indicator of the key to stop flashing.

Now the configuration file is found on the USB key in the folder \SeaCom2000 - ready to edit.

Insert the USB key into the computer that is meant to be used for SeaCom system

maintenance. The contents of the USB key \SeaCom2000 will show up like:

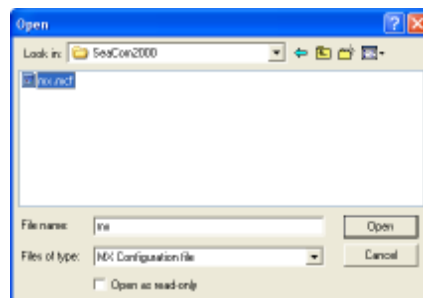The mx.mcf file is the configuration file

On the administrative computer locate the MXconfig.exe application.

The SeaCom supplier shall upon installation and commissioning of the system, make the MXconfig.exe file available to the person responsible for making changes to the system configuration. It is NOT on the USB key by default, and the version of the MXconfig.exe must match the version actually found on the SeaCom system
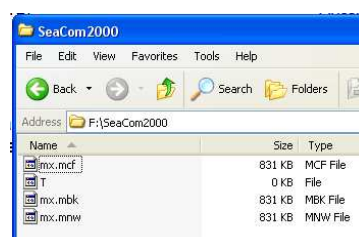
Launch the MXconfig application, and when the file open dialog form appears, navigate to the USB key disk \SeaCom2000 folder and open the mx.mcf file.

Now the configuration tool can be used and the alterations to the configuration be made.

After editing the mx.mcf file, close the MXconfig application.
Now the contents of the \SeaCom2000 folder on the USB key looks like:

The mx.mbk is a backup copy made by the MXconfig before editing.
The mx.mnw is the file holding the modified configuration.

Take the USB key and insert into the USB port of the SeaCom system. The system will now read the mx.mnw file, bring it into use, and delete it from the USB key. Verifying that the file is deleted is a way of getting a "received file acknowledge".

12.1.2 USB backup

When a USB disk is inserted, the system will automatically make a full system backup folder onto the USB disk. The SystemBackup folder holds a full copy of the contecnts of the SeaCom2000 folder of the CP board.
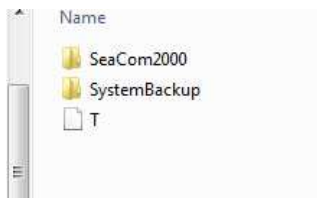
12.1.3 Network access

In order to get network access to the system disk of the SeaCom system, a network cable and a labtop PC running Windows XP is needed.

Connect the network port of the CP board to the network port of the labtop PC using a twisted network cable. Newer PC does not need the twist, and can connect using any network cable.

Next we must establish connection between the computers on software level.
For the CP2 boards of revision earlier than 0500, the fixed IP address approach must be followed, but for rev 05xx and upwards, a DHCP server is found on the CP board, giving the possibility of using automatically assigned IP address.

12.1.3.1 Setting a fixed IP address

Fixed IP address must be used on CP revision earlier than 0500.

Start by giving the network port of the labtop a fixed IP address.

On the labtop PC, use the Start menu button and choose settings and network connections.
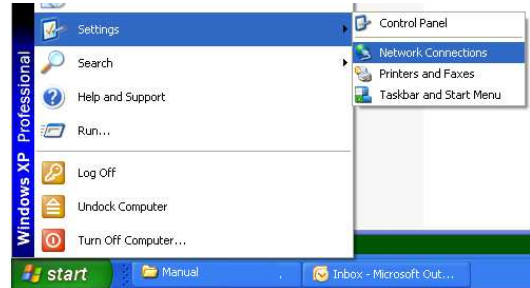
In the network list, choose the port to which the SeaCom system is connected, and navigate to the properties menu.

In the propertied dialog box for the network port, choose the TCP/IP properties button.

In the TCP/IP properties dialog box, enter a fixed IP address different from the address of the SeaCom system, which is 192.168.0.52.

The labtop now operates with a fixed IP address on the network port.

12.1.3.2   Using dynamic IP address

With CP2 rev 0500 and later, a DHCP service is found on the CP board, assigning IP addresses to computers connected.
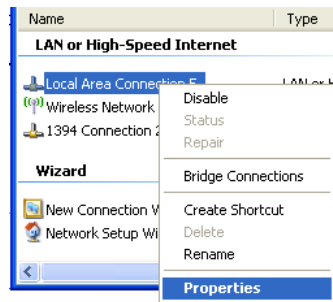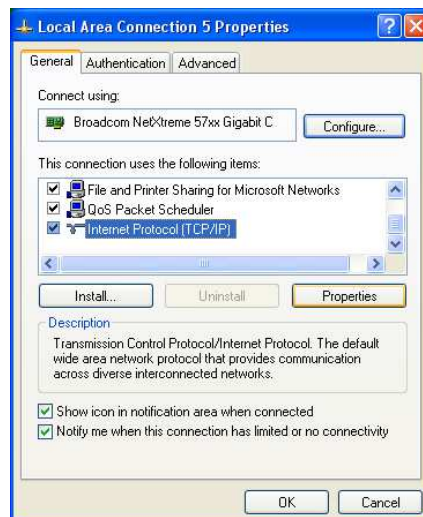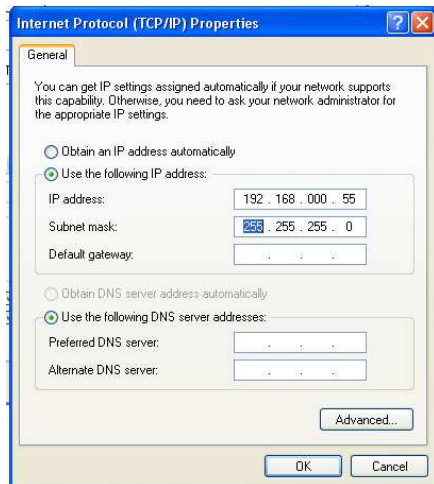As most labtops are configured for dynamic IP address, just connect, and you can find the SeaCom2000 disk using the explorer, or you can connect using the WINvnc (ref 14.3).

12.1.3.3   Mapping the system disk

Next task is to map the disk of the SeaCom system, so we can see and use the contents of the disk.

Use the Start menu button and navigate to Search for files or folders

Are You using Windowx XP you can follow the below steps.
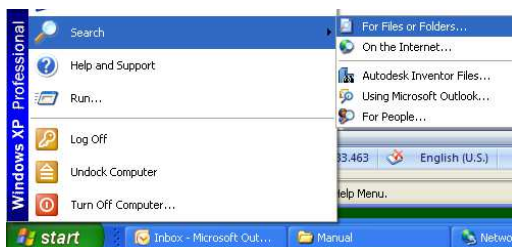
Navigate to the Search for computers menu and enter the computer name SeaCom2000 and click Search
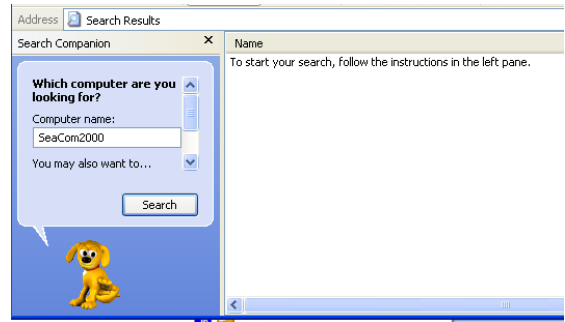
Wait for the search to show the computer
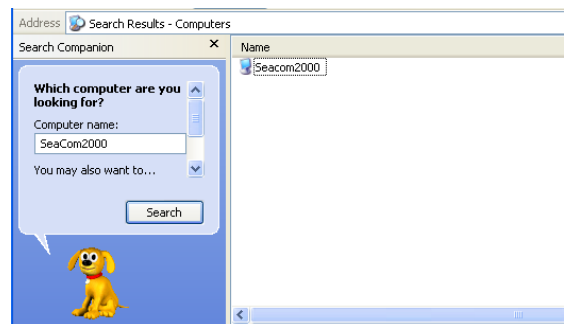
Are You using Windows 7 you can find the SeaCom2000 computer using the explorer looking for networks. If the computer does not show up automatically, try to search for it.
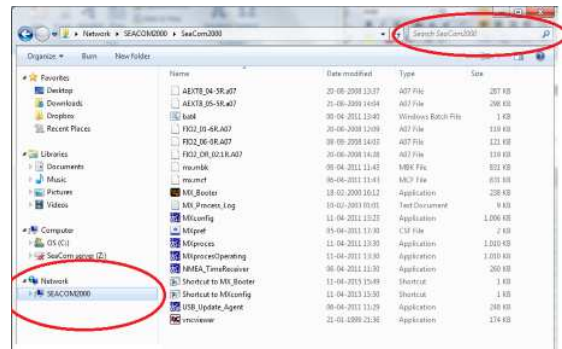
With any operating system, double click the Seacom2000 computer and it should show the password dialog box.

Enter:

User name	-	adm
Password	-	1017

And press OK.

When the contents of the system disk is displayed, right click on the SeaCom2000 folder and select the menu Map Network Drive
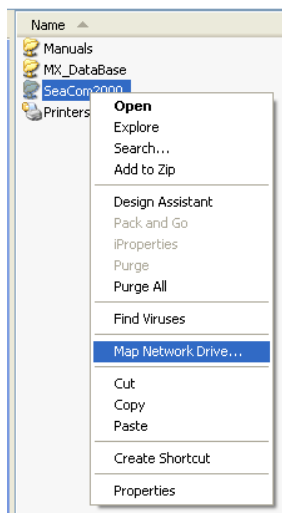
In the Map Network Drive dialog box, choose the drive S: and click finish.
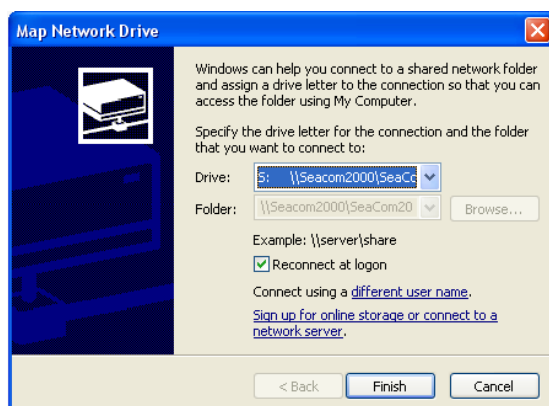
Now the system disk folder SeaCom2000 is mapped onto the labtop PC as the S: drive, and we are ready to inspect and use the contents using the Windows explorer.
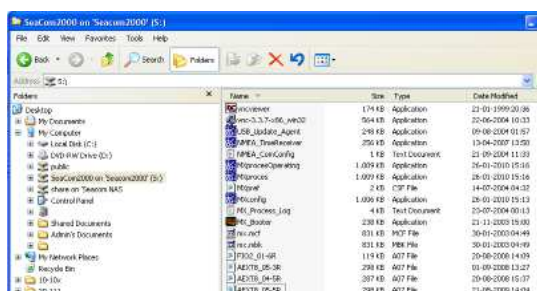
In the list, locate the MXconfig application and launch it.

When the file open dialog box appears, navigate to the S:\SeaCom2000 folder and open the mx.mcf file found there.
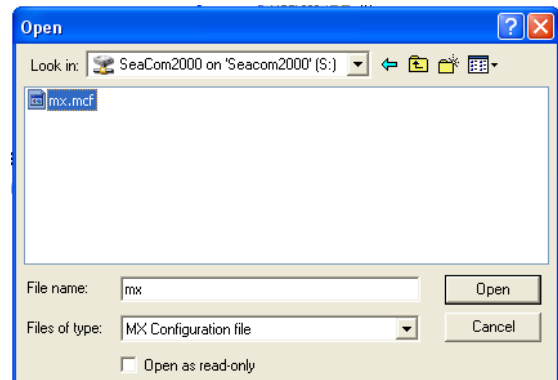
Now the configuration tool can be used and the alterations to the configuration be made.

After editing the mx.mcf file, close the MXconfig application. Now the application writes the modified configuration file onto the system disk, and brings it into operation.

12.2 General concepts

This chapter contains descriptions of basic set-up concepts.

12.2.1 Password

There are two levels of access in the MXconfig.exe application:

- User level
- Expert

When at first opening the MXconfig application, the user level is entered. On this level only call number, names and access privileges can be altered.

When entering the password: **1017** the expert level is entered, and all features of the configuration tool will be revealed.

12.2.2 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.
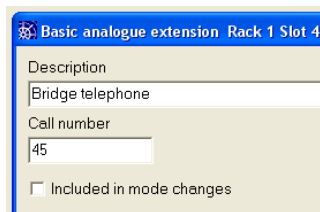
And all call numbers have a description attached, which is displayed in display telephones, and which can be used when printing telephone directories.

12.2.3 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.
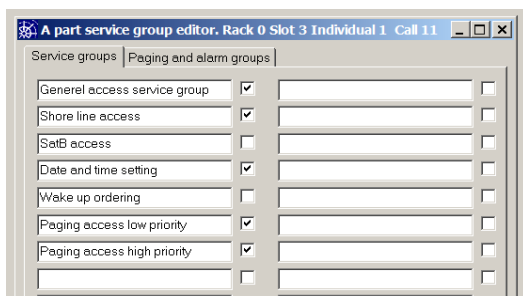
To open the editor used to include a call number to a service group, the below buttons are available for all call numbers:
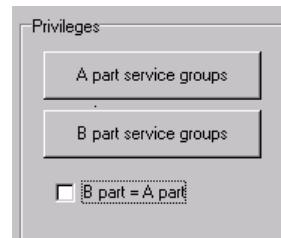
Assigning a service group to a call number can either be made for both directions – when the A=B is clicked, or a division between outgoing and incoming rights can be implemented when the A=B is unchecked.

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.

12.2.3.1 The 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.

12.2.4 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
- Day mode / night mode
- Conference

12.3 The MXconfig.exe application

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

12.3.1 The Files menu

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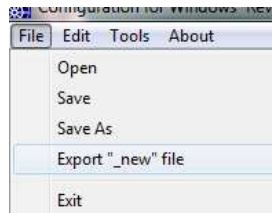
12.3.1.1 Open

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

12.3.1.2 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 Mxconfig application is closed.

12.3.1.3 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.

12.3.1.4 Export "_new" file

This menu is used to make the system create an "_new" file. This is a signal file used by the SeaCom 3000 system only.
Refer to own manual for SeaCom 3000.

12.3.1.5 Exit

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

12.3.2 The Edit menu

The edit menu is used to choose the way of accessing the configuration.
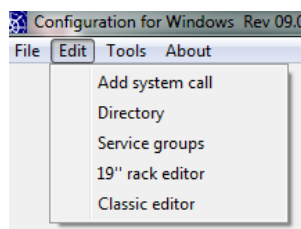
12.3.2.1 Add system call

Use this menu to open the system call selector.

12.3.2.2 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.

12.3.2.3 Service groups menu

This menu will open the service group name editor.
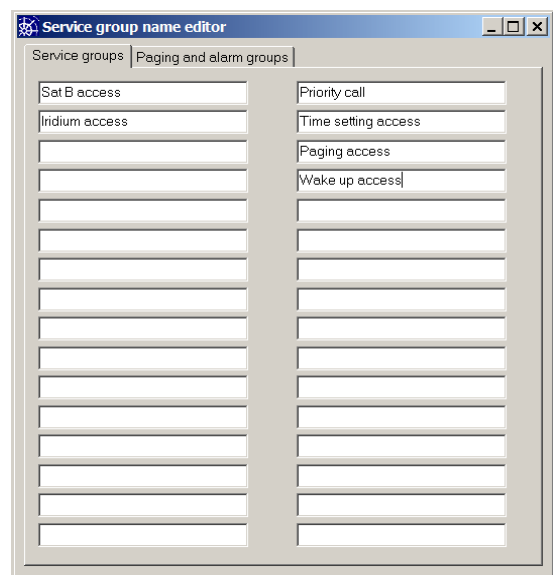
12.3.3 19" rack editor

This menu opens the physical editor to be used with SeaCom 2100 and 19" systems.
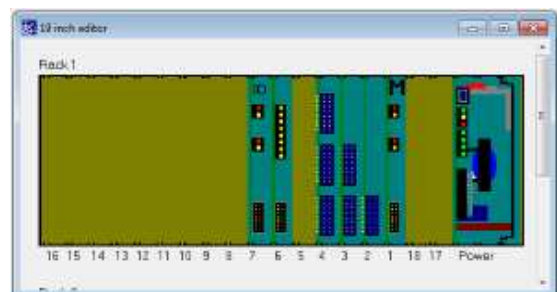
The window displays the boards mounted in the actual board magazine.
Note that board numbering starts from right going left.

12.3.4 Classic editor

This menu is only valid for the SeaCom 3000 system.
Refer to own manual for SeaCom 3000.
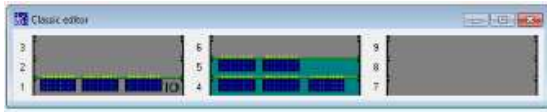
12.3.4.1 Selecting a new board type

Each rack 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.
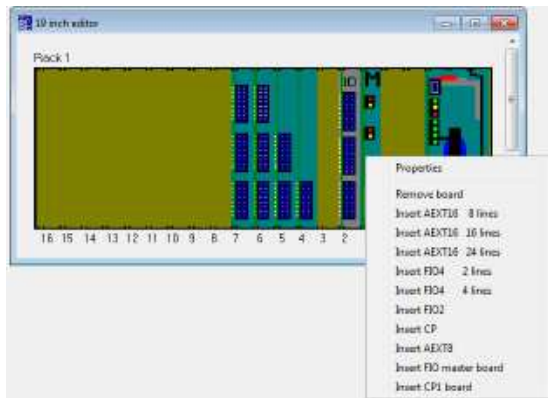
Note that this is a very powerful operation, as the old set-up data for the board location is lost, and new factory defaults are loaded.

The board types and their set-up parameters will be described in the following section.

The board editor is opened when double clicking on the picture of an installed board.

12.3.4.1.1 AEXT8 board

The 8 line analogue extension board editor looks like:
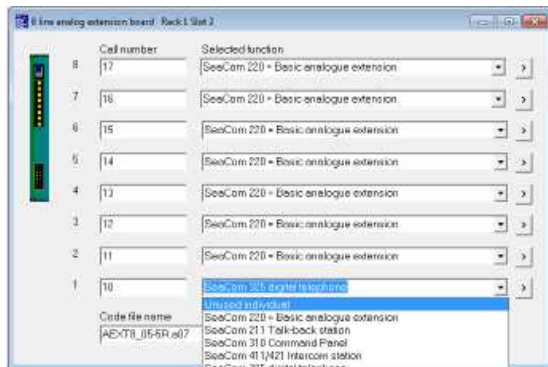
This board contains 8 individuals, which can be selected to operate either unused or as a basic analogue extension or as one of our communications station families.

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

12.3.4.1.2 FIO2 board

The 2 line FIO trunk board editor looks like:
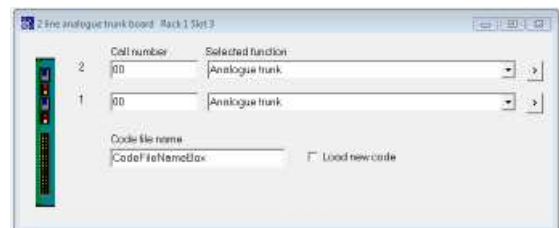
The two individuals of the trunk board can be given the functionality chosen by the drop down box below:
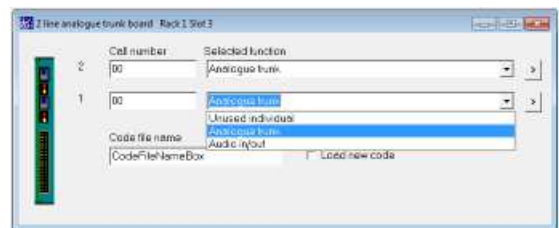
12.3.4.1.3 FIO2 Master

One and only one system master card must be found in order for the system to work.

The FIO2master, the AEXT16 and the FIO4 all can act as master.

The master board makes the backplane clock signals, holds the voice switch and makes communication to the PSU.

The figure below shows the set-up form of the FIO2 master card.
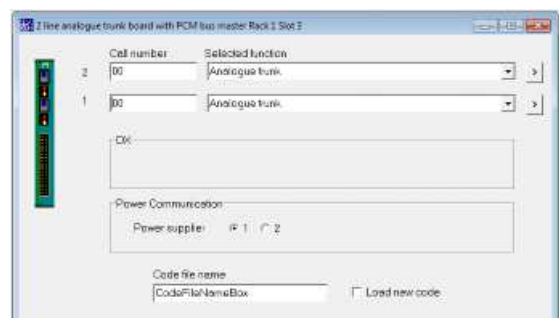
The properties form of the individuals can be opened from the board editor form by either double clicking on the call number or description or by pressing the > button.

12.3.4.1.4   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

12.3.4.1.5   FIO4 board

The FIO4 trunk board editor looks like:
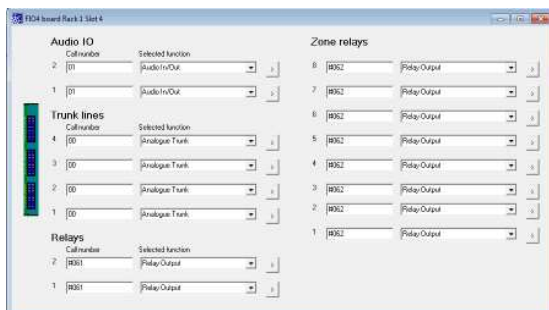
12.3.5   The tools menu

This menu is used for miscellaneous purposes:

12.3.5.1   Preferences

This menu opens the preferences editor form.
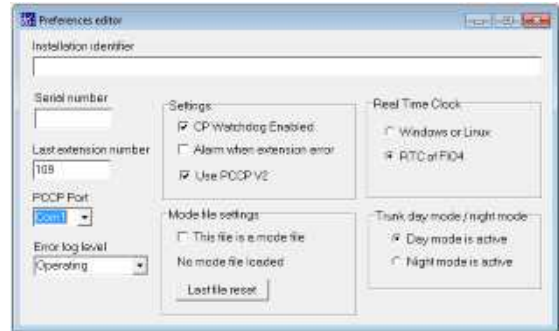
12.3.5.1.1   Installation identifier

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.

12.3.5.1.2   Serial number

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

12.3.5.1.3   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.

12.3.5.1.4   COM Port

Serial port used by the MXprocess.exe

12.3.5.1.5   Error log level

Selecting the error log level.

12.3.5.1.6   CP watchdog enable

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

12.3.5.1.7   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.

12.3.5.1.8   Use PCCP V2

Selecting the protocol between CP card and the system master card.
For FIO2master this box have to be unchecked.
When using AEXT16 or FIO4 as system master card, this box must be checked.

12.3.5.1.9   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.

12.3.5.1.10   Real time clock

This pane is used for selecting the source of system time. System time is used for displaying time on displays of telephones and for wake up purposes.

The SeaCom 2100 and 19'' systems are using CP cards running Windows. The CP2 and Windows do have a system clock of its own. So for these systems, check the "Windows or Linux".
Checking the "RTC of FIO4" will let the system read its time from an FIO4 master card, which has a RTC on board. This is used with SeaCom 3000 systems where the CP is running Linux.

12.3.5.1.11   Day mode / night mode

This is used to choose day or night mode on configuration / system startup time.
The user can dynamically swapt day and nignt mode by calling a "day / night mode" call number.

12.3.5.2   Renumber extensions

This menu is very powerful menu typically only used on start of a system programming.
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.

12.3.5.2.1   Change all extensions to SC325

Used to globally change all analogue extensions to the PA enabled SC325 telephone.

12.3.5.2.2   Clear file

WARNING. This menu will delete all content of the configuration file and install default values.

12.3.6   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
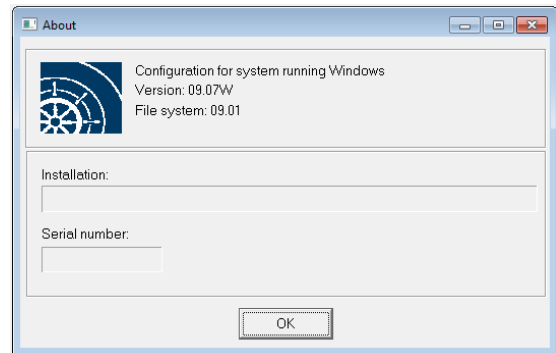
This form gives a number of system information's.

Installation: An identifier entered by the engineer installing the system

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

12.4   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 Mxconfig tool, as soon as a configuration file is opened.

12.4.1   Header field

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

12.4.2   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.

12.4.3   Pop up menu

Working with the telephone directory is primarily done through the use of the right mouse button pop up menu.
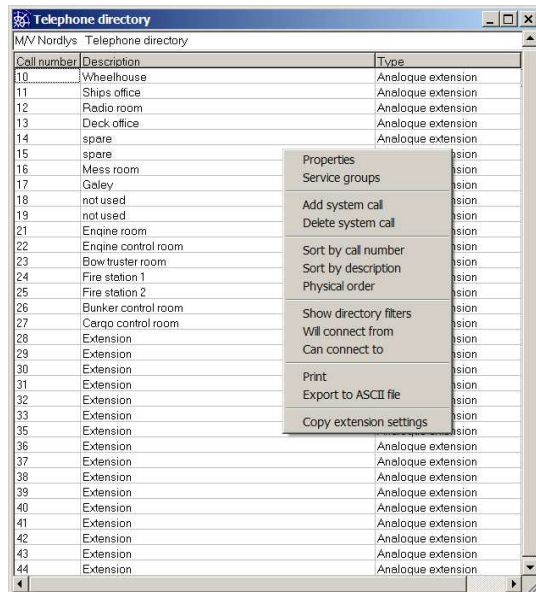
The items of this menu will be described in the following text. Note that the most powerful menu items are protected by the password. The daily user will have access only to a subset of the menu.

12.4.3.1   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.

12.4.3.2   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.

12.4.3.3   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.

12.4.3.4   Delete system call

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

12.4.3.5   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.

12.4.3.6   Sort by call number

This menu will sort the telephone directory by call number

12.4.3.7   Sort by description

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

12.4.3.8   Physical order

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

12.4.3.9   Show includes in mode file changes

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

12.4.3.10　Filtering the telephone directory

A versatile tool for filtering the telephone directory is available.

12.4.3.11　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.

12.4.3.12　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.

12.4.3.13　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.

12.4.3.14　Print

The print menu will send the currently displayed telephone directory to the printer. Note that it is always what is currently displayed in the directory that will be printed. This means that filtering can be used to make printouts of selected call numbers.
The header of the printed pages can be selected by the user by entering text in the header filed found above the telephoned directory grind.

12.4.3.15　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:

12.4.3.16　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.

12.5 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.

12.5.1 Basic analogue extension

This editor shows like:
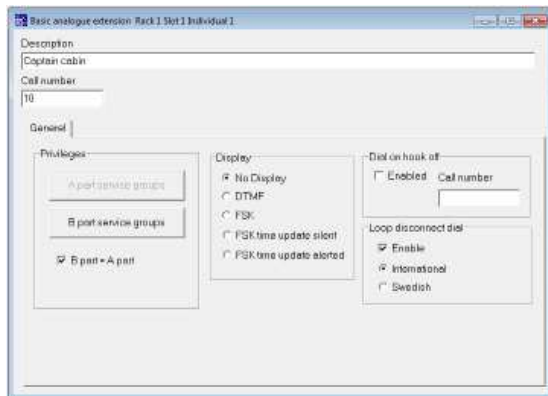
12.5.1.1   Call number / Description

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.

12.5.1.2   Equipment type

Not used.

12.5.1.3   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

12.5.1.4   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.

12.5.1.5   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.

12.5.1.6   No hook detect while ringing

Normally not enabled only If the system has to drive heavy loaded extension lines (6 – 10 telephone sets or 2 – 5 old bell type telephone sets

12.5.2 SeaCom 325

Our digital telephone does have its own individual type.:

12.5.2.1   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.

12.5.2.2 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

12.5.2.3 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.

12.5.2.4 No direct-in

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

12.5.3 SeaCom 211 Talk-back station

This individual type is used for the SeaCom 211 station.

12.5.3.1 Call number / Description

These items are standard individual items found on all individuals

12.5.3.2 Dial on PTT and buttons pressed

The SeaCom 211 unit 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.

12.5.3.3 No hook detect while ringing

Not used.

12.5.4 SeaCom 411 and 421 stations

These individual types covers the stations SeaCom 411 and SeaCom 421.
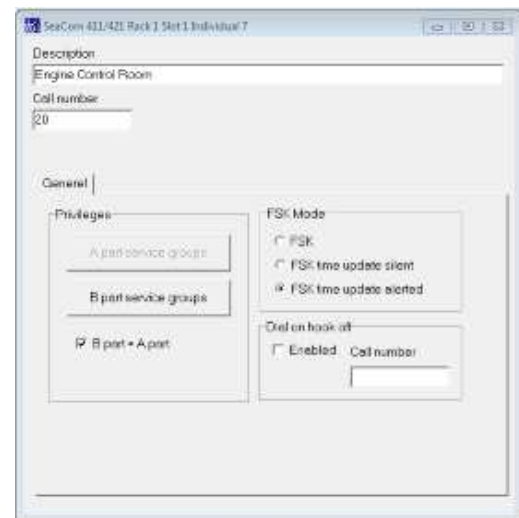
12.5.4.1 Call number / Description

These items are standard individual items found on all SeaCom individuals

12.5.4.2 FSK Mode

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

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

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

12.5.4.3 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.
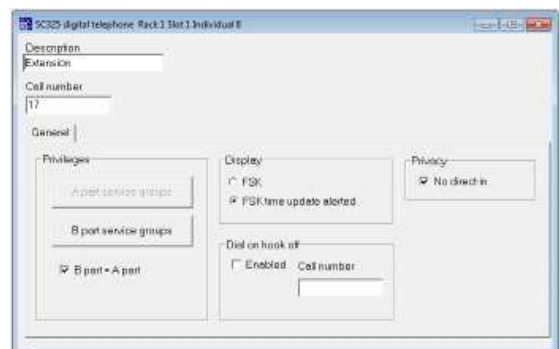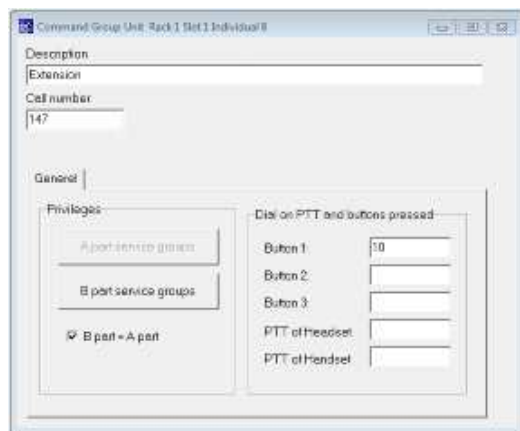
12.5.5 Analogue trunk

This is an individual of the FIO2 board.
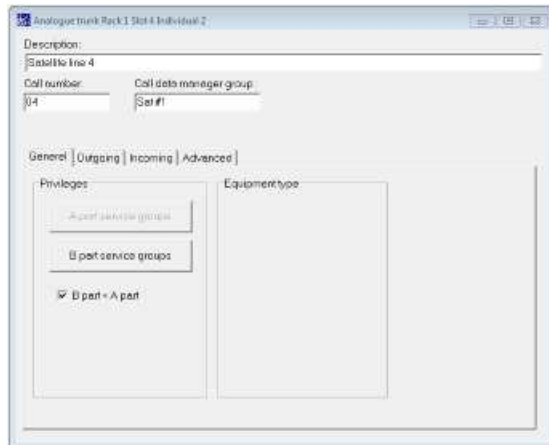
12.5.5.1 Call number / Description

These items are standard individual items found on all individuals

12.5.5.2 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. Refer to the Call Data Manager manual chapter 13.

12.5.5.3 Incoming calls handling

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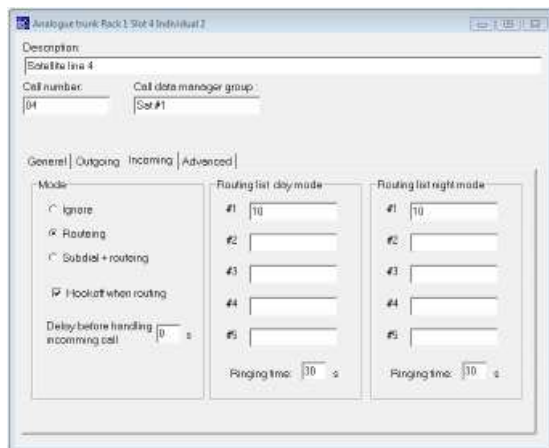
12.5.5.3.1 Incoming mode

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.

12.5.5.3.2 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.

12.5.5.3.3 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.

12.5.5.3.4 Incoming routing list

5 call numbers can be entered into the Incoming routing list. 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.

12.5.5.4 Outgoing calls

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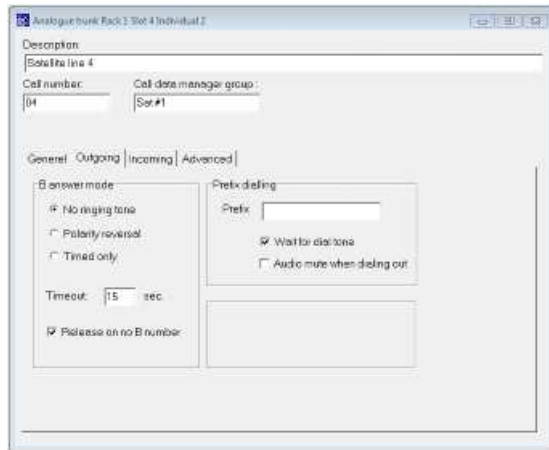
12.5.5.4.1   Prefix dialing

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.

12.5.5.4.2   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.

12.5.5.4.3   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.

12.5.5.5   Advanced options

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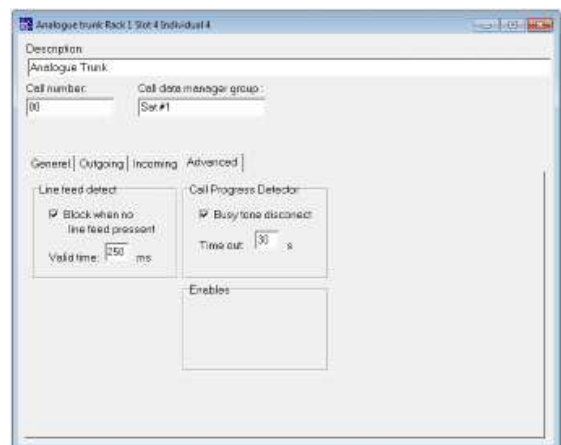
12.5.5.5.1   Line feed detector

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".

12.5.5.5.2   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.

12.5.5.5.3   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.

12.5.6   Audio In/Out
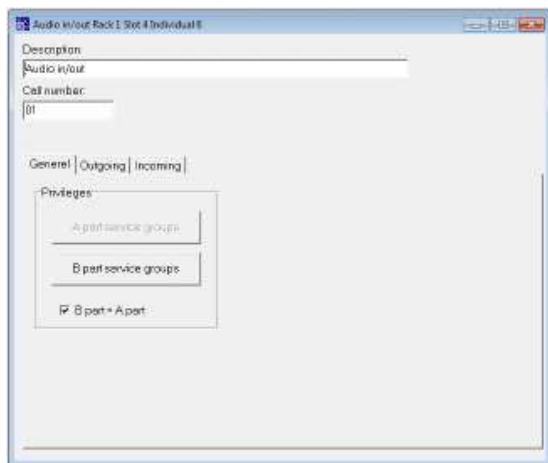
12.5.6.1   Call number / Description

These items are standard individual items found on all SeaCom 2000 individuals

12.5.6.2   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.
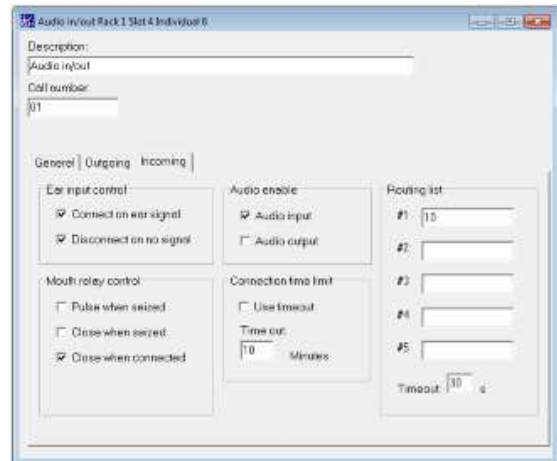
12.5.6.2.1   Ear input control

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.

12.5.6.2.2   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.

12.5.6.2.3   Audio enable

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

12.5.6.2.4   Connection time limit

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

12.5.6.2.5   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.

12.5.6.3   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.

12.5.6.3.1   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.

12.5.6.3.2   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.

12.5.6.3.3   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.

12.5.6.3.4   Connection time limit

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

12.5.7   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 SeaCom 411, SeaCom 421 and SeaCom 211 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.
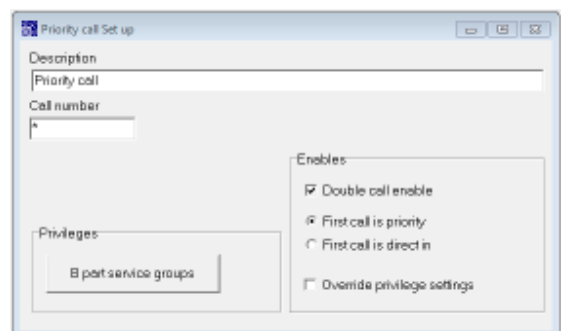
12.5.7.1   Call number / Description

These items are standard individual items found on all individuals

12.5.7.2   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.

12.5.8 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.

12.5.8.1 Call number / Description

These items are standard individual items found on all individuals

12.5.8.2 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)

12.5.8.3 Collect and append

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

12.5.8.4 Terminate by dialing

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

12.5.9 New dial tone

The standard dial system 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.

12.5.9.1 Call number / Description

These items are standard individual items found on all individuals

12.5.10 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.
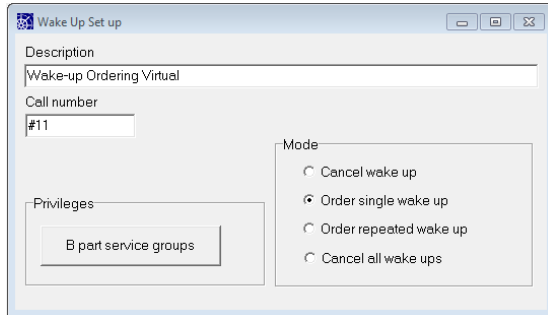

12.5.10.1 Call number / Description

These items are standard individual items found on all individuals

12.5.10.2 Call number list

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

12.5.11   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.

12.5.11.1   Call number / Description

These items are standard individual items found on all individuals

12.5.11.2   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 connected to the SeaCom system to be cancelled.

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

12.5.12   User account and PIN-code checker

When the Call Data Management system is to be used for implementing the user account and PIN code dialing on outgoing lines, a User account and PIN-code checker system must be installed.
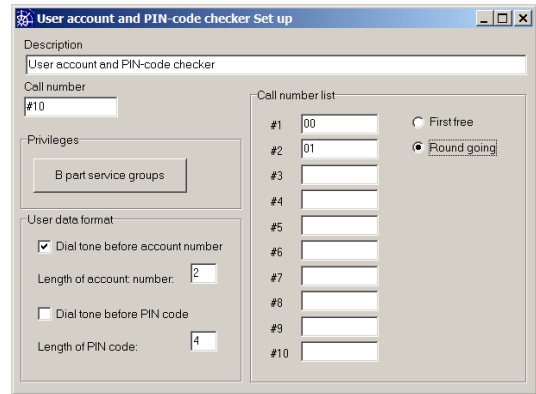
The dial sequence when using this type of outgoing call is:

- Dial the User Access System (default call number *#10*)
- Dial Your user account number (typically 2 digits)
- Dial Your PIN code (typically 4 digits)
- Access is given to one of 10 trunk lines

12.5.12.1   Call number / Description

These items are standard individual items found on all individuals

12.5.12.2   User data format

This window hold information of how many digits shall be collected for the user account number and the user pin code. The lengths must match the lengths used in the user data base. Refer to chapter 13 describing the CDM software. Check the "Dial tone before   --- " check box in order to present the dialer to a dial tone before either the account number and / or the PIN code.

12.5.12.3   Call number list

This window contains a list of 10 call numbers to which the CDM system shall route the call. Whenever a user has dialed an account number and a PIN code that is checked positively against the user data base, the list will come into use as a list of trunk lines, extensions etc. that the user is now given access to.

The use of this list can either be first free or it can be round going. First free can be used when a kind of least cost routing is desired, placing the call numbers of the cheapest trunk line at the top position.

12.5.13   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.

12.5.13.1   Call number / Description

These items are standard individual items found on all individuals

12.5.13.2   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.

12.5.14   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.

12.5.14.1   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.

12.5.14.2   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.

12.5.14.3   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.

12.5.15   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.
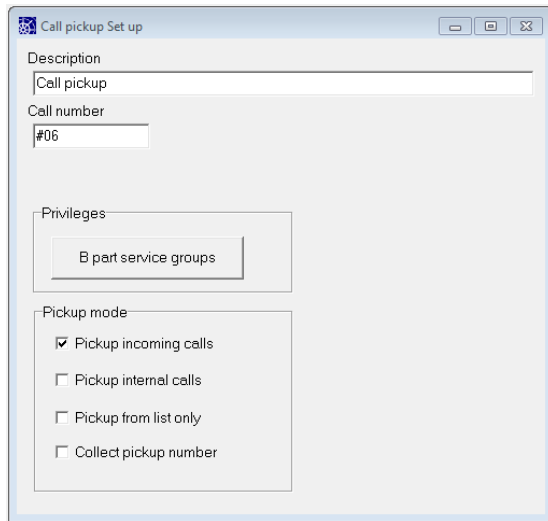
12.5.15.1   Call number / Description

These items are standard individual items found on all individuals

12.5.15.2   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.

12.5.16   Paging call

Paging call is used when activating the SeaCom 211, SeaCom 411 and SeaCom 421 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.

12.5.16.1   Call number / Description

These items are standard individual items found on all individuals

12.5.16.2   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.
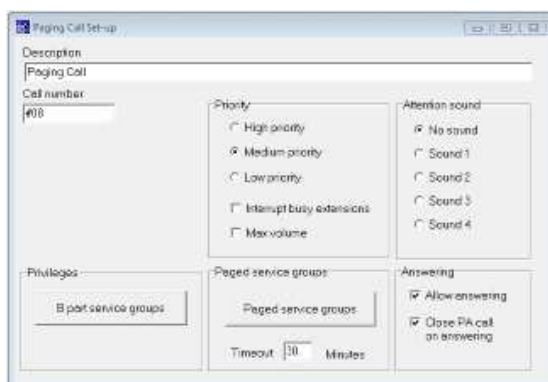
In order for simplicity, one page of the service group editor is devoted paging and alarm calls service groups. Choosing this page for implementing the paging groups is recommended although not necessary.

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

12.5.16.3   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.

12.5.16.4   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.

12.5.16.5   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.

12.5.16.6   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.

12.5.17   Semi-duplex conference

This system call number is used in order to implement a talk-back connection with one or more SeaCom 211 stations.
The full talk-back functionality is obtained when a SeaCom 411 is used as the initiator of the conference, as the MIC button of this type of stations can be used for controlling the speech direction.

12.5.17.1   Call number / Description

These items are standard individual items found on all individuals

12.5.17.2   Conference master

In a talk-back set up with a SeaCom 411 as master and one or more SeaCom 211 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 SeaCom 411 master station.

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

12.5.17.3   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 #.

12.5.17.4   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.

12.5.18   Alarm distribution call

An alarm call system call number is used when activating the SeaCom 211, SeaCom 411 and SeaCom 421 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.
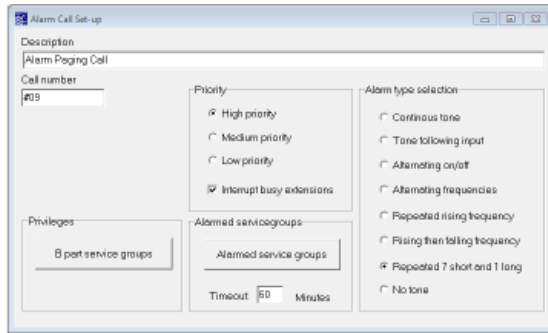
12.5.18.1   Call number / Description

These items are standard individual items found on all individuals

12.5.18.2   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.

12.5.18.3   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

### 12.5.18.4   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.

12.5.18.5   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.

12.5.19   Mode select system call

The mode of the system will be selected by a user calling a mode select system call number.
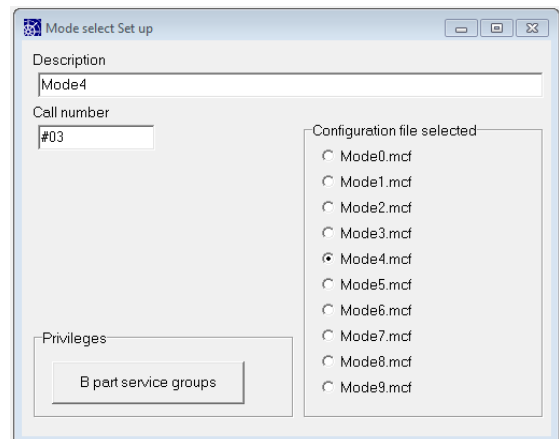
Selecting the mode of the system is a very power-full thing to do. When a user calls the mode select system call, the chosen mode configuration file will be copied into the operating mx.mcf file. After the call, the system waits for all conversations and connections to be terminated. When no connections 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.
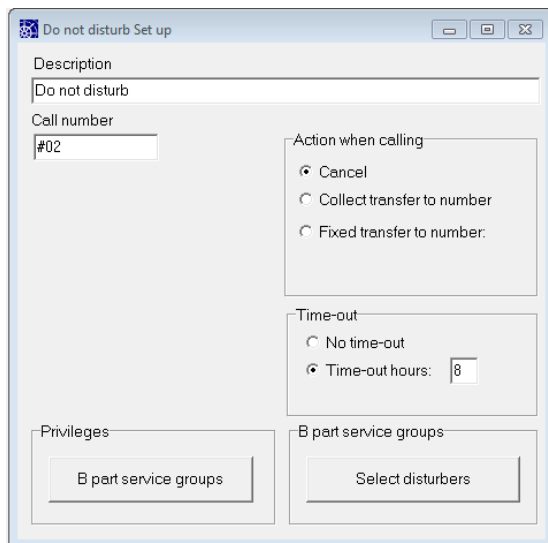
12.5.19.1   Call number / Description

These items are standard individual items found on all individuals

12.5.19.2   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.

12.5.20   '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.

12.5.20.1   Call number / Description

These items are standard individual items found on all individuals

12.5.20.2   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.

12.5.20.3   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.

12.5.20.4   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.

12.5.21   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.
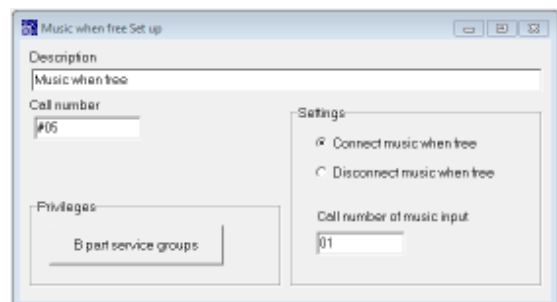
12.5.21.1   Call number / Description

These items are standard individual items found on all individuals

12.5.21.2   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 of a music input.

12.5.22   Direct-in request

Stations having the possibility of operating in hands-free can make use of this system call number to order hands-free hookof to conversation when a call comes in.
The SeaCom 411 and 421 stations and the SeaCom 325 do have this ability.

12.5.22.1   Call number / Description

These items are standard individual items found on all individuals

12.5.22.2   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 telephone calls.

12.5.23   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

12.5.23.1   Action when calling

The system call can be used for 3 purposes:

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

12.5.24   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.

12.5.24.1   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.

12.5.24.2   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.

12.5.24.3   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.

12.5.24.4   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.

13. Call Data Manager

This chapter covers the Call Data Management system (CDM). This system is used for registering and billing of calls via trunks lines.

13.1 General

The Call Data Manager is used to maintain the user database of the SeaCom system. In order for the system to be able to control access to, and to keep track of the use of outgoing telephone lines, a database of authorized users is found on the system disk.

The CDM application enables the user to inspect the call data records stored by the system, and to print the call data either as a simple print out or as an invoice printing. Exporting data to ASCII files is also possible.

The CDM application is used to maintain the charge rate database. The charge rates can be set for each type of outgoing line (trunk group) and a possibility for setting the rates depending on the destination of the telephone calls (B number) is given.

13.2 The client/server concept

The SeaCom system keeps the user and call data database on its disk, and it acts as the server of the database system.

The CDM application is meant to operate as a client on a network computer, typically located on the bridge or in the ships office.
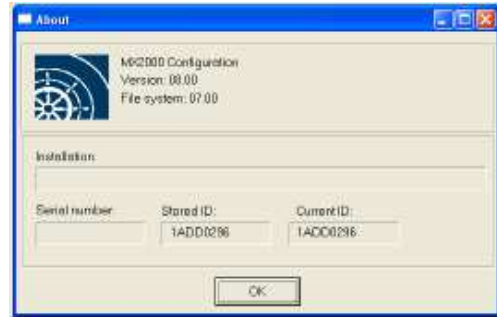
13.3 Installing the CDM

To bring the Call Data Manager in operation, the below steps must be carried out:

- Make a network connection between the SeaCom system and the administrative computer
- Activate the CDM engine on the SeaCom system.
- Install the CDM system on the administrative computer
- Install a User account and PIN-code checker system call number.

### 13.3.1 Activating the CDM engine

The CDM engine on the SeaCom system is already installed, and need only to be activated.

Open the system configuration, and use the about menu to show the about box

In the about box, read out the Stored ID and send these digits to SeaCom witch will return the activation key.

Use the password menu to enter the activation code. When the code is entered right, the below information will show up:
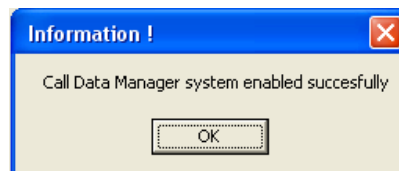
13.3.2 Installing the client software

The CDM software operating on the administrative client computer must be installed, and the connection to the database of the SeaCom system must be established.

To do this follow the below steps:

- Download the CDM install software from the SeaCom homepage and execute it.
- Use the windows explorer to locate and connect the disk of the SeaCom system (referenced as: system disk)
- Map the shared folder MX_DataBase of the system disk onto the client computer as the M: drive
- Use the START / PROGRAMS / BORLAND C++ / BDE Configuration to start the Data base engine configuration.
- In the BDE Configuration, chose the Aliases tab page
- Select the "MX_DataBase" alias
- In the Parameters window, set the PATH to M:
- In the BDE configuration, select the Drivers tab page
- Select the PARADOX driver from the Driver name list and set the NET DIR to M:
- Close the BDE configuration utility and restart Your computer
- The Call Data Manager shall now be available from the START menu.

13.3.3   Access call number

To use the system, at least one system access call number must be installed. Refer to chapter 12.5.12 describing the User account and PIN-code checker

13.4   Backup

All data, including call data, generated by the CDM system is stored in PARADOX data base files. These files are found on the system disk in the folder MX_DataBase.

It is the responsibility of the network administrator to set-up a kind of daily backup procedure for backing up this folder.

13.5   Firewall

When using the Call Data Manager, the SeaCom system is in network contact with other computers. Although the Windows XP Embedded operating system is rarely affected by virus, it is strongly recommended that the system is connected behind a properly configured firewall.

The SeaCom system does not hold any antivirus software and does not have an activated firewall itself.

It is the responsibility of the installer to make sure that the network security is properly set up.

13.6   General concepts

This chapter contains descriptions of basic concepts.

13.6.1   Users

Persons allowed to make outgoing call via the Call Data Management system is called users. Before conducting any telephone calls, the user must be entered into the user data base of the CDM system. This is done using the user editor.

13.6.2   Account number

All users are referenced by an account number. Typically the account number has a length of 2 digits, giving a total number of 100 users.
The length of the account number can be set longer if more users are needed.

13.6.3   PIN code

All users must keep their own self chosen PIN code. After installation of a new user into the user database, the PIN code is reset as "new". The first time the user makes an outgoing call using the account number, a self chosen PIN

code must be entered. This PIN code is saved in an encrypted format into the user database. The same PIN code must now be entered every time the user makes telephone calls. The length of the PIN code is selected upon the installation. Typically 4 digits.

13.6.4   Processing of a call

When a user wants to use an outgoing line for a telephone call, the dial sequence indicated below should be followed:

Dial '#10'

- CDM system access started

Dial the account number

- The account is checked

Dial the PIN code

- The PIN code is collected and checked
- The call is transferred to the trunk line

Dial out on the trunk

- Call going on

Hang up

- The system stores a call data record on the system disk

13.6.5   Call data records

When a telephone call is completed, the system creates a call data record, which is appended to the call data record table in the database. This call data record will contain:

- the date and time of the call
- the duration
- the destination number,
- the user account number
- charge, calculated using the charges table of the database.

13.6.6   Marking call data as invoiced

A new call data record will contain a mark saying "Non-invoiced". Whenever an invoice is printed, the operator will be prompted: "Do you want to mark call data invoiced". When answering "Yes", all printed call data will have their mark set to "Invoiced", and the accounts of the user invoiced will be zero set.

13.7   The CDM software

On the administrative computer, the CDM administration system will be started and used by the system administrator. This chapter describes the use of this software.

The CDM system holds 4 editors:

User data editor

- entering, printing and inspecting data of users = account holders.

Call data viewer

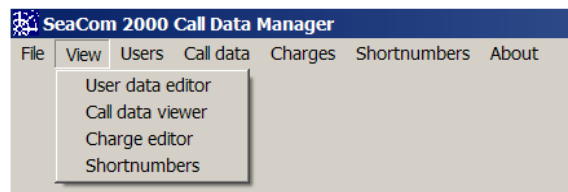- inspecting data of conducted calls

Charge editor

- entering, editing and inspecting the charges of the system

Short number

- inspecting and editing the short number list

13.7.1   The user editor

The user editor is opened by default when the application opens, whereas the call data viewer and the charge editor can be opened by the use of the View menu.

The call data viewer can be opened by double clicking a user in the user data editor, and all editors are opened whenever a menu operating on data covered by the editor is selected from the main menu.

13.7.1.1   Users data editor

The user data is inspected by the User data editor window. There is a right mouse pop up menu, or menus can be chosen from the main menu.
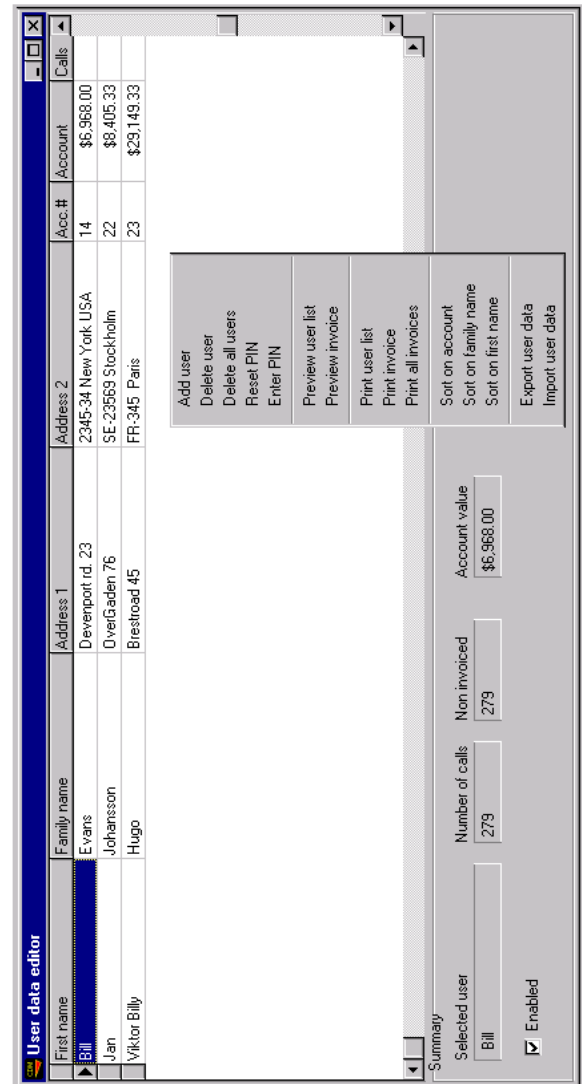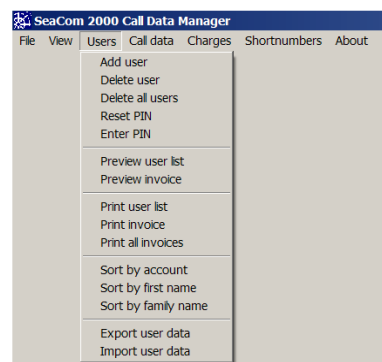
The main part of the user data editor contains the user data display. Navigating into the user data display is carried out by mouse clicking or pressing the arrow keys. When a user is selected, the account number of the user is propagated to the call data viewer so that this viewer will display call data of the user. The data cannot be edited directly in the string grid

13.7.1.1.1   User data

4 strings of 25 characters can freely be used for describing the user..

13.7.1.1.2   Account number

The account number of the user is displayed in the Acc# column. Be careful to enter only unique account numbers and to keep the length matching the selected account numbers lengths. The default length is 2 digits.

13.7.1.1.3   Account

The Account column displays the sum of charges of non-invoiced calls for the user.

13.7.1.1.4   Calls

The Calls column displays how many calls in total the user has made.

13.7.1.1.5   Non-invoiced

The Non-invoiced column shows the number of non-invoiced calls. The sum of charges of these calls makes up the account value.

13.7.1.1.6   The summary window

The summary window contains the "Selected user" field showing the currently selected user together with the most important data of the user.

13.7.1.2   The users menu

The Users menu can either be found on the top main menu line or by right clicking the User data editor window. User data is operated by selecting among the menu items.

13.7.1.2.1   Add user

Select the "Add user" menu if a new user is to be entered.

The account number can be selected freely, but when the form pops up, the system scans the data base for a free account number. It is not possible to use an account number that is already in use.

The 4strings: First name, Family name, Address1, Address 2 is used freely to describe the user. The Personal ID code is meant to hold the social security number of users, or

any other global unique person identifier. This ID code is stored with each call data record.

13.7.1.2.2   Deleting users

The currently selected user can be deleted by choosing the "Delete user" menu. If any call data of the user still exist in the database, the user can not be deleted, and you have to delete all call data records of the user before a delete attempt will be successful.

13.7.1.2.3   Delete all users

All users can be deleted by selecting this menu item. Only users having no call data will be deleted.

13.7.1.2.4   Reset PIN

When a user is just entered, the pin code will be reset to "new". On the first attempt by the user to make an outgoing call, the used PIN code will be stored in the user database.

13.7.1.2.5   Enter PIN

If the PIN code has to be selected during installation of a user, it is possible to enter the PIN code by using this menu.
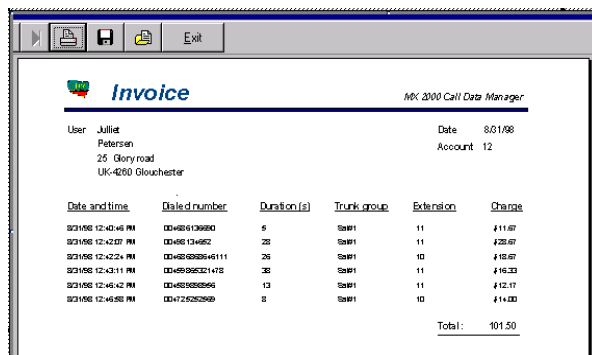
13.7.1.2.6   Previewing and printing of user list

A list of users can be previewed/printed by choosing the menu items "Preview user list" and "Print user list".

13.7.1.2.7   Printing invoices of all users

Use the menu "Print all invoices" when all existing users are to be invoiced. A loop printing invoices for all users one by one will be entered, after the last user, the operator will be asked if all users should be set as "invoiced", if the answer is yes the CDM will mark all call data "invoiced", and set all user accounts to zero.

13.7.1.2.8   Sorting user data

Sorting the user data is made by clicking the: Sort by account, Sort by first name and Sort by family name menus.
Sorting will affect the User data display and the printouts selected.

13.7.1.2.9   Export user data

This menu is used to initiate an export of user data to an ASCII file.
Records of the file is separated by CR character and fields of the records are separated by a TAB character. This allows for easy import to applications like Excel e.t.c.

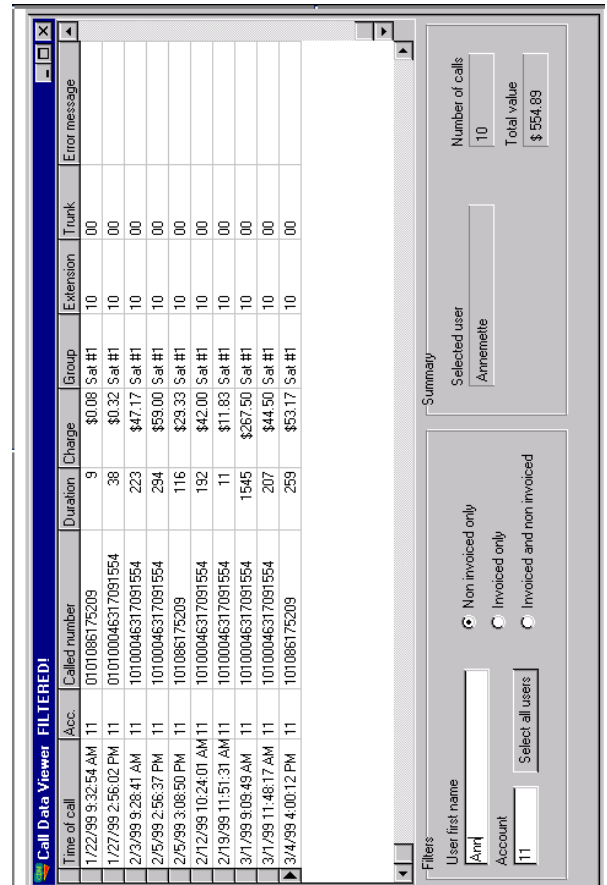The records of such an ASCII file holds the following fields:

- Account number
- First name
- Family name
- Address 1
- Address 2
- PIN code (encrypted)
- Account value
- Enabled Boolean
- Card number   (not used)'
- Number of calls
- Number of non invoiced calls
- Personal ID code
- Auxiliary field 1   (not used)
- Auxiliary field 2   (not used)
- Auxiliary account value (not used)

13.7.1.2.10   Import user data

User data can be imported from a file having the above format. Note that the account will be zero set during import.

13.7.2   The Call Data Viewer

The call data is inspected by the Call data viewer window.

The Call Data Viewer is opened by using the View menu from the main menu or by double clicking a user in the list

13.7.2.1   Time of call

Displays the data and time the call was started. The format of the date/time is determined by the set-up of the operating system.

13.7.2.2   Acc.

This is the account number of the user that made the call. No other reference to the user is stored.

13.7.2.3   Called number

This is the destination number (B number) of the call. The full B number is displayed.

13.7.2.4   Duration

The duration of the call in seconds. The duration is shown in seconds, although charge rates are entered in minutes.

13.7.2.5   Charge

This is the price of the call calculated when the call was terminated based on the trunk group, the B number and the duration together with the charge rates set-up by the user using the Charge Editor.

13.7.2.6   Group

The trunk group through which the call was made. This corresponds to the trunk group set up in the Trunk set up.

13.7.2.7   Extension

The local telephone number of the telephone from which the call was made.

13.7.2.8   Trunk

The call number of the trunk through which the call was made.

13.7.2.9   Error message

This field will be empty if no errors exist. If a an error is encountered during the attempt by the processor to append a call data record, this field will contain an error text.

13.7.2.10   Invoiced

The field will show false if the record is not marked invoiced, and will display true if the record is marked invoiced.

13.7.2.11   Selecting the users data

When selecting a user in the User Data Editor, the account number of the user is transported to the Account item of the filter window, thereby selecting call data of the selected user.
This means that when navigating through users in the User Data Editor, the Call Data Viewer will display call data of the currently selected user.

If this simple direct method of selecting which call data to view is not satisfying, the filter window described below will give some possibilities.

13.7.2.12   The filter window

The filter window contains 3 items helping the user to restrict the number of call data displayed.

13.7.2.12.1   User name filtering

By entering the first name of a user, into the "User first name" edit box, the account number of that user will be transferred to the "Account" edit box which in turn is used as the filter for the displayed call data.

Altering the "User first name" box is the same as altering the "First name" box of the User Data Editor.

13.7.2.12.2   Account filtering

If you wish to display call data for a known account number, then this can be entered directly into the "Account" edit box of the filter window.

13.7.2.12.3   Selecting all users

Press the "Select all users" if call data of all users are to be inspected.

13.7.2.12.4   Invoiced filtering

3 radio buttons on the filter window selects either all call data (Invoiced and non-invoiced) or all invoiced call data or all non-invoiced call data.

13.7.2.13   The summary window

The summary window contains 3 items.

13.7.2.13.1   Selected user

The leftmost item shows the currently selected user, or it shows an informative string whenever no single user is selected, or whenever an account filter is entered manually.

13.7.2.13.2   Totals display

The "Number of calls" box shows the number of calls displayed in the call data viewer string grid. If any filtering is included, this number will change always to reflect the number of calls to be seen in the display or printed if call data printing is selected.

The "Total value" read only item shows the sum of all call data records displayed in the call data display. Invoiced or non-invoiced, single user or many users. Always the total sum of charges.

13.7.2.14   Operating the call data

Operating the call data is carried out by means of the menu items.

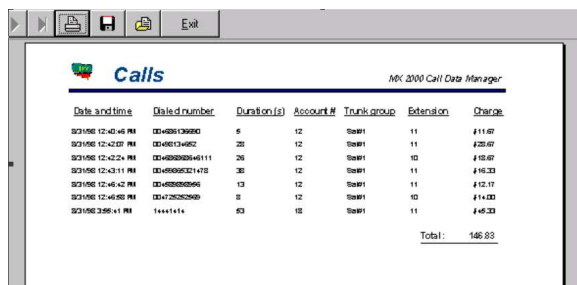### 13.7.2.14.1   Deleting call data

Call data can be deleted by using the "Delete selected call data". Choosing this menu item

will delete all call data displayed in the call data display. Be sure to apply the appropriate filtering before pressing the delete button.

Deleting call data is an important operation, as this is the only way to prevent the system disk from running out of disk space. The person responsible for the maintenance of call data must follow a strategy of, as an example, deleting call data every 3 months.

13.7.2.14.2   Previewing and printing of call data

Two menus "Preview call data" and "Print call data" implement the possibility of printing and previewing call data. The print out will contain exact the call data displayed in the call data display, so the right filtering must be applied before printing.
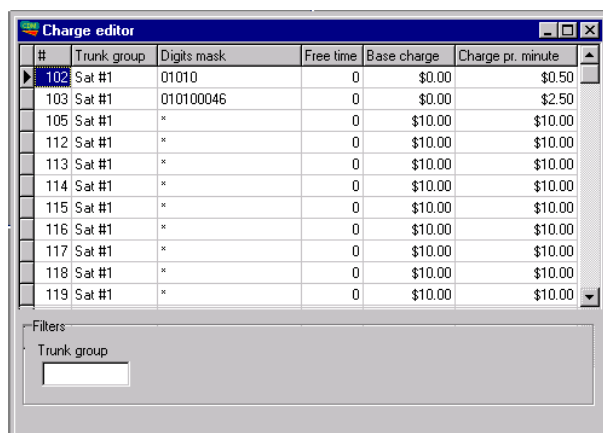

13.7.3   Charge editor

The prices used for calculating cost of telephone calls are put into the system using the charge editor.
The rates are base upon:

- Trunk group (i.e. the type of line used. Satellite, Cellular, shore line etc.)
- The B number
- The call duration.

The editor has a right mouse button pop up menu or the menus can be selected from the main menu line.

13.7.3.1   Trunk group

The trunk group column displays the trunk group for which the record is applicable.

This is the first data tested by system while trying to calculate the price of the calls. If no match between the trunk group actually used and any charge records found in the database, then an error exists, the price will be set to zero and an error message will be inserted into the call data record.

Ensure that the trunk group names entered in the Advanced Analogue Trunk set-up of the c can also be found in the charges database.

13.7.3.2   Digits mask

When, while calculating the price of a call, then the B digits are checked against the "Digits mask" of the record.

The digit masks states the digits that must be equal between the B number and the mask in order to apply the rates of the record. An example.

B number: 004586136690
Mask: 0045*** will match

B number: 004631480668
Mask: 0045   no match

B number   004586136690
Mask   0045**86   will match

As seen from the example, the '*' character can be used as "don't care" digits.

Be sure always to include a default mask = '*' and default rates for all trunk groups.

13.7.3.3   Free time

This field sets the minimum duration of a charged call, and sets the number of free seconds.
Calls having a duration less than the free time will not be charged (but still logged), and call having a duration that exceeds the free time will have the free time subtracted before the price is calculated on the basis of the following two rates.

13.7.3.4   Base charge and Charge pr. Minute

Call having a duration that exceeds the free time will have a price calculated as:

Charge = Base charge + ((Duration of call - Free Time) * Charge pr. Minute) / 60

13.7.4 Short number editor

Dialing out via the CDM system can be assisted by a short number conversion table. This table is edited using the short number editor.
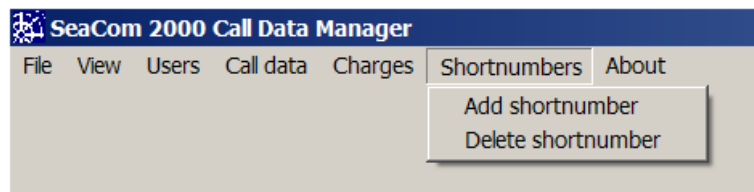
The short number editor has two columns, the first column showing the short numbers and the second showing the digits to be dialed out when the short number is used. Figure 14 shows an example. Note that the short numbers are shown with both the # terminator and without. If both dial timeout and the # end key has to be used, this must be the case for each short number.

14. CP and its applications

This chapter describes the contents of the system disk located on board the CP.

The CP runs a Windows XP Embedded operating system, which from the user point of view is a standard windows system but with a substantial reduced set of features.

14.1 Contents of the system folder

The folder SeaCom2000 will in the following be referred to as the system folder.

The content of the system folder is seen below:

Name ▼	Size	Type
vncviewer	174 KB	Application
vnc-3.3.7-x86_win32	564 KB	Application
USB_Update_Agent	248 KB	Application
NMEA_TimeReceiver	256 KB	Application
NMEA_ComConfig	1 KB	Text Document
MXprocesOperating	1.009 KB	Application
MXproces	1.009 KB	Application
MXpref	2 KB	CSF File
MXconfig	1.006 KB	Application
MX_Process_Log	4 KB	Text Document
MX_Booter	238 KB	Application
mx.mcf	831 KB	MCF File
mx.mbk	831 KB	MBK File
FIO2_01-6R	119 KB	A07 File
AEXT8_05-3R	298 KB	A07 File
AEXT8_04-5R	287 KB	A07 File
AEXT8_05-5R	298 KB	A07 File

vncviewer.exe
This file is an remote desktop application, using which the display of the CP board can be viewed.

vnc-3.3.7-x86_win32.exe
The install file for the vncviewer. It is not to be used by the user.

USB_Update_Agent.exe
A system application only.
Taking care of files when the USB stick programming is used.

NMEA_TimeReceiver.exe
A system application only.
The application receiving NMEA data from the serial port used for setting the real time clock of the system.

MXprocessOperating.exe
A system application only.
A work copy of the MXprocess.exe.

MXprocess.exe
The software which controls the SeaCom system. This is the very brain of the system.

MXpref.csf
A system file only.

MXconfig.exe
The configuration tool software. This is the file to executein order to edit the system configuration file: mx.mcf.

MX_Process_Log.txt
A system log file. This is a text file into which the system logs startup events, errors and warnings.

MX_Booter.exe
A system file.
This boot software makes the copy of the MXprocess.exe into MXprocessOperating.exe and launches the copy.

mx.mcf
The configuration file. This file stores all call numbers, extension descriptions, access privileges etc.

mx.mbk
A backup copy of the configuration file created by the MXconfig.exe.

*.A07
Executable files for extension and trunk boards.

ModeX.mcf
Configuration data are found in the file named mx.mcf. The name of the file, and its location on the system hard disk must be kept unchanged, in order for the data to be used by the system when operating. The mx.mcf file is open and in operation when the system is working.

By far the most installation makes use of only one configuration file only, but the SeaCom system gives the possibility of changing the file run time, in order to change the behavior of the system, for example depending on time of day (day mode / night mode), depending on crew, depending on being chartered or not, depending on being in dock or in service etc.

Swapping can be made between 10 files of the *.mcf file type, by a simple telephone call to a mode select system call number.

14.2 System installation backup

In order to make a backup of Your installation programming, You need to copy the mx.mcf file onto a backup media.

14.3 Vncviewer.exe

The vncviewer is a remote desktop tool. The SeaCom system provides a running vnc server, and the user can launch the vncviewer.exe in order to connect this server and get the system display visible.
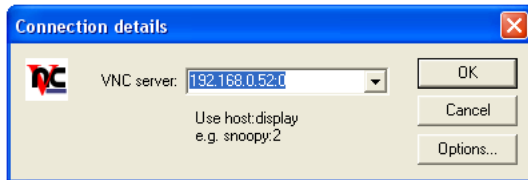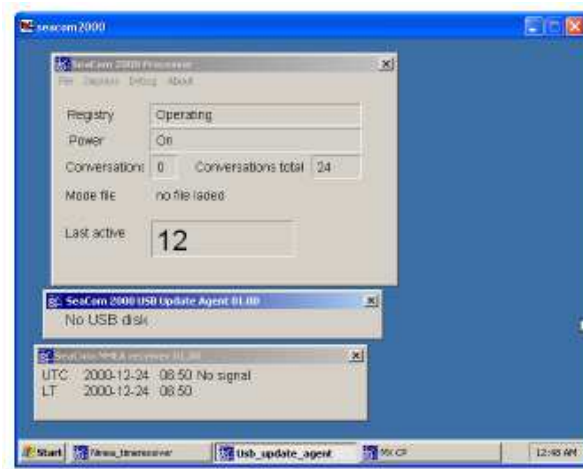

Use the IP address of the system:

192.168.0.52:0

Password:

1017

In order to work, it requires the PC connected to have a fixed IP address set
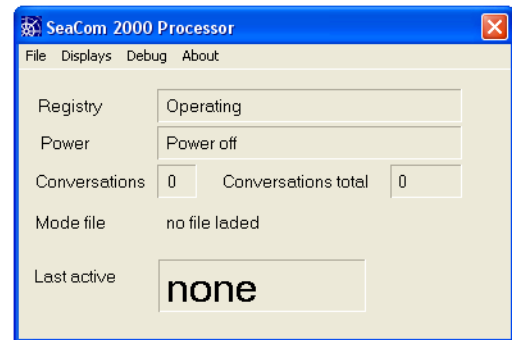
The vncviewer shows a display like:

Using the vncviewer is the only way of getting access to the windows of the on board applications described in the following text.

14.4 MXprocess

This is the software that controls the whole system. It has the below main window:

The main windows has a selection of menus, which by far the most are for software development and debugging.

The display "Last active" is very useful when commissioning a system.

The registry display is very useful when checking that al boards in a system is healthy and working.
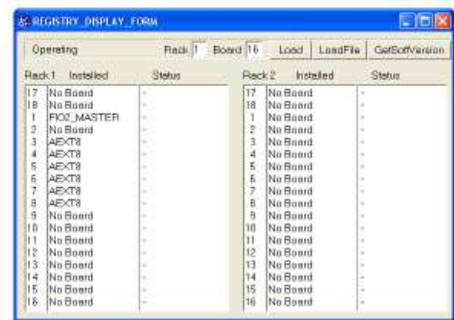
14.5 NMEA_TimeReceiver

This application used COM1 port in order to receive NMEA data from a central clock system.

The main window below is used to verify that data is received.
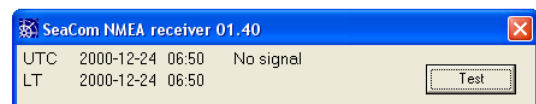
The data is NMEA 183 format. 4800 Baud, 8,N,1

The NMEA string received is the ZQZDA string, which hold UTC time and a local time offset.

$ZQZDA,hhmmss,dd,mm,yyyy,HH,MM*<CS><CR>

When data is received by the NMEA time receiver, the time will be distributed to the FSK display telephones which is set up to subscribe to time updates.

The time is distributed whenever the local time offset changes.

14.6 USB update agent

The USB_Update_Agent is handling the USB access to the configuration file.

This application currently checks if an USB stick is inserted, and if it detects an insertion, it follows the below flow:

bat4.bat file on system disk will be executed
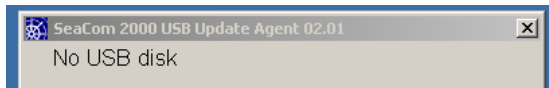**If** mx.mnw file exist on USB key
  then
    bat1.bat file on USB key will be executed
    move mx.mnw from USB to system disk.
    bat2.bat file on USB key will be executed
  else
    bat1.bat file on USB key will be executed
    copy mx.mnw from system disk to USB
    bat2.bat file on USB key will be executed

bat3.bat file on system disk is executed and deleted.

If the batch filed does not exist, the execution is, of course, skipped.

The batch file execution gives the possibility of copying files to/from the system disk.

Executing bat4 is new to CP2 revision 0500. bat4.bat creates a full system backup onto the USB stick. The system backup is a full copy of the contents of the SeaCom2000 folder found on the system disk.
This backup is useful to store whenever a system is commissioned, so both the configuration file but also the corresponding versions of CP software and configuration software are backed up.

The output to the USB stick looks like

15. List of spare parts

This chapter holds stock numbers for the items covered by this manual

Item description	Order nmbr
SeaCom 2100	10-092-0200
PSU2	10-110-1020
CP2	10-110-1011
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
CTU2	10-110-1061
CTU24-8	10-110-1208
CTU24	10-110-1224
FIO2 slave	10-110-1032
Ribbon cable 550mm	20-120-0020
Ribbon cable 650mm	20-120-0021
Ribbon cable 950mm	20-120-0025
PIM	20-110-0060
Plastic card guide	10-130-0000
SC211 talk-back station	10-102-0211
SC220 industrial telephone	10-102-0220
SC411 flush mount intercom	10-102-0411
SC421 wall mount intercom	10-102-0421
TX325 digital telephone	10-400-0050
SC325 digital telephone with PA	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

15. List of spare parts

16. Trouble shooting

This chapter gives some ideas for trouble shooting a faulty system.

When having a problem, it is wise to find out if the problem is global to all telephones and stations or the problem is related to one single extension or station.

Try a number of telephones or stations, and check if the same problem exist for many telephones or it relates only to one.
If the problem is related to one single telephone or station, then continue the fault fining reading paragraph 16.1, otherwise go to 16.3.

16.1 Problems with stations

Problems with stations or telephones can be caused of:

- Power supply missing
- Line or line card (AEXT8) error
- Broken station

Check if power is applied to the SeaCom 211, SeaCom 411 and SeaCom 421 which all require power. Check the fuses inside the exchange system if the station has no power. The 24V DC power to the station shall read 18-32V DC.
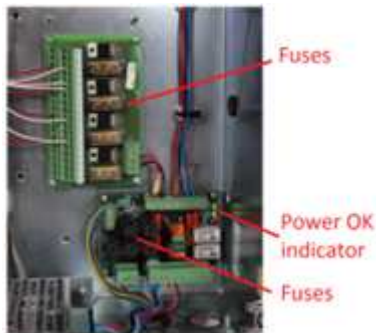
The telephone line from the exchange to the station can be broken or shorted. A good test is to connect an analogue telephone like the TX-250 (refer to 4.2.1) to the telephone line. It shall be possible to make calls to and from such for all station types.

If an analogue telephone is working, the station is broken and should be repaired. The analogue telephone can be left while repairing the station.

If the test with an analogue telephone is negative, then the problem is related to the line or line card of the exchange system. Verify that 45-50V DC is found between the two wires of the telephone line when station is idle. If there is no voltage, the line is broken or shorted.

If the problem is a line card problem, go to the exchange system, and try to swap line cards. Most installations have more than one AEXT8.

Remember to power off system before swapping the boards. There are not programming to be updated when swapping boards.
If the problem follows the extension board, then this should be replaced. While waiting for a new extension board, leave the board out that connects the least important stations.

16.2 Problems with trunk lines

Problems with outgoing calls via trunk lines can be caused by:

- Broken or shorted line
- Equipment connected to trunk line is not turned on or not working correctly
- Trunk line card (FIO2) is broken

The front of the FIO2 trunk line card has two sets of indicators – red and yellow -that gives indication of trunk line activity. The below signals should be verified.

When the line is connected, both indicators shall be off. If the red and the yellow are flashing, the trunk line is either not connected, broken or the equipment connected is off. Verify that the trunk line voltage when the line is idle is 25-50 V DC.

Make an outgoing call using the trunk line. In this case the yellow indicator must show activity.

Make an incoming call to the trunk line. In this case the red indicator shall show activity.

A good test of the trunk line is to connect an analogue telephone like the TX-250 (refer to 4.2.1) to the external equipment. If a call can be made using this telephone, the FIO2 board is faulty and should be replaced.

A good test is to connect the trunk line of the FIO2 to an extension line. This is a local loop back, and can be used to verify that the FIO2 trunk port is working as described above.

16.3 Exchange

The exchange is a complex thing and numerous problems can occur. In order to locate the faulty part, the below step by step start up of the system is recommended.

16.3.1 Step by step start up

Power off the system and slide all circuit boards out, as seen on this picture:
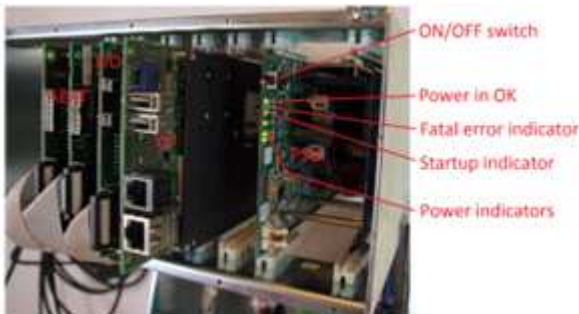

Leave the PSU fully mounted, and power on the system again. At the PIM module, the two green indicators must be on, indicating that the fuses found on the PIM module is OK.

PSU check

Verify that the topmost green indicator of the PSU is on, indicating that power is supplied to the swing frame part of the exchange.

Operate the ON/OFF switch of the PSU, and leave it in the down position (or pressed in on older PSU's). The turn on sequence shall start, where the yellow indicator of the PSU is flashing. After the turn on process all green indicators shall be on, indicating that the PSU is producing all the voltages needed by the system. All voltages is measured and controlled by the PSU, so when green indicator is on the voltage is present and OK. Note that the 80V AC indicator is only turned on shortly, and will normally be off, as it is turned on only when the system needs this voltage which is whenever an extension is hooked off.

If the red indicator of the PSU is on, it indicates a fatal error in the PSU board, and the PSU shall be replaced.

CP check

If the PSU is OK, then slide the CP board fully in and power on the system again.

The PSU shall still show all indicators on, still remembering that the 80V AC will be turned of after the turn on process.

The CP board shall start flashing the green disk indicator, indicating that the processor is starting and reading data from the disk. The flashing continues 1-2 minutes.

When the disk activity indicator stops flashing, a red indicator on the CP board shows flashing. This indicates that the CP processor application is running, trying to contact the FIO2 master, which is not there yet.
If the CP does not reach this state, it shall be replaced, and the backup system configuration installed.

Advanced CP check

More useful information can be obtained if a labtop PC is connected to the CP board, and the VNCviewer is started. Refer to chapter 12.1 and 14.3.

FIO2 master check

Once again power of and now slide in the FIO2 master board.

Power on and await the start sequence to perform. When the red communication indicators of the CP board start to flash, it shall now be accompanied by the green communication indicator, and a complementary set of indicators are found on the FIO2 master.

The alternating red – green flashing indicates that the FIO2 master and the CP board is communicating. If this communication is not established, the FIO2 master board or the CP board must be replaced.

AEXT8 check

Power of and slide one of the extension boards fully in. Restart the system and check that the telephones connected to the extension board is working correctly. If this is not the case for the first extension board, try the next one. Be sure only to have one and only one extension board fully in.

If the telephones are working OK, then slide the remaining extension boards (AEXT8) or trunk line boards (FIO2) in one by one. Each time restarting the system and verifying that it operates OK. This should finally point out the troublesome board. Note that one board with malfunction can corrupt the full system.

If the system is not working with one single extension board, then the FIO2 master have to be replaced, as the communication to the extension boards via backplane is not working.

More useful information can be obtained if a labtop PC is connected to the CP board, and the VNCviewer is started. Refer to chapter 12.1 and 14.3.