OpenCom 100

130
131
150

Mounting and Commissioning

User Guide

AASTRA

Welcome to Aastra

Thank you for choosing this Aastra product. Our product meets the strictest requirements with regard to quality and design.

The following operating instructions will assist you in using your OpenCom 100 and answer most of the questions that may arise.

If you require further technical support or information about other Aastra products, please refer to our website at **http://www.aastra.de** or **http://www.aastra.com**. It provides additional notes and tips on the product.

OpenCom 100 Product Family

This user guide applies to the OpenCom 100 product family comprising the OpenCom 130, OpenCom 131, OpenCom 150, OpenCom 510 and OpenCom X320 systems.

If a reference is made in the text to the OpenCom 100, the description applies to all systems; if the individual characteristics are different, a special note is given.

We hope you enjoy using OpenCom 100.

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1. Features

The OpenCom 100 is a communications system for integrated voice and data communication. The outstanding feature of this communications system is its modular structure:

- The **OpenCom 131** is equipped with all the necessary ports for connecting system telephones, IP system telephones, ISDN terminals and analogue terminals. The system enables Internet/intranet data communication, CTI applications and the system configuration with a standard web browser. Additionally, the OpenCom 131 is equipped with a slot for installing an “M100-AT” interface card. With this interface card, the OpenCom 131 can be connected to analogue trunk lines.

- Even with the smallest **OpenCom 130** version, it is possible to use all the most important communications applications. The basic module enables telephony with system telephones, IP system telephones, SIP system telephones, standard SIP telephones, ISDN telephones and analogue terminals, Internet / intranet data communication, CTI applications, sub-system operation and system configuration using a standard Web browser.

  The expansion module of the OpenCom 130 provides three slots for further interface cards. Using different combinations of interface cards, the configuration of the OpenCom 130 can be tailored exactly to your communications requirements. The need for additional U_{pn} ports supporting DECT, further S_0 ports or more a/b ports can be met using one or more interface cards without changing the system.

- **OpenCom 150** is a system that can be tailored exactly to your communications requirements. Initially, the main module doesn’t carry any telephony interfaces. They can be added by means of additional interface cards. The main module provides five slots for different combinations of interface cards. Unlike the OpenCom 130, the main module of the OpenCom 150 is not divided into basic and expansion modules.

- All named variations of the OpenCom 100 (OpenCom 130 / 131 / 150) offer additionally two special slots:

  – One slot to incorporate a V.24 interface card. This card provides a serial port.

  – One slot to incorporate doorstation equipment interface card.

- You can use an additional insertable memory card (CompactFlash) to operate further programme packages, viz. the OpenVoice and OpenAttendant digital voice memory and voice information systems.

- The special interface card M100-AT4 enables connecting analogue exchange lines. Additional information can be found in the user manual “M100-AT4 Interface Card”.

- The additional S_{2M} connector module allows you to operate an OpenCom 130 / 150 on a primary rate access.

Telephony
The OpenCom 100 communications system is designed to be connected to an ISDN basic access using the DSS1 protocol. System access (point-to-point) and multi-terminal access (point-to-multipoint) are both supported. The two forms of access can be configured in parallel.

For this purpose

- the OpenCom 131 includes two S_0 ports (one external one and one that can be switched between internal/external),

- the OpenCom 130 basic module includes two S_0 ports (one external one and one that can be switched between internal/external),

- the OpenCom 130 expansion module with additional interface cards provides up to eight further S_0 ports (switchable between internal/external). An overview of the available cards can be found under *Interface Cards* starting on page 43,

- the OpenCom 150 with interface cards provides up to 12 S_0 ports (switchable between internal/external). An overview of all possible interface card combinations can be found under *OpenCom 150 (5 Slots)* starting on page 46.

The OpenCom 100 complies with the regulations for telecommunications equipment. The DSS1 protocol is implemented.

The firmware of the OpenCom 100 is designed for configuring up to 300 users.

You can connect the following devices to the OpenCom 100:

- analogue terminals
- Euro-ISDN terminals
- System telephones Aastra 6771 (OpenPhone 71), Aastra 6773 (OpenPhone 73), Aastra 6775 (OpenPhone 75)
- System telephones OpenPhone 61, OpenPhone 63, OpenPhone 65
- RFP 22 and RFP 24 base stations and DECT handsets (via an RFP 22 / 24 base station on the DECT-enabled U_{pn} port of an interface card)

Note: On an OpenCom 131 the U_{pn} ports are not DECT-enabled. This means, DECT base stations and DECT terminals cannot be operated on this communications system.

An S_0 port can be used by Euro-ISDN terminals working in accordance with DSS1. A U_{pn} port is suitable for the Aastra 677x (OpenPhone) range of system telephones. RFP 22 / 24 base stations can also be connected to the DECT-enabled U_{pn} ports on interface cards for an OpenCom 130 and an OpenCom 150. An analogue port is used by standard analogue devices.

If the CNIP (calling name identification presentation) feature is supported by your network provider, the latter will show you the name of callers in addition to their number for each incoming trunk call. The OpenCom 100 supports the display of the name on system telephones. However, if you have created an entry in the telephone book of the OpenCom 100 under the number of the caller, this will be displayed instead.

The OpenCom 100 can be integrated into an existing network (LAN) and be used by all workstations as an Internet access router and mail client.

Configuration and programming of the OpenCom 100 is performed by means of a special Web browser (known as the “Web console”), which can be run on a connected PC.

The OpenCom 100 can also be configured at the customer service centre and maintained by means of remote configuration.

A PC can be connected via a retrofitted V.24 interface card to the COM port for the purpose of configuring the system or transferring connection data.

To connect the OpenCom 100 to existing company hardware, “actor” ports (output) and “sensor” ports (input) can be provided by retrofitting a doorstation interface card. For example, this can be used to operate a door opener and a doorbell via the system (this requires additional equipment).

There are two variants of doorstation equipment interface cards:

- On an OpenCom 130 or an OpenCom 131 a “M100-TFE” interface card can be operated. This card provides two actor ports and three sensor ports to connect doorstation equipment.

- On an OpenCom 150 a “M100-TFE-2” interface card can be operated. This card provides four actor ports and four sensor ports. This card enables to operate two entrance intercom systems, each providing two sensor ports for bell keys.

OpenCom 100 enables you to use CTI (**C**omputer **T**elephony **I**ntegration) applications. The TAPI (Telephony Application Programming Interface) and CSTA (Services for Computer Supported Telecommunications Applications) standards are supported for the purpose of integrating CTI applications. OpenCom 100 also features an integrated browser-based CTI application, the OpenCTI 50. The OpenCTI 50 allow users to call up and use telephone functions from their PCs.

Note: As an alternative to the COM port, call charges can also be transmitted to an external programme for recording charges via the CSTA (Computer Supported Telecommunications Application). There is a corresponding conversion programme available for doing so. This is on the product CD in the “\Aastra” directory. If you desire further information, please contact your representative or Aastra sales.

Cascading

Using the expansion module, the **OpenCom 130** can be cascaded with a second OpenCom 130 communications system. An Ethernet switch on the expansion module further enables applications featuring media convergence, such as the operation of a VoIP interface card M100-IP.

It is possible to cascade two **OpenCom 150** (rack version) units. Two ports of the Ethernet switch residing on the main module can be used externally. Two other ports are available for internal use with interface cards.

With the **OpenCom 131** no PBX cascading with another system is possible. However, PBX networking with a second infocom system is possible.

Further information regarding these configuration options can be found in the chapters *PBX Cascading* starting on page 148 and *PBX Networking* starting on page 153.

Packet Data in the D Channel
Some business applications, for instance POS terminals, cash registers or credit card terminals, require a permanent data connection over the X.25 packet data network. Packet data transfer through the ISDN D channel (according to X.31 via SAPI 16) can also be established between several S_0 interfaces of the OpenCom 100. Simultaneous connections are distinguished by means of a TEI (Terminal Endpoint Identifier).

X.31 packet data can be forwarded between two S_0 interfaces (for instance an internal and external S_0 interface). Equally, data can be forwarded ("routed") over permanent Q.SIG lines. Data can also be routed over an S_{2M} interface. It is possible to operate multiple terminals with the same TEI on different internal S_0 interfaces. A TEI mapping table allows these X.31 connections to be routed to the same external S_0 interface.

The routing table for X.31 packet data is set in the Configurator under **Telephony**: **Extended**: **X.31**. Additional information can be found in the Configurator online help files.

Internet Access
It is possible to connect individual PCs to the OpenCom 100 via the internal S_0 ports, or to connect an entire LAN to the OpenCom 100 via the Ethernet port. These PCs can access the Internet via the OpenCom 100. If Internet access is already available from an Internet service provider, this can be configured in the OpenCom 100. If the client network is not IP-capable, the OpenCom 100 can administer the IP configuration necessary for Internet access. The OpenCom 100 has an integrated DHCP server and a DNS server, which in this case take over IP address administration and name resolution for the client PCs.

The OpenCom 100 enables Internet access for all connected PCs by means of a common IP address. Only this is externally visible. The local IP addresses of the client PCs are translated to the IP address of the OpenCom 100 by network address translation (NAT). In this way the client PCs in the LAN cannot be reached directly from the Internet. This protects them from direct external attack. The LAN is additionally protected by the OpenCom 100 filter lists, which can be customised individually (firewall function).

Note: We recommend you to read through the explanations under *Useful Information on Internet Access* starting on page 99.

E-Mail
The OpenCom 100 has an integrated e-mail function that is able to use the POP3, APOP or IMAP4 protocols to check the Internet service provider for incoming mail. When configuring the OpenCom 100, e-mail account query can be configured for every member of staff. The OpenCom 100 then fetches the incoming e-mail headers (subjects) and senders from the mail server at set intervals, and forwards them to users' system terminal.

E-mail accounts for the sending e-mail can also can be configured for users. E-mails can then, for example, be sent directly from the **OpenCTI 50** to other users. In addition, users who have had a voicebox configured for themselves, can let themselves be notified of new voicebox messages via e-mail.

Important events and errors are kept by the OpenCom 100 in an internal log book: the error store. To inform or alert the system administrators, entries in the log book (system messages) can be sent via e-mail.

Further Network Features
You can offer staff the possibility of dialling into the LAN by means of RAS access.

A LAN-to-LAN link can also be implemented by ISDN. In this way two OpenCom 100s can connect their LANs by dial-in on demand.

A NET CAPI programme (driver software on the CD-ROM) enables you to use ISDN data communication, such as ISDN fax or Eurofile transfer, with workstation PCs which do not have a built-in ISDN card.

Further Telephony Features
Installing an extra memory card allows you to operate a digital voice memory and voice information system. For more information, refer to the user guides called "OpenVoice" and "OpenAttendant".

You can optimise your telephone communication by using the team functions and the call-queuing function.

With an additional licence, the web application "OpenCount" can be used. This web application enables you to register and store telephony connections and evaluate the connections with user defined filters. Further information can be found in the online help of the web console.

As your company's requirements grow, the OpenCom 100 can be networked with other telecom systems. The OpenCom 100 can then operate as a sub-system or DECT server. To operate the system as a DECT server the system's U_{pn} ports must be DECT-capable. It is also possible to create a telecom system with several networked telecom installations.

Voice over IP (VoIP)
The OpenCom 100 supports the connection of VoIP terminals and thereby allow telephony via the existing company network infrastructure. For this purpose, corded system terminals of the type "Aastra 677xip" are available. These devices have the same functionality and support the same features as the non-IP enabled system terminals "Aastra 677x". For users who wish to use PC supported telephony, the IP system terminals are also available as separate licensable software variations (OpenSoftphone). You will find further information in the chapter *Voice over IP (VoIP)* starting on page 102).

DECToverIP®
DECT networking via VoIP is another possible option for offices already extensively using VoIP telephony. The Radio Fixed Parts (RFPs) are connected via network data connections, so they do not occupy any U_{pn} ports and can use existing network connections. With DECToverIP, VoIP protocol data is changed into DECT-compatible voice data direct on the RFPs. DECT-RFPs and DECToverIP-RFP can be used together in combination in many cases; it is however not possible to switch between RFPs using different technologies during a call.

Glossary
Refer to the explanations in the glossary (supplied as a PDF file on the system CD).

2. Factory Settings on Delivery

The following basic settings and features are active on delivery. We recommend that you configure the OpenCom 100 to your individual requirements before putting it into operation (see *Configuration* starting on page 76).

The factory settings apply to smallest version of the OpenCom 131 and OpenCom 130 (which only features the basic module). If an expansion module with interface cards exists, the additional interfaces are initially unconfigured. You must therefore first configure the slots of the expansion module to commission the interfaces.

Note: The OpenCom 150 generally requires the slots to be configured before any interface can be commissioned.

2.1 Telephony Functions

2.1.1 OpenCom 130

- The $S_0$1 port is configured as a multi-terminal connection, and the $S_0$2 port as a system port.
- Aastra 677x system telephones with the telephone numbers 30 to 32 are configured on the three U_{pn} ports.
- Analogue terminals with the telephone numbers 10 to 13 are configured on the four a/b ports.

2.1.2 OpenCom 131

- The $S_0$1 port is configured as a multi-terminal connection, and the $S_0$2 port as a system port.
- Aastra 677x system telephones with the telephone numbers 30 to 32 are configured on the three U_{pn} ports.
- Analogue terminals with the telephone numbers 10 to 17 are configured on the eight a/b ports.

2.1.3 OpenCom 150

With the OpenCom 150 all ports (S_0, U_{pn} and a/b ports) are realised by installing a specific combination of interface cards for this purpose. An overview of available interface cards can be found in the “Interface cards”chapter in the section *OpenCom 150 (5 Slots)* starting on page 46.

- The OpenCom 100 is configured ready for operation in Germany.
- Analogue terminals: The dialling mode (pulse dialling or DTMF) is automatically detected.
- All corded terminals connected to the basic module ring when there are incoming external calls.
- The system PIN, for example for remote-programmable call diversion, is set at “0000”.

2.2 Authorisations

The use of functions by a terminal on the OpenCom 100 is regulated by means of authorisations. Authorisation is configured by means of user groups to which the users with their terminals are then assigned.

Three user groups are preset: “Administrators”, “Standard” and “Guests”. “Administrators” have access to all functions of the OpenCom 100 and unrestricted configuration rights. Users in the “Guests” group cannot configure the OpenCom 100, are not able to make external calls, and have only restricted use of the terminal functions of the OpenCom 100. The “Standard” user group, because of its default settings, is well suited as a starting point for the creation of user groups for normal users of the system (e.g. the staff members of a company).

Note: When the OpenCom 100 is commissioned, all connected terminals are initially in the “Administrators” group until a user logs on to the Web console. Subsequently, all terminals are automatically in the “Guests” group. For more details on the configuration of user groups, refer to the online help in the chapter entitled “User Manager”.

The following functions are delivered preset for user groups:

User group settings

Function / Authorisation	**Standard**	**Adminis-trators**	**Guests**
Applications			
Configurator	personal	Expert	View
Costs	-	+	-
Phone Book	+	+	+
ISP application	-	-	-
Courtesy Service	off	off	off
Phone Book			
Entries (personal)	20	20	0
Edit central	-	+	-
Dial in (outgoing)			
External	Inter-national	Inter-national	Incoming only
Immediate external line seizure	-	-	-
External line seizure over operator	-	-	-
LCR *)	+	+	-
Deactivate LCR *)	+	+	-
LCR at call forwarding to extern. *)	-	-	-
VIP call *)	+	+	-
PIN dial *)	-	-	-
Announcement *)	+	+	-
Intercom *)	+	+	-
Dialout for other phone *)	-	-	-
Instant connection*)	+	+	-
Callback on busy *)	+	+	-

User group settings

Function / Authorisation	Standard	Adminis-trators	Guests
Multiple seizure at the parallel terminal s *)	+	+	+
Switch authorization *)	-	-	-
Dial in (incoming)			
Pickup from group	+	+	-
Pickup selective	+	+	-
Take	+	+	+
Call removal *)	-	-	-
Calling suppression at the parallel terminal *)	-	-	-
Reaction: connection will be disconnected *)	-	-	-
Call queue *)	0	0	0
Call forwarding			
Call forwarding	+	+	-
Call forwarding to extern	+	+	-
Call forwarding of MSN groups	+	+	-
Call forwarding door call	+	+	-
Indicate call forwarding after time parallel *)	+	+	+
Call forwarding for other user *)	-	-	-
Prevent call forwarding by other user *)	-	-	-
Display: Call forwarding via *)	last for-warding	last for-warding	last for-warding
Connection *)			
External to external *)	+	+	-
3-party conference *)	+	+	-
Park call *)	+	+	-

User group settings

Function / Authorisation	Standard	Adminis-trators	Guests
MOH at external connections *)	+	+	+
MOH at internal connections *)	+	+	+
Protection			
Call protection	ringing tone	ringing tone	off
Call waiting protection	+	+	-
Announcement protection *)	-	-	-
Intercom protection *)	-	-	-
Pickup protection *)	-	-	-
Display phone number off (intern) *)	-	-	-
Display phone number off (extern) *)	-	-	-
Display phone number off/on per connection *)	+	+	-
Phone lock *)	+	+	-
Intercept *)	+	+	-
Lists			
Black lists	empty	empty	empty
White lists	empty	empty	empty
Special lists	1	1	1
Call filter	empty	empty	empty
Manage intern call list *)	+	+	-
Manage extern call list *)	+	+	-
Manage busy call list *)	+	+	-
Manage door call lists *)	+	+	-
System phones *)			
All keys locked *)	-	-	+

User group settings

Function / Authorisation	Standard	Adminis-trators	Guests
Programming function keys *)	+	+	+
Menu and ABC keys *)	+	+	+
DECT trunc keys *)	-	-	-
Disconnect ISP connection *)	+	+	-
Connection data *)			
Send incoming connections *)	-	-	-
Send outcoming connections *)	-	-	-
Number of suppressed digits *)	0	0	0
Incoming basic amount *)	0,00	0,00	0,00
Outgoing basic amount *)	0,00	0,00	0,00
Cost factor *)	100%	100%	100%
Network *)			
RAS *)	-	-	-
Callback *)	none	none	none
E-mail notification *)	+	+	-
Send E-mails*)	+	+	-
Other *)			
Speed dialling *)	+	+	-
Door opener *)	+	+	-
Keypad dialling *)	+	+	-
Time control *)	-	-	-
SMS stationary *)	-	-	-
Booking number may be set up*)	+	+	-
Send short messages*)	+	+	-

*) These settings are shown only in the Expert view.

The following important settings are active without further configuration:

- External authorisation: International numbers can be dialled from all configured terminals. External lines must be seized by entering a preset code.
- Call forwarding to internal and external numbers can be activated. Call forwarding after delay is performed after 20 seconds. Door calls and MSN groups can be forwarded. Call diversions for other users and call diversions by other users are deactivated.
- The telephone lock can be activated. The terminal PIN is “0000”.
- The white list, black list and call filters are not preconfigured and therefore not active. If these lists are configured, they can be activated for user groups. A special list of emergency phone numbers is preset and activated.
- The door opener can be opened from all terminals. Door calls can be forwarded.
- Every standard user can change the configuration of OpenCom 100.
- Every standard user can create a personal telephone book and edit entries in the central telephone book.
- Every standard user can read out the charges.
- Applications requiring a licence (e.g. OpenCount) can be used after being activated.
- RAS access is not allowed.
- The multi-company version is not activated.

2.3 Internet Functions

- RAS access (with or without callback) can be set up for every OpenCom 100 user. RAS access requires activation of the RAS authorisation.
- More than one mail account query can be set up for every user.

- Every user with a system terminal can be informed automatically of the receipt of e-mails.
- Users can disconnect existing Internet connections (via the OpenCom 100 Web console and from a system terminal if the function has been configured on that terminal).

The following IP addresses are preset for the network configuration:

- Host name: host
- IP address: 192.168.99.254
- Network mask: 255.255.255.0

The following addresses are transmitted to the client PCs in the LAN via DHCP or PPP:

- Gateway address: 192.168.99.254
- Domain name: domain
- Domain name server: 192.168.99.254
- PPP addresses: 192.168.100.0 to 192.168.100.10
- DHCP addresses: 192.168.99.130 to 192.168.99.169

You can change the IP settings in the **Configurator**. Check with the network administrator responsible for the LAN if you wish to do this.

3. Installation

3.1 Scope of Delivery

The delivery consists of:

- One OpenCom 131 communications system (in a basic version) or OpenCom 130 (in a basic version with a basic module) or OpenCom 150
- One connection cable for the ISDN S_0 port
- One set of mounting screws and wall plugs
- One plug-in power supply (of the TR25240-E-01A13 type) to supply the basic module

  With the OpenCom 130 this power supply is dimensioned to operate the basic module only. Installing the expansion module requires an additional power supply.
- One set of short user guides
- One CD including the complete documentation and software

The OpenCom 130 expansion set consists of:

- One expansion module
- One AC adapter with a connection cable to supply the expansion module with power
- One mounting set with which to install the expansion module and the AC adapter in the OpenCom 130 housing
- One (short) Ethernet connection cable with which to connect the basic module to the expansion module.

3.2 Safety Precautions

The CE symbol on the product confirms that it meets the technical guidelines on user safety and electromagnetic compatibility valid at the time of approval.

Please note: Installation and maintenance should only be performed by specially trained personnel. Always remove the power plug and the plug-in power supply from the mains socket before opening the housing cover and/or connecting devices to the OpenCom 100 ports.

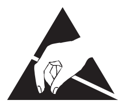
DANGER! **This device contains hazardous voltages. To make the system powerless, remove the power plug and the plug-in power supply from the socket.**

The OpenCom 100 may only be plugged into mains sockets with a protective earth conductor. Mount the OpenCom 100 only close to easily accessible sockets.

Only use the original plug-in power supply: No. 4512699 (TR25240-E-01A13 type) for an OpenCom 131 and for the basic module of an OpenCom 130.

The housing cover may only be opened by authorised personnel. Unauthorised opening of the housing cover and improper repair may damage the OpenCom 100 and invalidate the warrantee.

CAUTION! Static charges can damage the OpenCom 100. Make sure you discharge yourself and your tools before and while installing electrical and electronic components of the OpenCom 100.



Only terminals that deliver safety extra-low voltage (SELV) may be connected to the OpenCom 100. Proper use of authorised terminals meets this requirement.

Only terminals meeting the technical requirements may be connected to the analogue ports. For details, refer to the section entitled *a/b Ports* starting on page 51.

Use a shielded Ethernet cable (STP cable, Shielded Twisted Pair cable) to connect the OpenCom 100 to a Local Area Network (LAN).

Do not allow any fluid to penetrate the OpenCom 100, because this may cause electric shocks or short circuits.

Do not install the OpenCom 100 during a storm. Do not connect or disconnect lines during a storm.

The OpenCom 100 is designed for indoor use only. Lay the cables so that they cannot be walked on or tripped over.

The connection of external devices to the sensor/actor should be performed by a qualified electrician.

3.3 Declarations of Conformity

The OpenCom 130, OpenCom 131 and OpenCom 150 ITC systems conform to the requirements set down in the EU directive 99/5/EC.
The Declaration of Conformity can be viewed at the Aastra Web site at http://www.aastra.de or http://www.aastra.com.

3.4 Mounting Location

The ambient temperature for operating the OpenCom 100 must be between +5 and +40°C. The power supply must be 230 V/50 Hz AC. A separate fuse for the power supply is recommended.
To maintain the prescribed ambient temperature, mount the OpenCom 100 in a well-ventilated location, away from direct sources of heat.

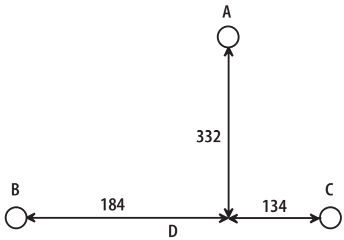
Do not position the OpenCom 100

- in front of or above heat sources such as radiators,
- in direct sunlight,
- behind curtains,
- in small, unventilated, damp rooms,
- on or near inflammable materials,
- or near high-frequency devices such as transmitters, X-ray or similar apparatus.

Use a separate 230 V power circuit and install overvoltage protection.

3.5 Wall Mounting

The OpenCom 100 is mounted on the wall with three screws as shown in this diagram:



Mounting plan

To fasten the screws at points B and C, remove the cover of the OpenCom 100 and insert the screws in the holes provided for this purpose. The OpenCom 100 is suspended from the screw at point A, so there must be a space of 3 mm between the screw and the wall.

3.6 Installing an Expansion Set

This chapter is addressed to customers operating an OpenCom 130 system and who wish to equip their infocom system with multiple interface cards for extra ports.

With the OpenCom 150 and OpenCom 131 infocom systems installing an expansion set is not necessary:

- With the OpenCom 150, all functions reside on only one large module. For this reason, the installation of an expansion set is not necessary for the OpenCom 150.

- The OpenCom 131 is equipped with a single slot for installing an additional interface card. Installing an expansion set is not necessary.

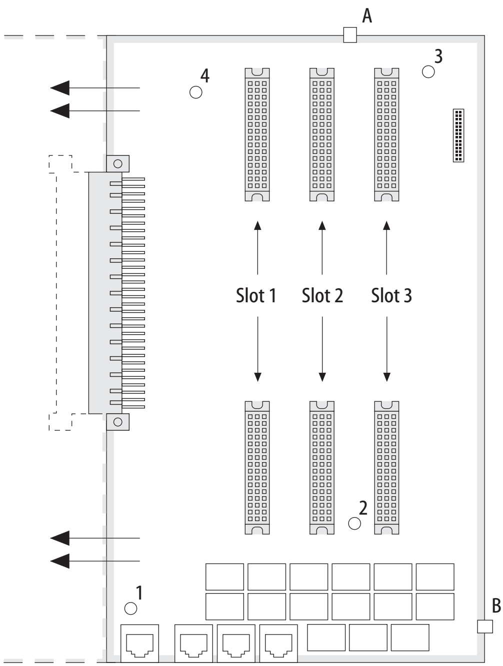
The OpenCom 130 has two separate modules offering more flexibility for extending the system. You can either install the expansion module when you first assemble the system or later as part of a system upgrade. In both cases, follow the mounting sequence as described here:

1. Turn off the OpenCom 130. Unplug the plug-in power supply from the socket. You should not install the expansion module or install or uninstall additional interface cards while the OpenCom 130 is turned on.

2. Open the housing cover of the OpenCom 130. In this case, carefully follow the *Safety Precautions* starting on page 25.

   The existing basic module is mounted in the left half of the housing. No components may be mounted in the right half of the housing.

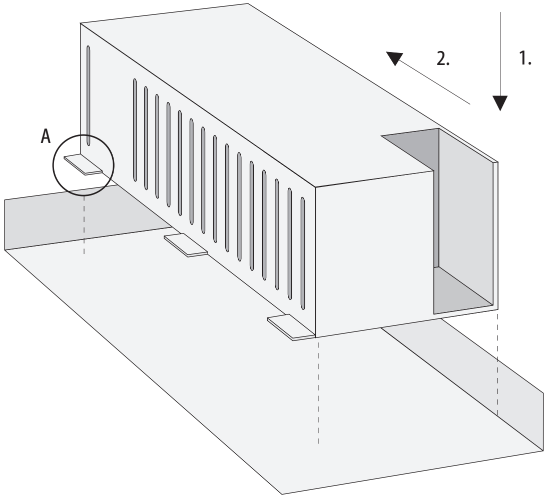
3. Place the expansion module in the intended mounting location in the right half of the housing. Be sure to align the 96-pin connector properly to the jack on the basic module. Push the expansion module towards the basic module so that both modules are securely connected to one another via the 96-pin connector.



Installing the expansion module

4. Carefully press the expansion module at the top and bottom right, pushing it into the locking hooks provided (see “A” and “B” in the diagram). Secure the

expansion module using the Phillips screws provided in the expansion set (see "1" to "4" in the diagram).



Installing the power supply for the expansion module

5. Place the power supply to the right of the expansion module. Carefully press the power supply into the mounting recess provided (shown in the diagram as "1"). Move the power supply forwards until it snaps into place in all six pressure terminals ("A", "2").

6. Connect the power supply output (flat conductor cable) to the appropriate jack of the expansion module. Insert the fully insulated connector of the mains supply in the power supply connection provided.

7. Establish an Ethernet connection between the LAN port of the basic module and the LAN1 port of the expansion module (see also *Positions of the Ports* starting on page 41). To do this, use the short Ethernet connection cable from the expansion set. The LAN0 port of the expansion module is intended for connection to a corporate LAN. You can connect an existing Ethernet connection cable to the LAN0 port of the expansion module.

You usually install at least one interface card on the expansion module. To do this, read the instructions in the following section.

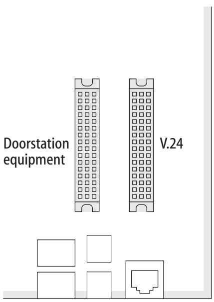
Please note: Two power supplies are provided for the OpenCom 130 with an expansion module. Always turn on the power supply of the expansion module first and then plug in the plug-in power supply.

3.7 Installing Interface Cards

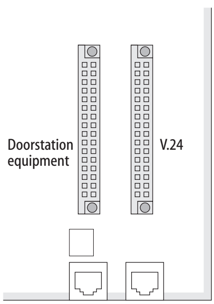
The expansion module and the basic module of the OpenCom 130 resp. the main module of the OpenCom 150 can be expanded using interface cards. The main module of the OpenCom 131 is quipped with one large and two small slots for installing interface cards.

Please note: Turn off the OpenCom 100. Unplug the plug-in power supply and (for an OpenCom 130 / 150) the main supply from the socket. You must not install or uninstall interface cards while the OpenCom 100 is turned on.

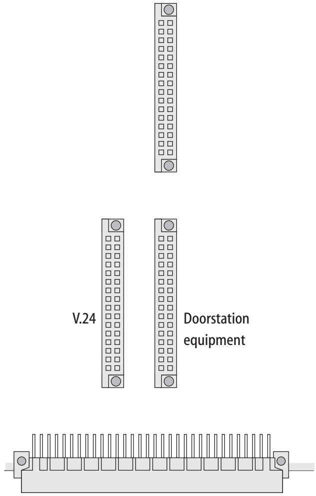
3.7.1 V.24 and Doorstation Equipment Slots



OpenCom 130: V.24 and doorstation equipment slots



OpenCom 131: V.24 and doorstation equipment slots



OpenCom 150: V.24 and doorstation equipment slots

The OpenCom 100 has two slots in which special interface cards can be operated (V.24 and doorstation equipment). You can see the location of these slots in the diagrams.

- The doorstation equipment interface card (also called “Door Phone” or “Door Bell”) provides “actor” ports and “sensor” ports.

  With the OpenCom 130 and the OpenCom 131 systems, an “M100-TFE” interface card can be used. For connecting a door station, this card type offers two actor and three sensor ports.

  With the OpenCom 150 system, an “M100-TFE-2” interface card can be used. This card type offers four actor and four sensor ports. With this card, two door stations can be connected with two call buttons for each door station.

- The V.24 interface card provides a serial port.

Proceed as described below to install one or both of these interface cards:

1. Turn off the OpenCom 100. Open the housing cover.

2. Remove the slot card from the transport packaging. Check that it is the correct type of slot card. (There is a sticker with the type name on the connector.)

CAUTION! Static charges can damage electronic components. Pay attention to the regulations regarding the handling of electrostatically sensitive components.

3. Carefully insert the interface card in the slot provided. The component side must face to the right.

   Ensure the plug-in connection is sitting securely.

4. Connect the required port cable to the corresponding pressure terminals or RJ45 jacks (see also *Positions of the Ports* starting on page 41).

5. Close the housing cover. Turn on the OpenCom 100.

You can query the status of the doorstation equipment interface card and the V.24 interface card in the Web console when the OpenCom 100 is operational again. To do this, call up the **Telephony: Ports: Slots** menu page. The **Status** column in the table displays a green tick beside the name of the interface card.

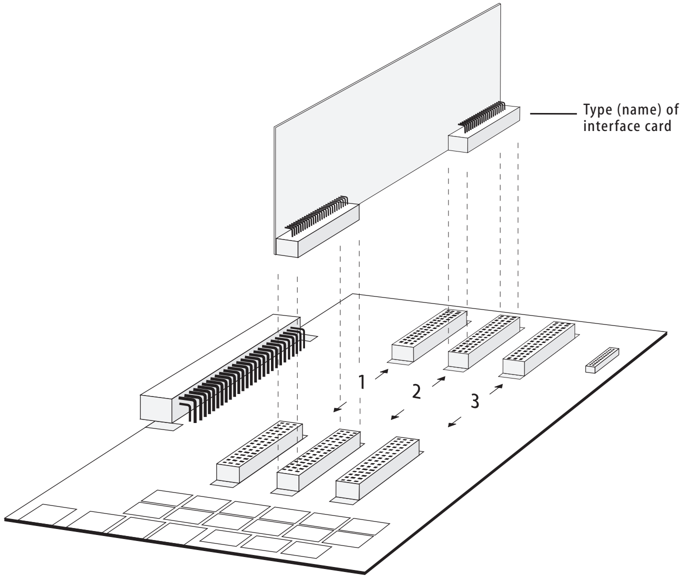
3.7.2 Slots for Additional Interface Cards

The OpenCom 131 offers one large slot to operate an additional interface card. The interface card is connected with two port jacks.

The OpenCom 130 has three (the OpenCom 150 five) large slots in which you can operate interface cards. Each interface card is connected to the expansion module resp. to the main module via two port jacks. The following properties characterise the large slots of the OpenCom 130 / 150:

- There is no prescribed order in which to use the jacks. You can, for example, therefore operate an interface card in slot 3 even though slot 2 is not occupied.
- Each of the slots is connected to a group of pressure terminals. Therefore there are also three pressure terminal groups on the expansion module of the OpenCom 130 (five on the main module of an OpenCom 150). To be able to distinguish these, all the pressure terminals in a group are the same colour.

- The slots are not of the same type. Therefore some of the available interface cards may not be operated in all slots. Note the overview under *Interface Cards* starting on page 43.



OpenCom 130: Installing an interface card in an expansion module slot

Proceed as described below to install an interface card:

1. Turn off the OpenCom 100. Open the housing cover.

2. Remove the slot card from the transport packaging. Check that it is the correct slot card type. There is a sticker with the type name on the connector.

CAUTION! Static charges can damage electronic components. Pay attention to the regulations regarding the handling of electrostatically sensitive components.

3. Carefully insert the interface card in the slot provided. The component side must face to the right.

Ensure the plug-in connection is sitting securely.

4. Connect the required port cable to the corresponding pressure terminals of the relevant pressure terminal group (see also *Positions of the Ports* starting on page 41).

5. Close the housing cover and turn on the OpenCom 100 again.

The software of the OpenCom 100 can detect the type of interface card present. The interface card must still be configured individually for commissioning.

You can query the status of the interface cards in the Web console when the OpenCom 100 is operational again. To do this, call up the **Telephony: Ports: Slots** menu page. The **Status** column of the table displays a green tick beside the slot name (for an OpenCom 130 e.g. **0/1**, **0/2** and **0/3**). The column of the table must list the correct type of interface card.

3.7.3 Installing an M100-AT4 Card

The **M100-AT4** Interface Card allows the OpenCom 100 to be connected to analogue trunk lines. The card has four analogue ports; the supported dialling mode is dual-tone multifrequency (DTMF).

Calling Line Identification Presentation (CLIP) via FSK is supported. Transmission of call charges data (12kHz or 16kHz pulses) is currently not supported.

Installing the Interface Card
Installing the M100-AT4 interface card is possible in the following slots (see also *Installing Interface Cards* starting on page 30).

- OpenCom 130: slot no. 3.
- OpenCom 150: slot no. 3, no. 4 or no. 5
- OpenCom 131: slot no. 1

For the pin out, please refer to *Positions of the Ports* starting on page 41. Use the **Configurator** in the Web console to configure the M100-AT4 Interface Card. Please also refer to the OpenCom 100 online help.

Configuring the slot

1. Log in to the **Configurator** as a user with administrator rights.

   Switch to the **Configurator**'s **Expert** mode to obtain all required dialogues.

2. Open the **Telephony: Ports: Slots** menu.

   Card type **a/b Trunk** is displayed in the **inserted** column for the selected **Slot** if the card is inserted into the slot.

Note: If the card type is not displayed, the interface card has not been detected. Ensure that the interface card has been installed correctly.

3. Click on the desired entry in the **Slot** table.

4. Select card type **a/b Trunk** from the **configured** list.

5. Click on the **Apply** button.

Creating a new bundle

After you have configured the slot, the trunk lines need to be configured. Start by creating a new bundle:

1. Open the **Telephony: Trunks: Bundle** menu.

   On the status page, the list of previously configured bundles is displayed.

2. Click on the **New** button.

3. Select the **Analog trunk** entry from the **Access type** list and enter a bundle name in the **Name** field.

4. Fill in the other fields of the dialogue. The settings in the **Max. time between ringing pulses**, **Time to ready line** and **Time to ready dial out** fields are mandatory.

5. Click on the **Apply** button.

Assigning a route

In order to allow the analogue lines of the M100-AT4 Interface Card to be used for outgoing calls, the bundle needs to be assigned to a route:

1. Open the **Telephony: Trunks: Route** menu.

   On the status page, the list of previously configured routes is displayed.

2. Click on the **New** button. Alternatively, click one of the route entries to assign the bundle to an existing route.

3. Select the bundle you have configured from the **Choice** list and enter a route name in the **Name** field (only if you are configuring a new route).

4. Click on the **Apply** button.

   The newly configured or changed route is displayed in the list on the status page.

Configuring the a/b ports

In the following step, you have to configure each of the M100-AT4 Interface Card ports:

1. Open the **Telephony: Ports: a/b** menu.

   The four M100-AT4 Interface Card ports are displayed for the selected slot.

2. Click on the desired a/b port.

3. Select the configured bundle from the **Bundle** list. Enter the desired access number in the **Port Ph. No.** field.

4. Click on the **Apply** button.

Configuring the call distribution

In order to allow incoming calls to be signalled to the designated terminals, the access numbers configured for the M100-AT4 Interface Card a/b ports must be assigned to internal call numbers:

1. Open the **Telephony: Call Distribution: Incoming** menu.

Previously assigned access numbers are displayed in the list. The list also displays the access numbers of the a/b ports you have configured.

2. In the **Phone no.** column, click the entry you wish to assign internal call numbers to.

3. Enter up to ten internal call numbers to which incoming external calls are to be signalled.

4. Click on the **Apply** button.

5. Repeat steps 2 to 4 until all allocations have been set in the call distribution.

3.8 Available Ports

The OpenCom 100 has the following ports (see also *Positions of the Ports* starting on page 41):

3.8.1 OpenCom 130

The listed interfaces and ports are located on the basic module of the OpenCom 130. Further interfaces and ports can be added by installing the expansion set and additional interface cards (see *Installing an Expansion Set* starting on page 27 and *Installing Interface Cards* starting on page 30).

- One S_0 port to connect to an external S_0 bus (usually the NTBA), designed as an RJ45 jack ($S_0$1)

- One switchable S_0 port ($S_0$2), which can be connected as either an internal or external S_0 bus. The internal connection is via a pressure terminal and the external connection via an RJ45 jack.

- Three U_{pn} ports, designed as pressure terminals (U_{pn}1 to U_{pn}3)

- Four analogue a/b ports, designed as pressure terminals (a/b1 to a/b4)

- One slot to incorporate a doorstation equipment interface card. The following ports can be used with such a card:

two actor ports for connection to a door opener and the intercom of doorstation equipment. These are designed as pressure terminals (actor1 to activate a door opener and actor2 to activate doorstation equipment);

three sensor ports for connection to the bell keys of doorstation equipment, designed as pressure terminals (sensor 1 to sensor 3)

- One slot to incorporate a V.24 interface card. The following port can be used with the V.24 add-on card:

  one COM port to connect to a PC to configure and transmit connection data, designed as an RJ45 jack

- One port to the LAN (10BaseT), designed as an RJ45 jack
- One CompactFlash socket to incorporate a type I or type II CompactFlash memory card. These cards are necessary to operate the OpenVoice and Open-Attendant programme packages. Only use high speed memory cards which are recommended by Aastra. Other memory cards or “Microdrive” type memory cards may not be able to maintain the required access speed.

Tip: For the stored voice announcements and a recording time of one hour a memory card with 32 Mbyte is required.

- One port jack to connect the plug-in power supply to power the basic module. Only use the original OpenCom 130 plug-in power supply provided in the supply scope to power the basic module.

3.8.2 OpenCom 131

- One S_0 port to connect to an external S_0 bus (usually the NTBA), designed as an RJ45 jack ($S_0$1)
- One switchable S_0 port ($S_0$2), which can be connected as either an internal or external S_0 bus. The internal connection is via a pressure terminal and the external connection via an RJ45 jack.
- Three U_{pn} ports, designed as pressure terminals (U_{pn}1 to U_{pn}3)
- Eight analogue a/b ports, designed as pressure terminals (a/b1 to a/b8)

- One slot to incorporate a doorstation equipment interface card. The following ports can be used with such a card:

  two actor ports for connection to a door opener and the intercom of doorstation equipment. These are designed as pressure terminals (actor1 to activate a door opener and actor2 to activate doorstation equipment);

  three sensor ports for connection to the bell keys of doorstation equipment, designed as pressure terminals (sensor 1 to sensor 3)

- One slot to incorporate a V.24 interface card. The following port can be used with the V.24 add-on card:

  one COM port to connect to a PC to configure and transmit connection data, designed as an RJ45 jack

- One port to the LAN (10BaseT), designed as an RJ45 jack

- One CompactFlash socket to incorporate a type I or type II CompactFlash memory card. These cards are necessary to operate the OpenVoice and OpenAttendant programme packages. Only use high speed memory cards which are recommended by Aastra. Other memory cards or “Microdrive” type memory cards may not be able to maintain the required access speed.

Tip: For the stored voice announcements and a recording time of one hour a memory card with 32 Mbyte is required.

- One port jack to connect the plug-in power supply. Only use the original OpenCom 131 plug-in power supply provided in the supply scope.

3.8.3 OpenCom 150

The listed interfaces and ports are located on the main module or on the connection module of the OpenCom 150. Further interfaces and ports can be added by installing additional interface cards (see *Installing Interface Cards* starting on page 30).

- One slot to incorporate a double doorstation equipment interface card. The following ports can be used with such a card:

Four actor ports for connection to a door opener and the intercom of door-station equipment. These are designed as pressure terminals (actor1 and actor3 to activate door openers, actor2 and actor4 to activate doorstation equipment)

Four sensor ports for connection to the bell keys of doorstation equipment, designed as pressure terminals (sensor 1 to sensor 4)

- One slot to incorporate a V.24 add-on card. The following port can be used with the V.24 add-on card:

  one COM port to connect to a PC to configure and transmit connection data, designed as an RJ45 jack

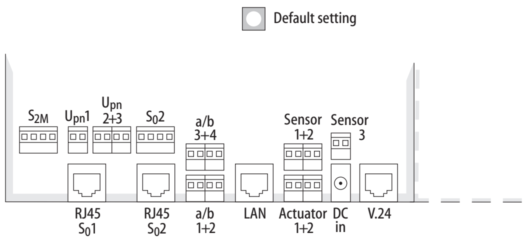
- Two PCM ports for the connection of up to three systems, designed as an RJ45 socket (PCM)
- Two ports to the LAN (100BaseT), designed as RJ45 jacks
- One CompactFlash socket to incorporate a type I or type II CompactFlash memory card. These cards are necessary to operate the OpenVoice and Open-Attendant programme packages. Only use high speed memory cards which are recommended by Aastra. Other memory cards or “Microdrive” type memory cards may not be able to maintain the required access speed.

Tip: For the stored voice announcements and a recording time of one hour a memory card with 32 Mbyte is required.

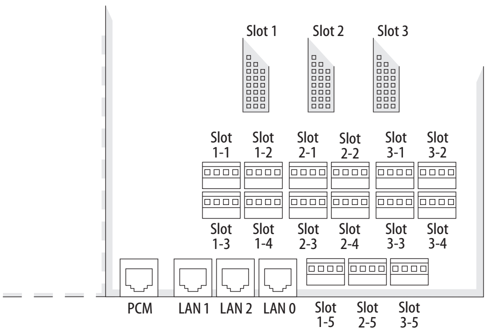
- One port jack to connect the plug-in power supply. Only use the original OpenCom 150 plug-in power supply provided in the supply scope.

3.8.4 Positions of the Ports

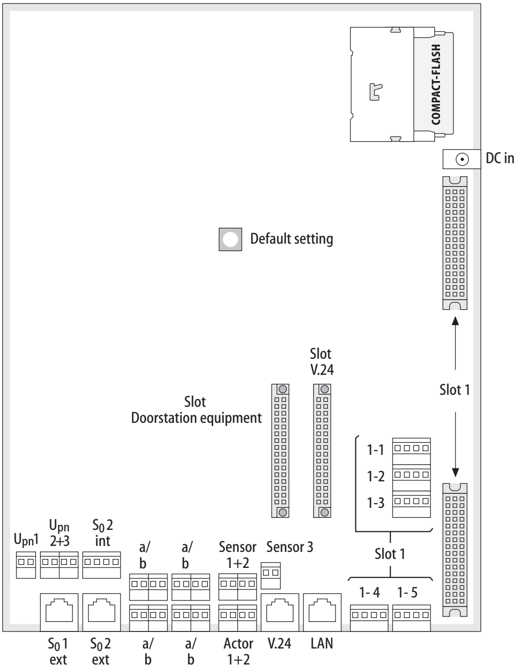
The following diagrams show the positions of the ports:



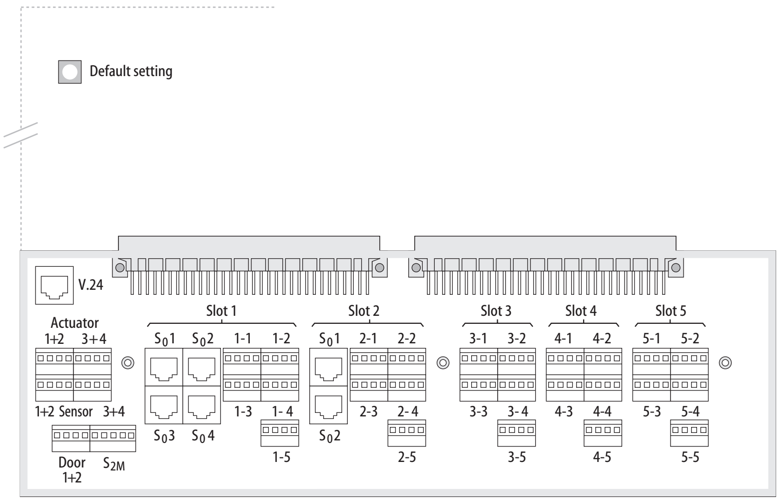
Position of the ports on the basic module (OpenCom 130)



Position of the ports on the expansion module (OpenCom 130)



Position of the ports on the OpenCom 131



Position of the ports on the OpenCom 150

3.9 Interface Cards

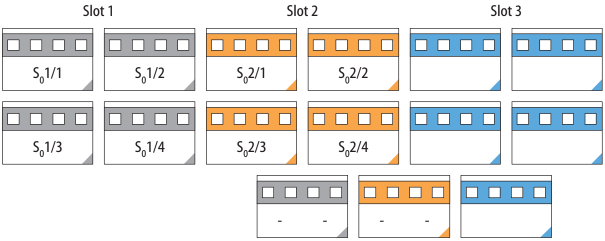
3.9.1 OpenCom 131 (1 Slot)

An interface card of type “M100-AT4” can be installed in the free large slot. With this interface card it is possible to connect the OpenCom 131 to analogue trunk lines.

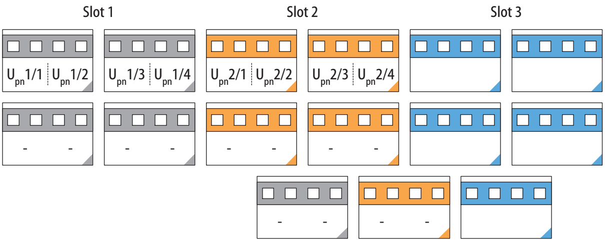
3.9.2 OpenCom 130 (3 Slots)

The following overview shows the available interface cards.

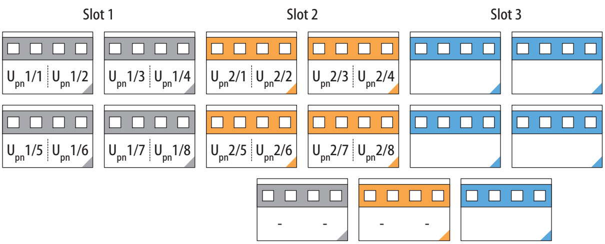
Interface card	**Slots**			**Special features**
	1	**2**	**3**	
M100-S4: 4 x S_0	●	●		S_0 are switchable internally/ externally
M100-U4d: 4 x U_{pn}	●	●		U_{pn} are DECT-enabled
M100-U8d: 8 x U_{pn}	●	●		U_{pn} are DECT-enabled
M100-S2U6d: 2 x S_0 and 6 x U_{pn}	●	●		U_{pn} are DECT-enabled S_0 are switchable internally/ externally
M100-S2A6: 2 x S_0 and 6 x a/b	●	●		S_0 are switchable internally/ externally
M100-A4: 4 x a/b	●	●	●	
M100- A8: 8 x a/b	●	●	●	
M100-AT4			●	4 analogue trunk lines
M100-IP		●	●	Voice over IP



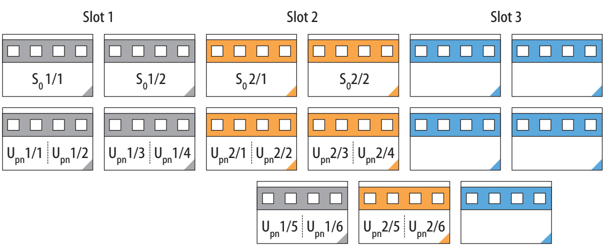
Ports: 4 x S_0



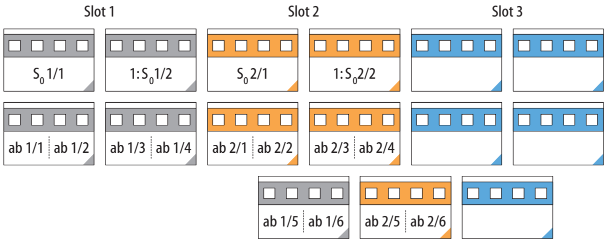
Ports: 4 x U_{pn}



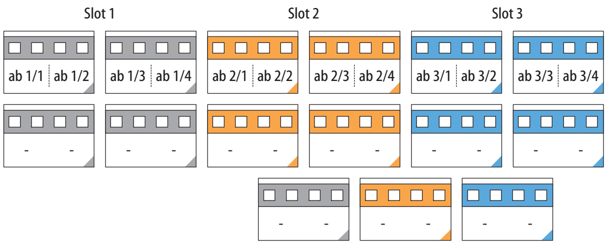
Ports: 8 x U_{pn}



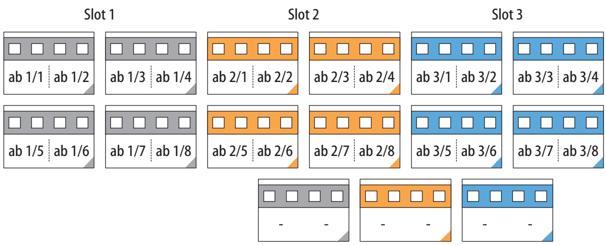
Ports: 2 x S_0 and 6 x U_{pn}



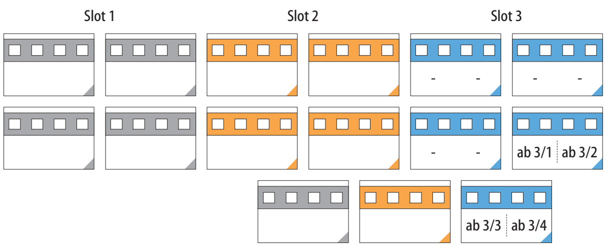
Ports: 2 x S_0 and 6 x a/b



Ports: 4 x a/b



Ports: 8 x a/b

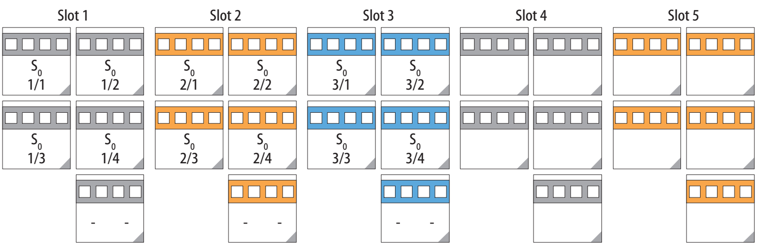


M100-AT4 Interface Card (4 analogue trunk lines)

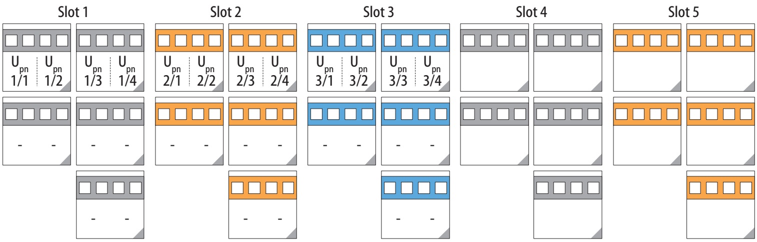
3.9.3 OpenCom 150 (5 Slots)

The following overview shows the available interface cards.

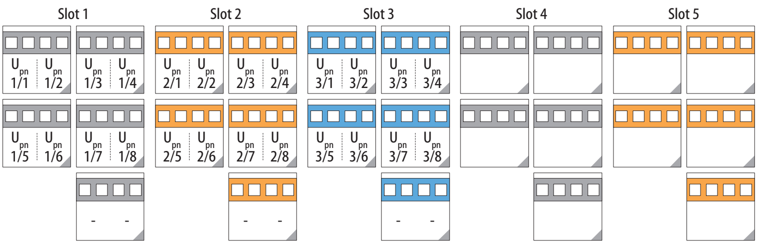
Interface card	**Slots**					**Special features**
	1	**2**	**3**			
M100-S4: 4 x S_0	●	●	●			S_0 are switchable internally/ externally
M100-U4d: 4 x U_{pn}	●	●	●			U_{pn} are DECT-enabled
M100-U8d: 8 x U_{pn}	●	●	●			U_{pn} are DECT-enabled
M100-S2U6d: 2 x S_0 and 6 x U_{pn}	●	●	●			U_{pn} are DECT-enabled S_0 are switchable internally/ externally
M100-S2A6: 2 x S_0 and 6 x a/b		●	●			S_0 are switchable internally/ externally
M100-A4: 4 x a/b		●	●	●	●	
M100-A8: 8 x a/b		● *)	●	●	●	*) In case no doorstation equipment interface card is used
M100-AT4			●	●	●	4 analogue trunk lines
M100-IP			●	●		Voice over IP



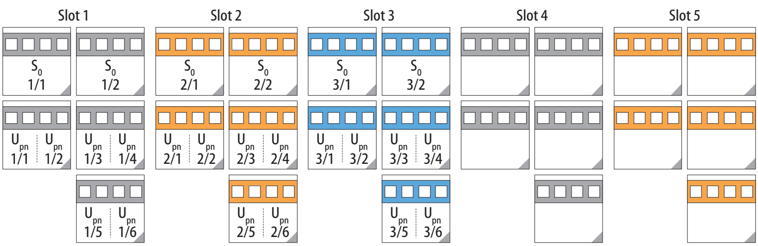
Ports: 4 x S_0



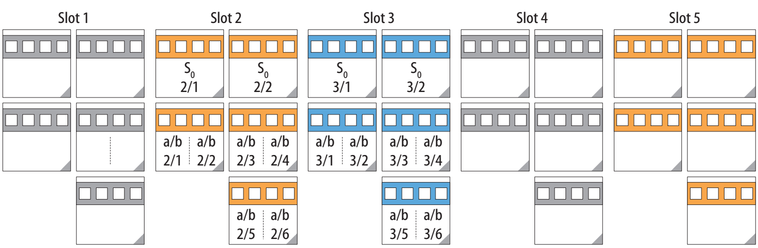
Ports: 4 x U_{pn}



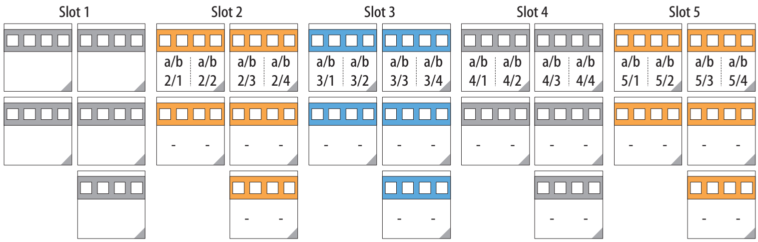
Ports: 8 x U_{pn}



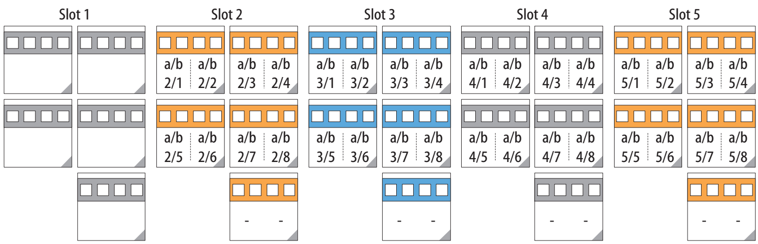
Ports: 2 x S_0 and 6 x U_{pn}



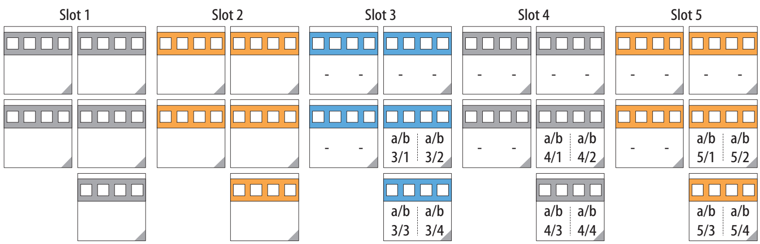
Ports: 2 x S_0 and 6 x a/b



Ports: 4 x a/b



Ports: 8 x a/b



M100-AT4 Interface Card (4 analogue trunk lines)

3.10 Port Assignment, Termination, Cable Lengths

3.10.1 S_0 Ports

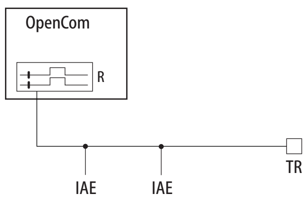
Whether you use the switchable S_0 ports for internal or external communication depends on your communications requirements and the existing basic accesses.

Note that the S_0 bus requires a terminating resistor of 100 ohms at each end.

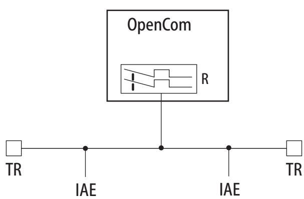
In the case of the OpenCom 100, the S_0 buses are terminated by software. You make this setting in the S_0 port configuration in the **Configurator** on the Web console.

You can connect up to eight terminals on every internal S_0 bus; up to three of the terminals can operate without an external power supply. The length of the four-wire cable of an internal S_0 bus must not exceed 150 m. The power consumption of each internal S_0 bus is approx. 2 W.

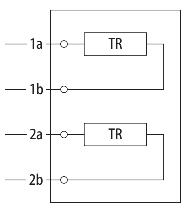
S_0 Ports on Interface Cards



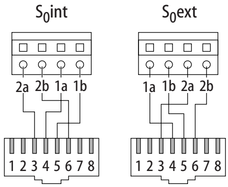
The S_0 bus is terminated at one end by the OpenCom 100.
IAE = ISDN socket (German: "ISDN Anschluss Einheit") or an ISDN terminal.
TR = terminating resistor, the S_0 termination. The TR must be at the termination of the line. This can also be done by an appropriately wired IAE.



The S_0 bus is terminated by the TR at the ends.



Termination on an ISDN socket



Switchable S_0 port on a pressure terminal

You can add further S_0 ports to the OpenCom 130 / 150 by installing suitable interface cards in a slot. These S_0 ports can be switched between internal and external operation.

Annotation concerning an OpenCom 130:
In contrast to the basic module, the expansion module does not provide any additional RJ45 jacks for external S_0 ports. Therefore you can also use the pressure terminals of the expansion module for an external S_0 port. The port assignment of the pressure terminals is changed when the switch is made from internal to external operation. This can be seen in the diagram.

Tip: Let us say you have activated an IAE on an internal S_0 port, for example. If you switch this S_0 port to external operation, you require a crossed ISDN line to connect the IAE to an NTBA. The assignment of a crossed line is described in the chapter *PBX Networking* under *Direct Connection* starting on page 156.

3.10.2 U_{pn} Ports

Each of the U_{pn} ports enable the connection of a RFP 22 / 24 DECT base station, an Aastra 6771 / 6773 / 6775 (OpenPhone 71 / 73 / 75) or an OpenPhone 61 / 63 / 65 telephone using a twin-wire cable.

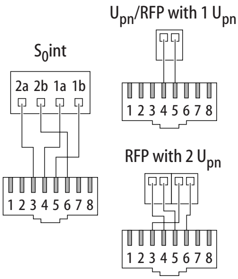
Note: DECT base stations cannot be operated on the U_{pn} ports of the basic module of an **OpenCom 130**. This is only possible for U_{pn} ports on the interface cards of the expansion module.

Note: Only the system telephones Aastra 6771 / 6773 / 6775 (OpenPhone 71 / 73 / 75) and OpenPhone 61 / 63 / 65 can be operated on the U_{pn} ports of an **OpenCom 131**. Operating a DECT base station is not possible on these U_{pn} ports.

The maximum permissible length of the twin-wire cable on a U_{pn} port is 500 m. This line may only be laid inside buildings.

The maximum permissible length of the twin-wire cable on a U_{pn} port of an interface card is 1,000 m when 0.6 mm cable (with twisted pairs) is used (these interface cards are available for an OpenCom 130 / 150).

The power consumption of each U_{pn} port is approx. 3 W.



Pin assignment of the S_0 and U_{pn} ports

3.10.3 a/b Ports

The a/b ports are for operating analogue devices (e.g. a fax machine, modem or telephone). The maximum permissible length of the cable is 1,000 m when twin-wire 0.6 mm cable (with twisted pairs) is used.

Doorstation equipment can be activated on an a/b1 port (OpenCom 130, OpenCom 131). In this case, an electronic switch enables the low-frequency voltage to be separated from the feed.

Tip: It is possible to operate analogue trunk lines with an additional interface card. Further explanations regarding this interface card can be found in the “M100-AT4 Interface Card” manual.

3.10.4 Actor/Sensor

In order to operate an entrance intercom and door opener, you need four twin-wire cables:

- one cable between the entrance intercom and the a/b1 port,
- one cable between the door opener and the Actor1 port,
- one cable between the intercom input and the Actor2 port to activate the amplifier as well as
- one cable between the doorbell and the sensor port.

Only use entrance intercoms and door openers complying with the German FTZ Guideline no. 123D12.

A door station of the type “DoorLine” can also be operated on the OpenCom 100. This intercom is connected via a “DoorLine” module to one of a/b ports of the OpenCom 100 (see *Intercom System (for a/b)* starting on page 57).

3.10.5 LAN Port

The LAN port of the OpenCom 131 and the LAN port on the basic module of the OpenCom 130 enables integration of the OpenCom 100 into an existing in-house LAN by means of a 10 Mbit hub.

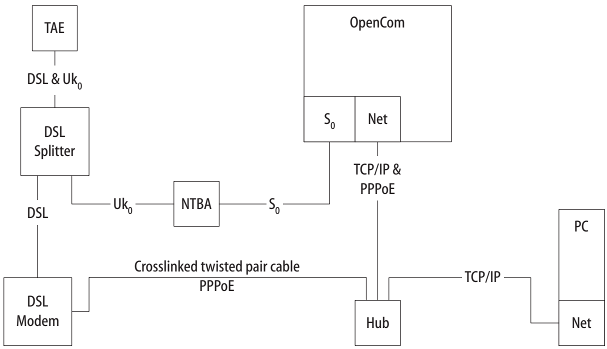
LAN Ports on the Expansion Module of the OpenCom 130
The LAN ports on the expansion module (LAN0, LAN1 and LAN2) lead to the Ethernet switch of the expansion module. These LAN ports support 10 Mbit/s and 100 Mbit/s transmission speeds in half- or full-duplex operation. The change in transmission rate and mode of operation is automatic (“auto-sensing function”). The switch is also automatic for connections which require a crossed LAN line. For this reason, you can also use an uncrossed LAN line for a connection to another hub or switch.

LAN Ports of the OpenCom 150
The technical characteristics of the OpenCom 150 LAN ports (LAN0 and LAN1) are similar to the LAN ports of the OpenCom 130 described in the previous section.

A LAN line (twisted-pair line in accordance with 10BaseT or 100BaseTX) can be up to 100 m long. Secure operation with 100 Mbit/s requires the use of category 5 lines and line sockets. Use a shielded Ethernet cable (STP cable, Shielded Twisted Pair cable).

3.10.6 DSL Port

External DSL modems can be connected via the LAN port. In the case of the OpenCom 100, the output of the DSL modem (NTBBA) is led to the LAN port of the OpenCom 100 via an external switch or hub. The router subsequently converts the PPPoE protocol to the TCP/IP protocol of the LAN.



Connecting the OpenCom 100 to the network via ISDN and DSL

Connection of the DSL modem is via a crossover twisted-pair line. You can also use a switchable port on the hub, which is usually indicated by an “X”.

Note: If an expansion module is installed in an **OpenCom 130**, you can also use the unused LAN2 port to activate the DSL modem. Due to the “auto-crossover” function, you do not require a cross-wired line with a LAN port of the expansion module. If you are operating this OpenCom 130 as a slave system in a cascaded PBX, the LAN0 port on the expansion module of the slave system can be used instead (see *PBX Cascading* starting on page 148).

3.10.7 PCM Port

The PCM port is used cascade two communications systems together (see *PBX Cascading* starting on page 148).

3.11 Power Failure

In the event of a power failure, all the contents in the memory (programme and user data) are saved without change. The internal clock continues to run for 24 hours. If the power failure lasts longer than 24 hours, the time and date are reset to the factory setting when power is switched on again. When the first external outgoing call is made, the time and date are set to the current value as given by the exchange.

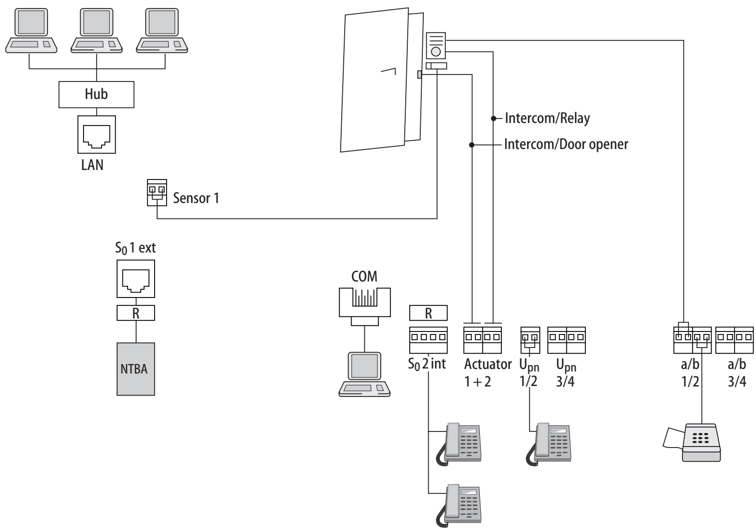
On the multi-terminal access, the OpenCom 130 include an emergency service. In the event of a power failure, the external $S_0$1 port is switched over to the $S_0$2 port so that you can still use a connected telephone to make a call.

Emergency operation on a system access is not possible.

3.12 Connectible Terminals

The OpenCom 100 ports already offer a large number of possibilities for connecting terminals. By installing additional interface cards (in an OpenCom 130 / 150) the number of ports can be increased as required.

One of the many possible configurations is shown in the following diagram.



Example of port assignment of the OpenCom 130 with terminals

3.12.1 Internal/External S_0 Ports

All S_0 ports can be operated externally, i.e. on an ISDN network termination unit. The $S_0$2 port of an OpenCom 130 and an OpenCom 131 can also be connected internally. S_0 ports on interface cards (available for an OpenCom 130 and OpenCom 150) can also be switched external/internal. It is not possible to use both assignments simultaneously.

Up to eight devices per bus (ISDN telephones, ISDN fax machines, ISDN base stations, ISDN cordless telephones, ISDN adapters for the PC among others) can be connected to the internal S_0 ports by twin-pair cables. The power for three of these devices can be supplied by the bus; if more devices are used, they then require their own power supplies. The internal S_0 buses enable point-to-multipoint calls as per the DSS1 protocol (Euro-ISDN).

3.12.2 U_{pn} Ports

A system terminal can be connected to every U_{pn} port by a twin-wire cable.

The Aastra 6771 / 6773 / 6775 (OpenPhone 71 / 73 / 75) and the OpenPhone 61 / 63 / 65 are cord-bound system terminals. The OpenPhones 63 and 65 can be cascaded on a U_{pn} adapter so that you can operate two OpenPhones on one U_{pn} port (for further details, refer to the “OpenPhone 61, 63, 65” user guide).

The RFP 22 / 24 base station is required for the use of cordless system telephones (e.g. OpenPhone 2x, Aastra 142d or Aastra 610d / 620d /630d).

Note: You can only operate DECT base stations on the ports of U_{pn} interface cards. These cards are available for an OpenCom 130 and an OpenCom 150.

If this base station is connected to a U_{pn} port of an interface card, four simultaneous calls are possible with the handsets. If the base station is connected to two U_{pn} ports, eight simultaneous calls are possible. However, note that only as many external connections are possible as there are externally connected B-channels available.

If you use a cordless handset such as the Aastra 610d / 620d /630d, this can also be contacted by other base stations using the GAP/CAP standard.

3.12.3 a/b Ports

The a/b ports a/b1 to a/b4 can be used for connecting analogue terminals. These can be for voice or data communication, and use DTMF or pulse dialling, e.g.

- analogue telephones
- class 3 fax machines
- analogue modems (external or internal)
- external devices for music on hold
- external voice mail systems.

Additional a/b ports can be provided by installing interface cards (on an OpenCom 130 / 150).

Please note: Adhere to the following notes and recommendations regarding the connection of analogue devices. Devices not meeting the technical requirements of the OpenCom 100 can cause damage to it.

Analogue Telephones
If analogue telephones are to be used, we recommend the use of devices featuring voice-frequency (VF) signalling, as the additional features of the OpenCom 100 cannot be used with pulse dialling.

Modems
The maximum transmission rate for analogue modems is 33.6 kbit/s (V.34+).

Music on Hold
A suitable external device for music on hold is the Genius 2000, manufactured by Speech Design. If you do not operate an external MoH device, the OpenCom 100 offers an internal MoH, which you can load in the Web console **Configurator** in the **System: Components** menu. For details, refer to the online Help.

Please note: Use only devices with an input impedance of 600 ohms, floating connection, for external music on hold. Incorrect input impedance can cause irreparable damage to the OpenCom 100.

Voice Mail
If you are using an external voice mail system, it must be capable of handling the number of digits used for internal telephone numbers, e.g. five digits if you have configured five-digit internal numbers. We recommend the following Speech Design products: Memo 200 / 300 / 400 or Memo 200-A / 300-A / 400-A.

The external voice mail system can be connected to internal a/b ports as well as to internal S_0 ports. For both port types the voice mail system can activate the notification for system terminals with the code procedures *68 resp. #68.

Intercom System (for a/b)
The intercom systems “DoorLine T01 / 02” and “DoorLine T03 / 04” of the Deutsche Telekom can be connected via the “DoorLine M06” to any a/b port. The “DoorLine” module provides the actor for the door opener contact.

Observe the following when connecting:

- The intercom system and the “DoorLine” module should be set to their factory settings.

- In the **Telephony**: **Ports**: **a/b**: **Change** menu in the Configurator, select **Door-station 2-wire** under **Type**. Deactivate the **Actuator** option, if you want to use the actor port of the OpenCom 100 instead of the “DoorLine” relay. The “DoorLine” actor can be operated only when the speech channel is open at the same time. The internal actor can be operated at any time.

- The “DoorLine” intercom system has a number of bell keys to which you can assign different call numbers in the **Telephony**: **Ports**: **Doorbell** menu in the Configurator.

- You can call the “DoorLine” intercom system by entering the code procedure *102.

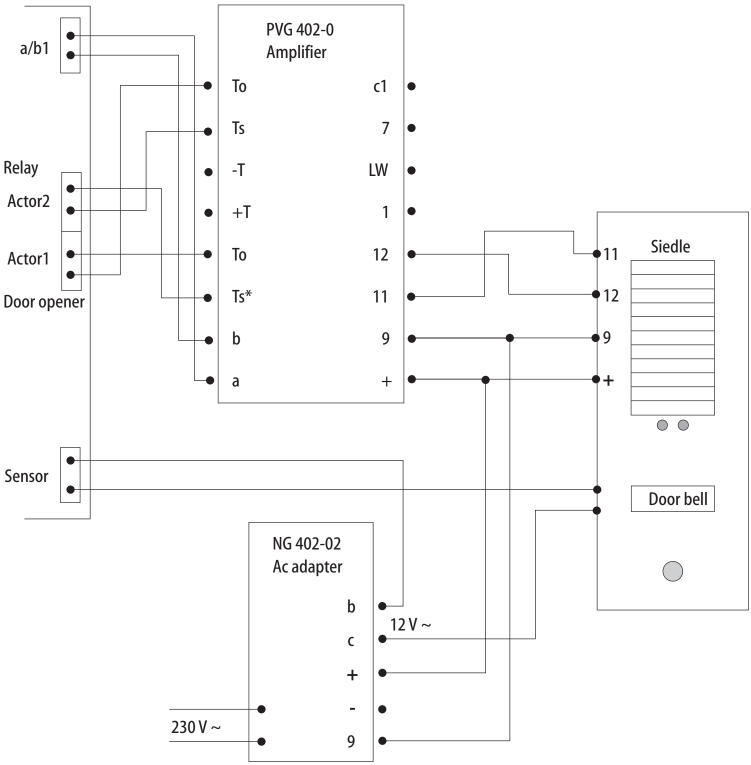
- The “DoorLine” intercom system can be connected to any a/b port. However, you can use only one “DoorLine” with the OpenCom 100.

For details on installing and configuring the “DoorLine” intercom system, refer to the product user guide.

The intercom system should be installed by qualified personnel only as sensor/ actor contacts will need to be connected to the “DoorLine” module.

3.12.4 Actor/Sensor Ports

For the assignment of the ports, refer to the section *Available Ports* starting on page 37. The OpenCom 100 also functions together with a Freehand Entry-Phone manufactured by Siedle or Behnke.



Connection of doorway equipment produced by Siedle

Note: The above diagram shows the usage of the "PVG 402-0" module (which merely serves as an example). Other modules can also be operated, such as its successor, "PVG 602-01".

3.12.5 COM Port

By installing the V.24 interface card a serial port on the COM interface is provided.

Please note: The connection line for the COM port can be up to three metres long.

A PC for configuring the OpenCom 100 or transmitting call data can be connected to the COM port. This call data can be evaluated in detail with the call charge registration programme OpenCount.

3.12.6 LAN Port

Using the LAN port (Ethernet) you can integrate an OpenCom 100 into your corporate network (local area network), and thus use it, among other things, as an IP router for accessing the Internet.

LAN Ports on the Expansion Module of the OpenCom 130
The LAN ports of the Ethernet switches on the expansion module process Ethernet data traffic with different degrees of priority. You should therefore assign the three LAN ports on the expansion module as follows:

- LAN2: Use this port if you cascade a second communications system as a slave system (see *PBX Cascading* starting on page 148).
- LAN1: Here connect the short Ethernet connection line to the basic module.
- LAN0: You should use the port with the lowest priority to connect to your corporate network.

Both internal LAN ports of the Ethernet switch are reserved for usage on interface cards.

LAN Ports of the OpenCom 150

- LAN1: Use this port if you cascade a second communications system as a slave system (see *PBX Cascading* starting on page 148).
- LAN0: You should use the port with the lowest priority to connect to your corporate network.

3.13 Installing the Memory Card

To operate the OpenAttendant and OpenVoice programme packages you require a memory card (Compact Flash card Type I or Type II). The memory card contains the following files:

- the voice files and welcome/closing texts for the voicemail system **OpenVoice** as well as messages which callers have left,
- the audio files for the programme package **OpenAttendant**, with which you can create individual information systems such as voice portals and voicemail systems.

Only use high speed memory cards which are recommended by Aastra. Other memory cards or "Microdrive" type memory cards may not be able to maintain the required access speed.

Note: If you use a memory card other than one recommended by Aastra, Aastra will not accept any responsibility for the correct functioning of the **OpenVoice** and **OpenAttendant** programme packages.

For further information, please contact your local dealer or the Aastra sales department.

The following instructions are intended for persons authorised to install such memory cards, in the event that there is still no memory card available in the communications system.

3.13.1 Safety Precautions

Please note: The memory card may only be installed by authorised personnel because the slot for the card is located on the module board of the communications system.

DANGER! **High voltage inside the device.**

3.13.2 To Install a Memory Card …

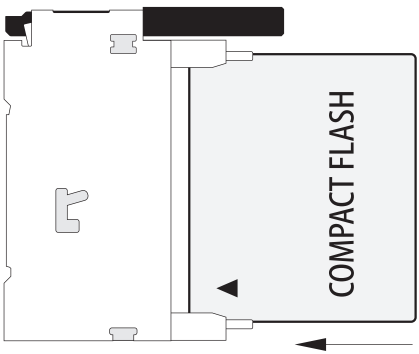
1. Unplug the OpenCom 100.

2. Open the OpenCom 100 housing cover.

CAUTION!	Make sure you protect yourself against electrostatic discharge.

Static electricity can damage the memory card. To avoid such damage, the static electricity must be earthed from your body. Work only in an anti-static environment. If possible, use earthing underlays or anti-static mats.

3. Insert the memory card in the base on the upper edge of the module board.



Inserting a Compact Flash card

4. Replace the housing cover.

5. Insert the mains plug.

The OpenCom 100 is restarted. The configuration last loaded is reactivated.

4. Aastra 677x: Extensions and Accessories

4.1 Power Supply Unit

The power supply unit 4516000 (in Britain operate only the AC adapter with the part no. 4516001) is required in the following cases:

- when connecting a key extension to an Aastra 6771 / 6773 / 6775 (OpenPhone 7x) system telephone (see also the chapter *Key Extensions* starting on page 63)
- when using the Aastra 6773ip / 6775ip (OpenPhone 7x IP) IP system telephones (with or without key extension) where no Power over LAN is available in the network

Connecting the Power Supply Unit to an IP Telephone
The connector for the power supply unit is in the bottom of the telephone's casing and is indicated by the symbol .

1. Plug the power supply unit's RJ45 jack into the socket provided.
2. Pass the power supply unit's cable through the recesses on the underside of the IP system telephone.
3. Connect the power supply unit to the mains power supply (see *Connecting the Key Extension* starting on page 65).

4.2 Key Extensions

Up to three key extensions can be connected to system telephones: either three key extensions of the model Aastra M671 or three key extensions of the model Aastra M676. A combination of these key extensions is, however, not possible.

The following equipment combinations are possible:

Key extension …	**… with the features**	**connectable to a system telephone**
Aastra M671	– 36 keys with LED indicator – Labelling on label strips	– Aastra 6773 (OpenPhone 73) – Aastra 6773ip (OpenPhone 73 IP) – Aastra 6775 (OpenPhone 75) – Aastra 6775ip (OpenPhone 75 IP)
Aastra M676	– 20 keys with LED indicator – 3 keys with LED indicator to shift levels; enables programming of 60 storage locations on each key extension – Labelling of the keys over the display; each key is assigned to a display line	– Aastra 6775 (OpenPhone 75) – Aastra 6775ip (OpenPhone 75 IP)

The number of key extensions connected to a system telephone (up to three) can be set in the **Configurator** of the OpenCom 100’s Web Console (in the menu **Telephony**: **Ports**: **Upn** or **Telephony**: **Devices**: **VoIP Phones**). Here the keys can also be programmed as call keys or assigned functions or destination call numbers. Users can change this programming as required. The maximum distance between the connecting socket that the telephone/key extension device combination is operating through and the OpenCom 100 must be less than 1000 metres. You will need a plug-in power supply unit no. 4516000 (in Britain operate only the AC adapter with the part no. 4516001) to provide power. The power supply unit is plugged into the last in the series of key extensions.

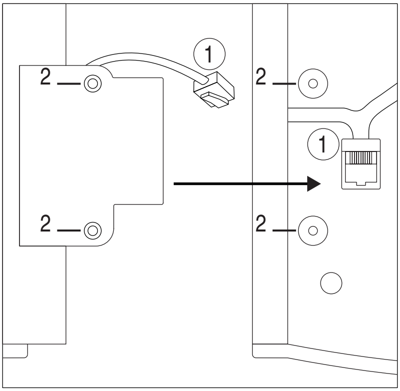
Configuration	Needs Power Supply Unit
U_{pn} system telephone	No
U_{pn} system telephone with 1-3 key extensions	Yes
IP system telephone	Yes
IP system telephone with 1-3 key extensions	Yes
IP system telephone with PoE (Power over Ethernet)	No
IP system telephone with 1-3 key extensions and PoE	No

A system telephone requires a power supply unit if a key extension is installed. When using PoE, an IP system telephone requires no power supply unit.

Connecting the Key Extension

CAUTION! Guard against static charges!
Static charges can damage the OpenCom 100's electronic components. Make sure you discharge yourself and your tools before and during any installation work on the OpenCom 100 and any connected terminals. Use discharging underlays or antistatic mats where possible.

Please note: **Never** attach a key extension to a system telephone that is already connected to OpenCom 100. Pull the telephone plug out of the socket before screwing the key extension onto it.



Underside of the device: key extension (left) and system telephone (right)

This symbol on the system telephone indicates the connector for the key extension. It is on the underside of the telephone. This symbol on the key extension indicates the connector for a further key extension.

This symbol on the key extension indicates the connector for the power supply unit and is on the underside of the device. This is the same connector which can be used instead of connecting an additional key extension.

1. Plug the key extension's RJ45 jack into the system telephone's RJ45 socket (1).
2. Screw the key extension onto the system telephone (2).
3. Plug the power supply unit's RJ45 jack into the socket provided on the right-hand side of the key extension.
4. Pass the power supply unit's cable through the recesses provided on the underside of the key extension and the system telephone.
5. Connect the power supply unit to the mains power supply.
6. Connect the system telephone with the U_{pn} or ethernet port.

4.3 Headset

A headset can be connected to the Aastra 6771 / 6773 / 6775 (OpenPhone 7x) system telephones and to the Aastra 6773ip / 6775ip (OpenPhone 7x IP) IP system telephones.

The headset must comply with the DHSG standard (connection via RJ45 jack). The manufacturers Plantronics and GN Netcom make devices suitable for this purpose. Alternatively, you can connect a “normal” headset (RJ11 jack) using an adapter. The headset must comply with DIN Norm EN 60950-1 Point 6.2 (“Safety of information technology equipment including electrical business equipment”).

Connecting a Headset to a System Telephone
The connector for the headset is in the bottom of the system telephone’s casing and is indicated by this symbol .

1. Plug the RJ45 jack on the headset cable into the socket provided.

2. Pass the cable through the recesses provided on the underside of the system telephone.

3. Activate the headset on the system telephone in the menu **Phone settings: Headset** (see also the system telephone’s user guide).

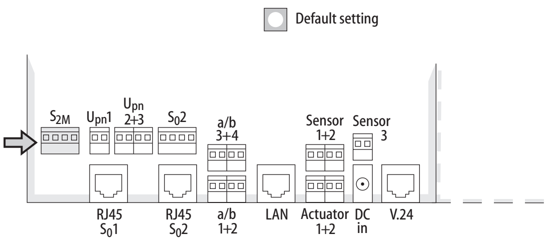
5. S_{2M} Connector Module

The S_{2M} connector module allows you to operate an OpenCom 130 / 150 on a primary rate access. This access provides up to 30 voice channels. You can also use the S_{2M} port to network two systems, e.g. in order to use the OpenCom 130 / 150 as a subsidiary system or DECT server.

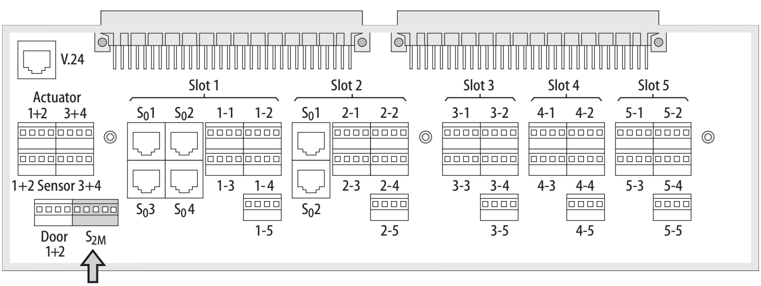
Note on DECT Applications in Conjunction with S_{2M}

If an S_{2M} connector module is installed in the OpenCom 130 / 150, please consider the following restriction for DECT applications: It is not sufficient, to install a U_{pn} interface card (e.g. “4 x U_{pn}”) into slot 2 (or into slot 3 with the OpenCom 150). In every case, you need to run a U_{pn} interface card (“4 x U_{pn}”, “8 x U_{pn}” or “2 x S_0 and 6 x U_{pn}”) in slot 1 in order to operate DECT applications properly.

Note: The maximum cable attenuation between the OpenCom 130 / 150 and the other system (NT or telephone exchange) must not exceed 6 dB. This corresponds to a length of approx. 150 to 200 m, depending on the type of cable used.



Location of the S_{2M} port on the basic module of the OpenCom 130



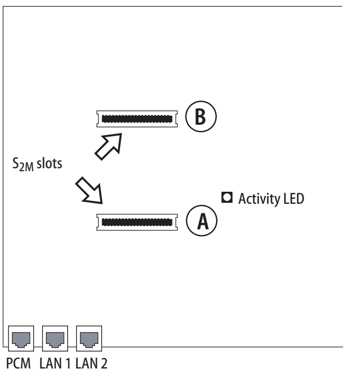
Location of the S_{2M} port on the OpenCom 150

5.1 Installation

Please note: The module should only be installed by trained personnel.

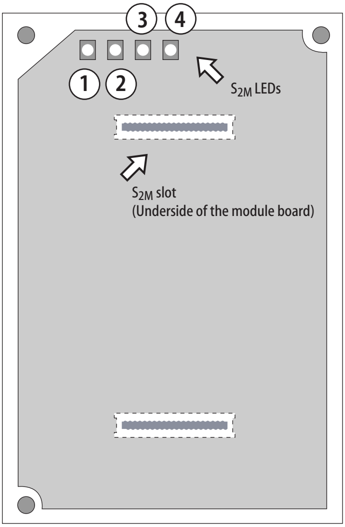
DANGER! The device contains hazardous voltages.

1. Pull out the mains plug of the OpenCom 130 / 150.
2. Open the housing cover of the OpenCom 130 / 150.
3. On the module board, remove the protective covers of the two S_{2M} slots (A) and (B).



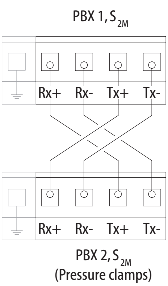
Position of the S_{2M} slots on the module board

4. Insert the S_{2M} module into the slots, making sure you insert the LED side of the module into the upper slot (B).



Position of the LEDs on the S_{2M} module

5. Wire the S_{2M} port of the connector module to the NT or the other PBX according to the following drawing. Make sure the RX and TX lines are crossed over (connect the RX lines of the OpenCom 130 / 150 to the TX lines of the other PBX).



Example of the wiring of the S2M port for two OpenCom 130 / 150s

On the OpenCom 150 the pressure terminal is a 5-pin connector. The additional terminal is used to connect the shielding of a shielded line. The shielding has to be stripped as short as possible (approx. 3 mm).

6. If necessary, check to make sure that the NT is fed by its own power supply.

7. Reconnect the OpenCom 130 / 150 to the power supply.

8. Configure the S_{2M} module in the **Configurator** of the OpenCom 130 / 150 (refer to *Configuration* starting on page 76). Check the level 1 status with reference to the LEDs on the module. None of the LEDs are illuminated before the S_{2M} module has been configured.

LED Display

LED	Meaning
1 (green)	Synchronous (= ok)
2 (green)	Blue alarm
3 (yellow)	Remote alarm or out of sync (yellow alarm)
4 (red)	Loss of signal (red alarm)

9. Pull out the mains plug of the OpenCom 130 / 150 again and close the housing.

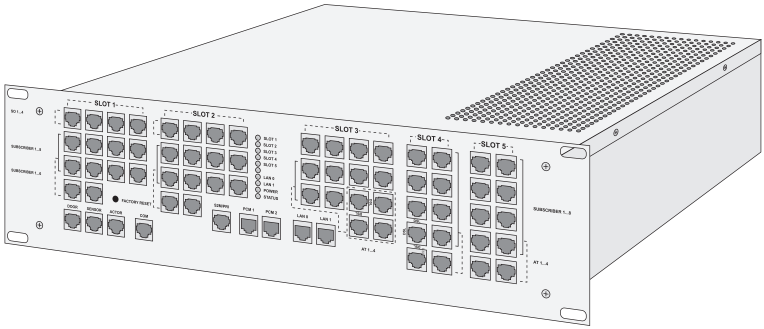
10. Reconnect the OpenCom 130 / 150 to the power supply.

5.2 Configuration

The menu item **S_{2M}** appears in the **Configurator**, menu **Telephony: Ports** after installation of the S_{2M} module. Configure the port in the input mask of this menu item. Further information on this can be found in the online help.

6. Mounting the OpenCom 150 Rack InfoCom System

The OpenCom 150 is also available as rack version for mounting in a standard 19" EIA rackmount cabinet.



OpenCom 150 Rack Frame and Ports

6.1 Safety Precautions

The system needs to be mounted in earthed cabinets or cases. Lines and cables connected to the communications system must only be laid inside buildings.

Use a shielded Ethernet cable (STP cable, Shielded Twisted Pair cable) to connect the OpenCom 150 to a Local Area Network (LAN).

The ambient temperature of the OpenCom 150 Rack infocom system should not exceed 55°C. If the device is installed together with other active components, it may be necessary to mount additional ventilation fans in the installation cabinet.

Patch cables have to be connected before connecting the system to the power supply. Installation of the system, and in particular connection to the power supply and protective earthing, should only be performed by skilled, qualified personnel. EN, IEC regulations, along with other recognised technical rules regarding safety, must be observed.

Please note: Before opening the device, pull out the plug.

6.2 Technical Data

(only if different from the OpenCom 150)

Dimensions:

- Width: 19-inch panel with flange for mounting in installation cabinet
- Height: 3U
- Depth: approx. 340 mm

Weight:
approx. 7.8 kg

Connection of 230 VAC power supply on rear side via inlet connector for non-heating apparatus

Connection of all ports via RJ 45 jacks on front panel

Note: The ISDN S_0 ports on the front panel may only be switched in one direction (internal or external). Use of the external port with a crossover (Rx-Tx) patch cable in parallel to the S_0 internal port (for two S_0 terminals on the same bus) may severely limit the range of the S_0 bus.

6.3 Pinning of RJ 45 Jacks

U_{pn}, a/b	Pin 4-5
Intercom	Pin 3-4 (Door 1) Pin 5-6 (Door 2)
Sensor	Pin 1-2 (S1) Pin 3-4 (S2) Pin 5-6 (S3) Pin 7-8 (S4)
Actor	Pin 1-2 (A1) Pin 3-4 (A2) Pin 5-6 (A3) Pin 7-8 (A4)
S_0	Pin 4-5, 3-6
S_{2M}	Pin 4-5, 1-2
100 Base-T	Pin 1-2, 3-6

For each of the 5 slots, there is a corresponding port field with several RJ 45 sockets on the front panel. The port fields are labelled “SLOT 1” to “SLOT 5”. Depending on the type of interface card used in a slot, the meaning of the corresponding RJ 45 sockets changes:

Label	Interface Card	Ports
S0 1..4 (Slot 1, 2 & 3)	2 x S_0 + 6 x U_{pn}	$S_0$1, $S_0$2
	2 x S_0 + 6 x a/b	$S_0$1, $S_0$2
	4 x S_0	$S_0$1 ... $S_0$4
SUBSCRIBER 1..8 (Slot 1, 2 & 3)	4 x U_{pn}	U_{pn}1 ... U_{pn}4
	8 x U_{pn}	U_{pn}1 ... U_{pn}8
	4 x a/b	a/b1 ... a/b4
	8 x a/b	a/b1 ... a/b8
SUBSCRIBER 1..6 (Slot 1, 2 & 3)	2 x S_0 + 6 x U_{pn}	U_{pn}1 ... U_{pn}6
	2 x S_0 + 6 x a/b	a/b1 ... a/b6

Label	Interface Card	Ports
SUBSCRIBER 1..8 (Slot 4 & 5)	4 x a/b	a/b1 ... a/b4
	8 x a/b	a/b1 ... a/b8
AT 1..4 (Slot 3, 4 & 5)	M100-AT4	a/b1 ... a/b4

An overview of the interface cards can be found under the heading *OpenCom 150 (5 Slots)* starting on page 46.

For each of the five slots, there is a LED on the front panel of the OpenCom 150. These LEDs are labelled **SLOT1**..**SLOT5**. An LED will show a constant light, if an interface card is inserted into the corresponding slot and the operating software has detected an interface card. An LED will blink, if an error condition was detected.

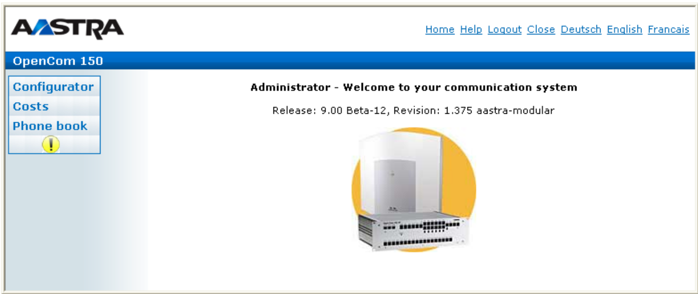
6.4 Scope of Delivery

- One communications system OpenCom 150 Rack
- One AC adapter with connection cable
- One set of short user guides
- One CD

Note: Note for the Aastra installer: Please download and install the latest released software from our Web site/ partnership area.

7. Configuration

Configuration and programming of the OpenCom 100 is performed by the **Configurator**, a special software application integrated into the system. The **Configurator** is operated via the Web console, which can be run on any PC connected to the OpenCom 100.



The OpenCom 100 Web console (this screenshot: OpenCom 150)

Using the Web console, you can:

- perform the initial configuration of the OpenCom 100,
- configure users of the OpenCom 100 and authorise them to use certain system services,
- carry out further system maintenance,
- use PC-supported telephony functions,
- read out call charge information,
- access the OpenCom 100 telephone book.

The Web console has an integrated online help function that offers comprehensive information on configuration and maintenance of the OpenCom 100 (see *Loading the Online Help* starting on page 81).

Note: In order to use all the new system software functions, we recommend that you download the latest software from our Web site at http://www.aastra.de or http://www.aastra.com.

For the initial configuration you can connect the PC to the OpenCom 100 via the Ethernet port. The TCP/IP network protocol is used to set up a connection via one of these ports. You can then open the Web console of the OpenCom 100 and call up the **Configurator** from there.

Note: To avoid problems with existing network installations, the OpenCom 100's DHCP server is designed for static address assignment in its factory settings. The OpenCom 100's IP address is always 168.99.254 in its factory settings.

7.1 Brief Guide to Initial Configuration

Setting up a first connection is quite simple with a standard Windows PC:

1. Connect the PC's network card with one of the OpenCom 100.'s LAN ports Use a cross-wired Ethernet cable to do this. You can also use an uncrossed Ethernet cable for a connection via a LAN port to the OpenCom 130's expansion module.

2. Windows 2000/XP: log on as a user with "Administrator" rights.

3. You will find the IP settings in Windows 2000/XP under **Start**: **Settings**: **Network connections**: **Local Area Connection**. Open the dialogue box **Local Area Connection Properties**, and then the dialogue box **Internet Protocol TC/IP Properties** (see figure: *Setting the IP address in Windows XP* on page 78).

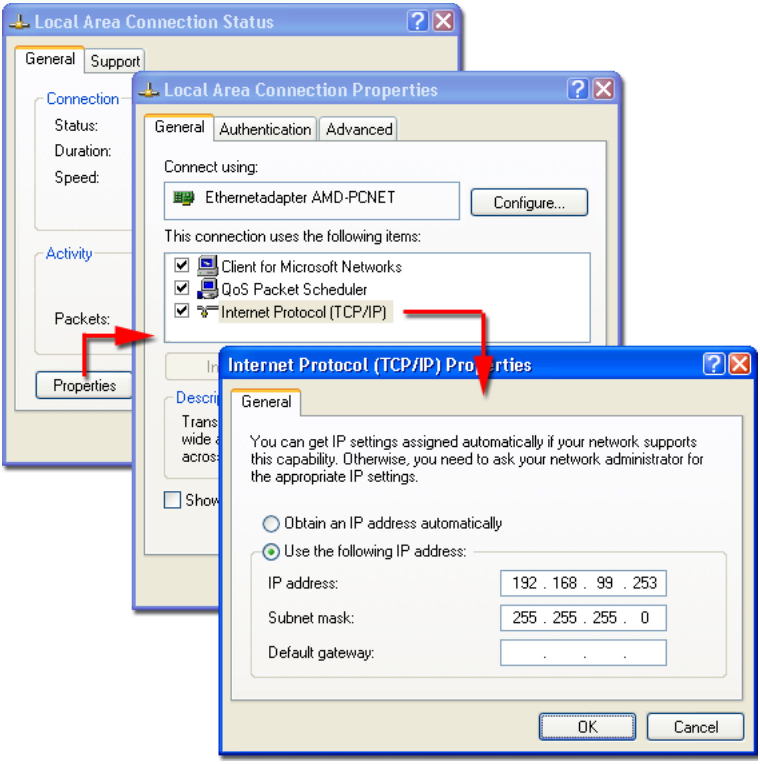
4. Note down the existing settings so that you can restore them after completing the initial configuration.

5. Change the **IP Address** to 192.168.99.253. Change the **subnet mask** to 255.255.255.0, confirm with **OK** and **Close**.

6. Start a Web browser and in the address field enter "http://192.168.99.254/".

The Web console's log-on page will be displayed. Enter the user name "Administrator" without a password for the initial configuration.

Note: To support your next configuration steps, you should activate the **Assistant** mode on the entry page of the **Configurator**. Please also pay attention to the online help.



Setting the IP address in Windows XP

Tip: To find out the IP address of the Web console, enter the code digit procedure *182 on a connected system telephone. You can also view the net mask by entering the procedure *183. The PC's IP address must be in this network range.

Note: Deactivate any connection via a proxy server which has been configured. Open the Internet Explorer, go to the menu **Extras** and open the **Internet options** dialogue box. Select the **Connections** register and deactivate the **Proxy Server**.

7.2 Configuring the OpenCom 100

7.2.1 Preparing the Configuration

Before starting with the configuration, make sure you have the following documents at hand:

- An overview of the ports
- A list of the terminals to be connected
- A list of the IPEIs, if you wish to log on DECT terminals in the secure procedure
- A list of the users to be set up (staff entitled to use the services of the OpenCom 100) with their names, departments, and the internal call numbers you want to allocate to them
- For Internet access: the Internet service provider access data.

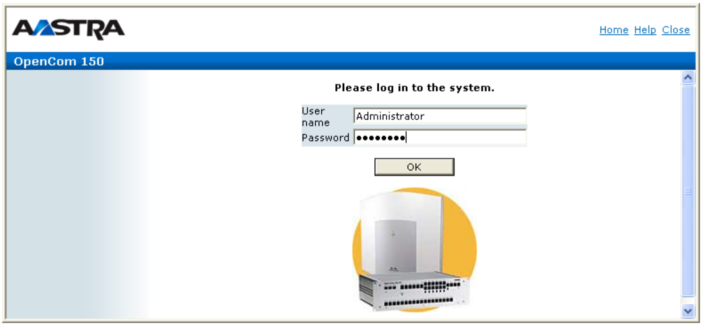
Data not available for initial configuration can be updated or corrected at a later date.

7.2.2 Starting the Web Console

1. Start your Web browser. Enter the OpenCom 100 IP address in the “Address” box: http://192.168.99.254/.

   If the configuration PC gets its IP address automatically from the OpenCom 100 or if the OpenCom 100 is entered as the domain name server, you can also start the Web console by entering the DNS name. The DNS name in the factory setting is **host.domain**. You can change this in the **Configurator** (**Network: LAN** menu).

2. The OpenCom 100 Web console is started. First set the country in which you are operating the OpenCom 100, and in which language the Web console is to be displayed.

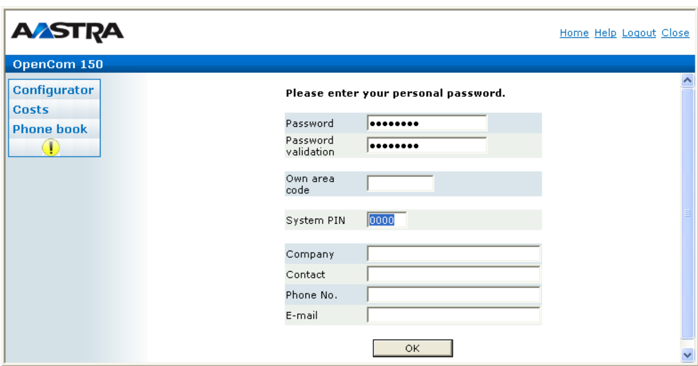


OpenCom 100 (this screenshot: OpenCom 150): log-on dialogue box

3. To commence configuration, you must first log on. For the initial configuration, enter your:

   user name: “Administrator”
   password: for the initial configuration, leave this box blank.

4. Confirm this by clicking on **OK**. This puts all connected terminals into the “Guest” user group with restricted user rights. In this way you prevent international external calls from the terminals, for example, while you are configuring the OpenCom 100 and the users.



OpenCom 100 (this screenshot: OpenCom 150): dialogue box for initial access

5. The software opens a dialogue for initial access. Determine an administrator password and enter it in this dialogue. Also fill in the other input fields.

6. Confirm your input with **Apply**.

7. Click on the **Configurator** button on the home page.

You will find notes on using the **Configurator** and in the online help. Click on **Help** in the menu bar or click on **TOC** to activate an overview of help topics.

7.2.3 Loading the Online Help

The online help can be loaded in the **Configurator**:

1. Go to the **System: Components** menu. Select the entry **Online Help** and click on **Browse**.
2. Look for one of the language-specific ZIP files in the "OLH" directory of the product CD. Confirm your choice by clicking on **Open**.
3. Then click on **Load** to transfer the online help to the system.

Please note: After completion of the loading operation, the system will take a few minutes to analyse the transferred file.

Note: You can download the latest version of the online help from http://www.aastra.de or http://www.aastra.com.

7.2.4 Finishing the Configuration

1. When you have completed all the settings in the **Configurator**, you must save the configuration (see also *Saving and Loading the Configuration on page 81*).
2. Then select the **Log-off** command in the upper menu bar.

7.2.5 Saving and Loading the Configuration

Configurations are saved in a file archive and can be loaded to the OpenCom 100 either locally from a connected configuration PC, or by remote configuration. The following configuration and customer data can be saved and loaded again:

- Telephony and network parameters
- User data

- Telephone book entries
- LCR tables

Note, that for security considerations the passwords for external SIP trunks are not re-established if you restore a data backup from another (foreign) communication system. For further information, refer to the online help documentation under the topic **System: Data backup**.

7.2.6 Preconfiguration

Configuration of the OpenCom 100 can be prepared at your Aastra Customer Service Centre or by an authorised Aastra dealer. For this purpose, a OpenCom 100 installed here is programmed with the customer data (e.g. user data, call distribution schemes, cord-bound terminals). This data is stored and then loaded into the OpenCom 100 at the customer's site by a service technician.

This prepared configuration must be completed at the customer's site (LAN configuration and DECT terminals).

For configuration of the OpenCom 100 Internet functions, first ask the responsible system administrator for details of the customer's LAN prerequisites.

7.2.7 Offline Configurator

With the aid of the offline configurator a configuration for the system can be issued and created on a Windows PC. Thereby most of the configuration points are available. Each system type of the product family and firmware version release 7.0 or higher, has its own offline configurator; this is managed with the aid of a starter programme. The starter programme is included on the product CD. The operating systems Windows 2000 and XP are supported.

You can find further information in the chapter *PC Offline Configuration* starting on page 208.

7.2.8 Remote Configuration

The OpenCom 100 configuration can be edited or updated by the customer service centre or authorised dealer via a remote access connection or via an Internet connection (reverse ssh tunnel).

Connection via ISDN

The connection for remote configuration is established using an internal RAS access. The RAS access is activated when a data call from one of the registered phone numbers is registered. During remote configuration, the OpenCom 100 is blocked for RAS access by any further users.

If no RAS access has been configured in the OpenCom 100 Configurator, you can use a code procedure to release the internal RAS access for remote configuration, and then block it again. The manual activation is automatically cancelled 30 minutes after the last configuration activity. You can add additional phone numbers for remote configuration on the configurator's **Telephony**: **Extended**: **Remote service** menu page. Activate the **ISDN** option for the desired phone numbers.

Connection via Internet

The connection for remote configuration is established indirectly over the Internet. The connection session starts with a voice call from one of the registered phone numbers. During the call, connection parameters (IP address, keys) will be sent using a DTMF sequence. Afterwards, the OpenCom 100 communications system establishes a secured SSH connection to an Internet server. The existing SSH connection allows to reverse-connect to the OpenCom 100 communications system's configurator. For this type of connection, the OpenCom 100 communications system utilizes your Internet account. Establishing an SSH connection via this Internet account must be allowed for this purpose.

The manual activation of the remote configuration activates the DTMF evaluation for the next voice call. You can add additional phone numbers for remote configuration on the configurator's Telephony: Extended: Remote service menu page. Activate the IP option for the desired phone numbers.

Note: You also need to activate the desired type of connection directly on the **Telephony**: **Extended**: **Remote service**.

Manual Activation

The following code number procedures activate or de-activate the remote configuration access. These code digits can be entered on any standard terminal or system telephone.

Remote Configuration On (Log-on with Administrator Password)

* 1 9 * (system PIN) #

Remote Configuration On (Log-on with Temporary Password)
* 1 9 * (system PIN) * (temporary password) #

Remote Configuration Off
1 9

Please note: The system PIN is preset to “0000” and it is absolutely imperative that the system administrator changes it to prevent undesirable remote maintenance.

The customer service centre/authorised dealer can log into the OpenCom 100 communications system as an administrator:

- User name: “Administrator”
- Password: [administrator password]

Note: If you do not wish to let the customer service centre/authorised dealer know the administrator password, you can define a temporary password for remote configuration with at least five digits.

Using remote configuration, all OpenCom 100 settings with the exception of the system PIN can be edited or updated. New software versions of the OpenCom 100 and the software for the connected system terminals and base stations can also be installed (see the **System: Firmware** menu in the **Configurator**).

For security reasons, settings in the **Configurator**, **Network** menu should only be edited on site to avoid malfunctions or failures in the customer’s LAN (e.g. due to IP address conflicts). Refer to the chapter entitled *Configuration Examples* starting on page 90, where interaction between the OpenCom 100 and a LAN is explained.

Forced Logoff of Another User by the Administrator
If the user “Administrator” logs in and another user with administration rights is already logged in, then the administrator can forcibly log the other user off in order to configure. This functionality, for example can be used, when configuring remotely, when a user has forgotten to log out.

In order to forcibly log a user off:

1. The “Administrator” user logs-in with the administration password.

2. They open the **Configurator**.

   A message shows which user is currently configuring the system.

3. The administrator clicks on the **Take over config rights** button.

   The other user can make no further modifications of the configuration.

7.2.9 Codes for IP Configuration

The IP configuration of the OpenCom 100 is performed on the Web console in the **Configurator**, in the **Network: LAN** menu.

In the event that the IP configuration of the OpenCom 100 has to be changed and access via the Web console is not possible, you can also use a code digit procedure to change these basic settings. Entry can be made from an analogue telephone, an ISDN telephone and from system telephones.

Set IP Address
* 1 8 2 (system PIN) *
(www) * (xxx) * (yyy) * (zzz) #

Set NET Mask
* 1 8 3 (system PIN) *
(www) * (xxx) * (yyy) * (zzz) #

Example
Enter: * 1 8 2
0 0 0 0 * 1 9 2 * 1 6 8 * 9 9 * 2 5 4 #
If required, initiate a system restart with the following procedure:

Trigger Restart
* 1 8 5 (system PIN) #

Use the PIN you entered in the dialogue box for initial access. The factory setting is “0000”.

7.2.10 Receiving System Messages as E-Mail

Important events and errors are kept by the OpenCom 100 in an internal log book: the error store. To inform or alert the system administrators, entries in the log book (system messages) can be sent via e-mail.

In order not to notified of every error, the administrator can define corresponding log filters (in the **Configurator**, the **Diagnosis: Filter** menu). These filters define which errors (category, severity, number per time interval) should be notified. The e-mails always include an internal event or error number, as well as an explanation of the message. Further, extra parameters (such as the port number when a trunk line drops out) are also provided.

The mail account for this service (**Account for LOG filter**) is configured in the **Configurator**, **Network: E-Mail** menu.

7.2.11 Loading SW Updates

New versions of the system and terminal software can be loaded to the system.

New software versions of the OpenCom 100 are loaded from the configuration PC, which accesses the **Configurator** (see the **System: Firmware** menu). For information on connecting a configuration PC, see *Brief Guide to Initial Configuration* on page 77.

The terminal software is part of the OpenCom 100 software and is automatically loaded into the terminals via the OpenCom 100 if the software version in the terminal is different from the terminal software stored in the OpenCom 100.

Note: If you are operating an PBX cascade, new system software is automatically transferred to the slave system from the master system.

For further information, refer to the online help documentation under the item **System: Firmware**.

7.2.12 Resetting the System Data

You can restore the factory settings of the OpenCom 100 in the Configurator. If this is not possible, refer to the next section entitled *Basic Hardware Settings Switch*.

Please note: If this is done, all individual settings and the user data are then lost. For this reason, you should back up your configuration regularly, the best time to do so being after every change. For details, refer to the chapter entitled *Saving and Loading the Configuration* starting on page 81 and to the Web console online help.

Proceed as follows:

1. In the Configurator, call up the **System**: **Restart** menu.
2. Click on **Restart with Defaults**.
3. Confirm this by pressing “OK” when the query dialogue box opens.

7.2.13 Basic Hardware Settings Switch

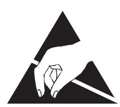
The OpenCom 100 configuration can also be returned to the factory settings by means of the basic hardware settings switch.

Please note: If the factory settings are restored, all customer settings and user data will be lost.

To restore the OpenCom 100 basic settings, proceed as follows:

1. Switch off the OpenCom 100 by disconnecting the power plug and the plug-in power supply of the basic module (OpenCom 130).
2. Remove the cover.

CAUTION! Static charges can damage electronic devices. Observe the regulations regarding electrostatically sensitive components.



3. The basic settings switch is designed as a key switch. The location of the switch can be found in the chapter entitled *Interface Cards* starting on page 43. Press and hold the switch.

4. Replace the power plug in the mains socket. Wait about 30 seconds until the indicator on the front of the OpenCom 100 constantly flashes.

5. Disconnect the power plug from the mains socket again.

6. Release the key switch.

   The system data is now reset.

7. Replace the power plug in the mains socket.

   The OpenCom 100 will now reboot in the default configuration. The procedure is completed when all connected system terminals show the time on their displays.

8. Log on to the Web console (see *Starting the Web Console* on page 79). Configure the OpenCom 100 (possibly by loading a saved configuration; see *Saving and Loading the Configuration on page 81*).

7.2.14 Generating Your Own MoH Files

The OpenCom 100 comes with an internal MoH file for Music on Hold. The OpenCom 100 product CD contains a number of MoH files with different volume levels, which you can load at a later time as necessary.

The file format for non-resident Music on Hold is *.wav. You can also save your own MoH in a *.wav file and load it into the OpenCom 100.

If you have a Windows operating system, you can use the “Sound Recorder” programme to generate your own MoH file. This programme is usually located in the Windows directory called “Multimedia”.

The MoH file must be coded with 8000 Hz, 8 bit mono in accordance with CCITT, A-Law. This coding is required for the OpenCom 100 and can be set in the “Sound Recorder” when you save the file under **Format** (CCITT, A-Law) and **Attributes** (8000 Hz, 8 bit mono). The maximum allowable size for a MoH file is 256 KB (approx. 32 sec. play time). If a larger file is loaded then this will be “truncated” and thereby will also only be played for 32 seconds. The MoH capacity can be subdivided in a maximum of 5 files. These files can be used for different companies or for internal and external calls.

Note: If you don't have the Sound Recorder programme or the appropriate codec on your Windows operating system, you should install these components from your Windows CD.

Load your MoH file in the Web console's **Configurator**, in the **System: Components** menu.

Note: When generating your own MoH file, you may incur a fee for the use of non-resident melodies (e.g. a GEMA fee in Germany or MCPS fee in the UK). The MoH files that come with your OpenCom 100 can be used free of charge.

8. Configuration Examples

8.1 OpenCom 100 in Computer Networks

One of the outstanding features of the OpenCom 100 is the integration of telephony and computer networks. Connect the OpenCom 100 via a computer network (LAN) with suitably configured workstations, and you can use its network features from these workstations. Using a Web browser you can access:

- the OpenCom 100 Configurator
- call charge administration
- the OpenCTI 50, with which telephone functions can be used on a PC
- the OpenCom 100 central telephone book and your personal telephone book as well as to the company telephone book (if the multi-company variant is activated).

In addition, the OpenCom 100 can be used as an Internet access server. RAS access can also be implemented using the OpenCom 100, which enables the integration of external staff in the LAN.

In this chapter you will find several examples of configurations showing integration of the OpenCom 100 in a LAN. Which example applies to your situation depends on the size and properties of the existing or planned LAN infrastructure.

Note: Several menu entries mentioned in this chapter are available only, if you switch on the **Level: Expert** in the top level dialogue of the **Configurator**.

The following LAN prerequisites are possible:

Server configuration in the LAN	OpenCom 100 Functions
No IP server present	OpenCom 100 functions as DHCP and DNS server
IP server present DHCP server present	System Administrator must assign IP address and DNS name for OpenCom 100

Server configuration in the LAN	OpenCom 100 Functions
IP server present No DHCP server present	Special case when integrating the OpenCom 100 in a LAN; settings in the **Network: LAN** menu must be coordinated with the responsible system administrator

8.2 Introduction to TCP/IP

In a single LAN it is possible to use various protocols for the transmission of data. The connection between a workstation computer and the OpenCom 100 runs via the IP protocol (also named TCP/IP) used on the Internet. IP can be used together with other protocols (e.g. NetBEUI, AppleTalk or IPX/SPX) on the same network.

Every device participating in data transmission using IP requires a unique IP address. An IP address consists of four groups of digits from 0 to 255, each separated by a full stop. The supplementary protocols DHCP and PPP automatically assign IP addresses to devices. Class C networks normally use IP addresses in which the first three numbers are the same and the last number is uniquely assigned to a specific device in the LAN. On the Internet, unique addresses assigned by a special organisation created for this purpose are used. Within a LAN, you can use addresses which are not unique world-wide:

IP Range	Common Netmask	Comment
192.168.0.0-192.168.255.255	255.255.255.0	256 smaller networks
172.16.0.0-172.31.255.255	255.255.0.0	16 medium networks
10.0.0.0-10.255.255.255	255.0.0.0	1 large network

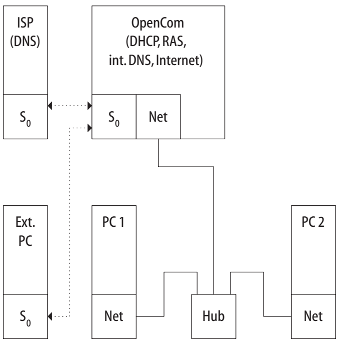
IP enables the establishment of connections via one or more intermediate stations. The decision whether to connect directly or indirectly to the partner device depends on the network mask. The network mask for a class C network is 255.255.255.0. If the IP address of the partner device does not fit the network mask, the connection is established via the default gateway. If a device knows several data routes to different intermediate stations, one speaks of a router.

The domain name system (DNS) resolves a plain text DNS name into an IP address. The DNS is a hierarchically structured database, distributed worldwide. A DNS server can supply information on the names and IP addresses for which it is responsible. For all other information, a DNS server contacts other DNS servers. For the establishment of every connection from the workstation, it is possible to give either an IP address, or a name that a DNS server resolves into an IP address.

Note: For further explanations of technical terms, refer to the Glossary on the CD supplied.

8.3 OpenCom 100 in a Serverless LAN

In a peer-to-peer network, the workstations are connected to one another via network cables. In many networks, the cables run in the form of a star from a central hub or switch. Such networks do not require special servers. This configuration example is also valid for a LAN with a server using a protocol other than IP (e.g. AppleTalk or IPX/SPX).



The OpenCom 100 in a serverless LAN

In a serverless LAN, the OpenCom 100 takes over the IP configuration of the connected workstations. All IP settings necessary for the workstations are assigned by the OpenCom 100 via DHCP (dynamic host configuration protocol). In this operating mode, an IP address space reserved for such networks is used:

192.168.99.254	OpenCom 100 IP address
255.255.255.0	Network mask (class C network)
192.168.99.254	DNS server IP address
192.168.99.254	Default gateway IP address

Install the IP network protocol and a Web browser for every workstation computer which is to have access to the OpenCom 100 network features.

8.3.1 DNS Name Resolution

In a serverless LAN, the internal DNS name resolution is performed by the OpenCom 100. If you type the string “host.domain” into your browser, a DNS request is sent to the OpenCom 100 IP address. The OpenCom 100 responds with the correct IP address, so that the **Configurator** home page can be called up.

In a peer-to-peer network (Windows network), the workstations each have a name which is displayed in the network environment. These NetBIOS names can differ from the DNS names assigned to the workstations by the OpenCom 100. The OpenCom 100 is not visible in the network environment.

8.3.2 Internet Access

If access to an ISP has been configured on the OpenCom 100, the OpenCom 100 can be operated as an Internet access server without any additional configuration of the workstations. When you want to see a Web page, you simply type the URL (uniform resource locator; Internet address; “http://...”) in your browser. In a serverless LAN, the OpenCom 100 is configured as a DNS server and default gateway. The workstation computer therefore sends its Internet connection request to the OpenCom 100.

In almost all cases, the request will contain a DNS name which is unknown in the internal network. When you type a URL into your browser, the OpenCom 100 receives the request to find the corresponding IP address. If the name is unknown in the LAN, the request is forwarded to an ISP’s external DNS server.

Note: Workstation computers automatically add a domain name to URLs without a dot. You specify this domain name in the **Configurator**. For example, if you have configured “firm.co.uk” as the domain name, an access request for “www.firm.co.uk” will be interpreted as a local DNS request which does not lead to the establishment of an Internet connection. For this reason, you should choose a name which is not used in the Internet as the domain name (“firm-opencom.co.uk” for example).

8.3.3 RAS Access

You can establish a connection to the OpenCom 100 from an external PC via an ISDN card.

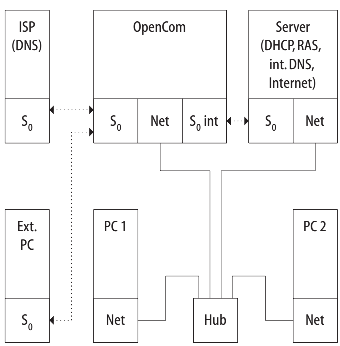
The necessary IP settings are transmitted by the OpenCom 100 on establishment of the connection. The computer that has dialled in has access to all services in the LAN that can be used via the IP protocol. The authorisation for RAS access is set up in the **Configurator** via the **User Manager: User Groups** menu.

The technical properties of the connection can be configured in the **Configurator** via the **Network**: **RAS** menu. Further information can be found in the online help of the web console.

In a serverless LAN, Windows uses the NetBIOS protocol for accessing files and printers via the network environment. NetBIOS can use NetBEUI, IPX/SPX or IP as the transport protocol. In the network environment, you can only access files and printers on workstations using IP for NetBIOS.

8.4 OpenCom 100 in a LAN with an IP-enabled Server

In a LAN with an IP-enabled server, you should coordinate integration of the OpenCom 100 with the responsible network administrator. You must decide on the IP address space to be used and which network services (DHCP, DNS, RAS, Internet access) the OpenCom 100 is to handle in the LAN.



The OpenCom 100 in a LAN with an IP-enabled server

In many cases, an IP-enabled server configures the IP settings via DHCP for all workstations. In networks in which the IP settings are made manually, you have to enter the corresponding IP settings in the OpenCom 100 **Configurator** (**Network: LAN** menu). Additionally you should change the DHCP server to static address assignment (in the **Network**: **DHCP** menu) to enable the OpenCom 100, for example to configure connected VoIP system telephones. In certain cases you have to restrict the DHCP function of the IP-enabled server to ignore the MAC addresses for such terminals.

Dynamic Address Assignment for Specific Devices

In addition to the static address assignment, you can also use dynamic address assignment if you limit dynamic assignment to specific devices. You can use this e.g. to easily configure VoIP devices because you do not need to assign a fixed IP address during setup.

1. Call up the **Network**: **DHCP** page. Click **Change**.

2. In the **Status** selection, change to the **Dynamic address assignment** or to the **Dynamic and static address assignment** setting.

3. In the **Devices** selection, change to the **with configured MAC only** setting.

   For SIP system telephones (Aastra 673xi and Aastra 675xi), it is also possible to omit the configuration of the MAC address (see *Aastra 673xi/675xi DHCP* starting on page 124). In the **Devices** selection, change to the **with configured MAC only and SIP system devices** setting. Please note, that you also need to exclude from the configuration of an external DHCP server all MAC addresses starting with 00:08:5D.

4. Confirm with **Apply**.

By activating the limitation to specific devices, only known devices can get a DHCP answer. For this, the IP address is taken from the address range used for dynamic address allocation. If you setup a new VoIP device, it is sufficient to enter only the MAC address.

8.4.1 DNS Name Resolution

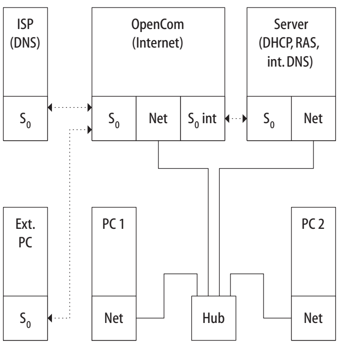
In a LAN with an IP-enabled server, the latter is also responsible for DNS name resolution. If you want to start the **Configurator** by entering a DNS name, you must link this name on the server with the IP address used by the OpenCom 100. For further information, refer to the server documentation.

Note: To access the OpenCom 100 under the same IP address after a restart, you must specify this IP address permanently on a DHCP server. On a DHCP server it is possible to link the MAC address of a network card with a specific IP address. You will find details in the server documentation.

8.4.2 Internet Access

You can also use the OpenCom 100 as an Internet access server in a LAN with an IP-enabled server. To do this, you must enter the OpenCom 100 IP address on the server as the default gateway. In addition, you must edit the internal DNS server configuration so that the resolution of external DNS names is forwarded to the OpenCom 100.

In this example, the Internet connection is established from a workstation computer via the server, which in turn requests Internet access from the OpenCom 100.



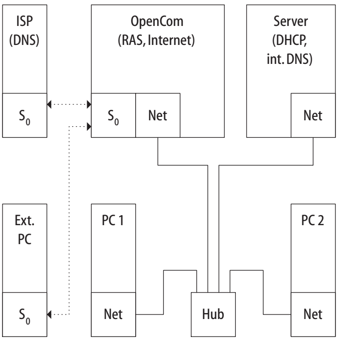
The OpenCom 100 as a DNS server in a LAN with an IP server

There are two different ways of suitably configuring the internal DNS server. You can enter the OpenCom 100 IP address as a DNS forwarder. If you require access to extended DNS information, you can also configure the DNS server for a recursive

DNS request without the DNS forwarder. For further explanation, refer to the DNS server documentation.

8.4.3 RAS Access

In a LAN with an IP-enabled server you can also enable external computers to dial in via the OpenCom 100. To do this, you should coordinate with the network administrator the IP address space which can be assigned to an external computer dialling in, and enter it in the **Configurator**, **Network: RAS**: **ISDN** menu, under **Address Range**.



RAS access by the OpenCom 100 in a LAN with an IP server

The user account administered by the OpenCom 100, with which dialling in is permitted, only allows the establishment of direct and anonymous TCP/IP connections such as HTTP, FTP or SMTP connections. If you additionally want to allow file or printer access in the network, you must set up a suitable user account on the addressed server for network log-in. If you use the same log-in name for the OpenCom 100 user account and the same password for the network log-in, you have to enter this combination only once when dialling in.

Note: In a larger Windows network with several segments, the lists of computer names visible in the network environment can no longer be established by broadcasts. In this case you use a special WINS server whose address the OpenCom 100 does not make known to the workstation computer when dialling in with ISDN. For this reason, you enter the address of a WINS server manually in the network settings of the workstation.

8.5 Branch Link

You can use the OpenCom 100 to interlink two LANs via ISDN.

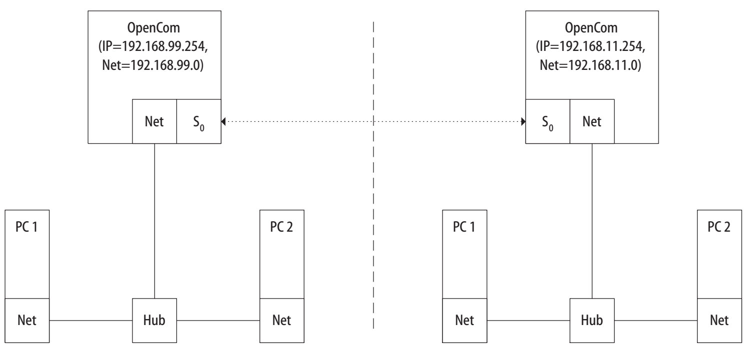
To do this, you configure two OpenCom 100 systems so that they can dial in to each other.

In order for this to work, the two LANs must be configured for different IP address ranges (subnetworks). For at least one of the OpenCom 100 systems, change the prescribed address range for the LAN.

In the **Configurator**, **Network: Branch** menu you can configure the dial-in settings. The OpenCom 100 will set up a connection whenever a IP data transfer to the other LAN is requested.

Note that such a connection is only set up when specific requests are made. These can be for FTP file transfers, e-mails or downloading Web pages. Name resolution via broadcasts is not possible. If you wish to use the LAN-to-LAN link to access files and printers in the Windows network, you need an IP-enabled server that administers the name resolution for the Windows network.

As the IP address range, you can select one of the 256 class C subnetworks designed for local LANs. Select a class C sub-network in the range from 192.168.0.0 to 192.168.255.0.



The OpenCom 100 in a LAN-to-LAN link

8.6 Useful Information on Internet Access

8.6.1 Costs

The OpenCom 100 uses a router function to access the Internet, which means that it automatically establishes an Internet connection when required and terminates the connection after a certain period of time if no data are being transmitted.

Unfortunately, programmes other than those typically intended to access the Internet (such as your browser or your e-mail software) may send out data packets which cause an Internet connection to be established, even if these programmes are not strictly Internet-associated applications. Examples of such programmes are the Microsoft™ XP™ operating system, various multimedia programmes such as Realplayer™ and anti-virus applications that may establish an Internet connection for automatic updates (the so-called “phone home function”).

It is therefore highly advisable to limit ISP access by specifying the maximum monthly connection time under **Connection time per month** in the **Network**: **WAN**: **[Provider]** menu on the web console.

8.6.2 Using the Web

A Web browser not only enables you to use the OpenCom 100 **Configurator** from every workstation computer but also to obtain a wealth of information from the Internet. Simply enter the desired URL in the address field of the browser. Access from a stand-alone PC via an online service differs from Internet access via the OpenCom 100 in the following respects:

- When you request a Web page, dialling in results automatically. There is no display of dialogues with manual confirmation of dialling in or hanging up.
- Requesting Web pages is not a connection-orientated service. When the Web page has been loaded completely, the TCP/IP connection is cleared. If you do not request further Web pages, the OpenCom 100 automatically releases the connection to the Internet after a certain, specifiable duration.
- It is possible to call up Web pages simultaneously from several workstations.
- The OpenCom 100 can block access to certain Web pages by means of filter lists.

8.6.3 E-Mail

One of the most important services in the Internet is e-mail. E-mails are buffered in individual e-mail accounts on a mail server. Mail servers are operated by ISPs for example. With the OpenCom 100 you can set up one or more e-mail accounts for every user account configured on the OpenCom 100. These e-mail accounts are then checked at regular intervals.

If there are new e-mails in an e-mail account, and the OpenCom 100 has been configured for this function, the user specified in the OpenCom 100 user account is notified of the new e-mail on his system terminal. Aastra 677x (OpenPhone 7x) and Aastra 610d / 620d / 630d system terminals can also display information such as the sender or the subject of the e-mail.

8.6.4 NAT

Network address translation (NAT) is activated on accessing the Internet (ISP). You require this feature in order to translate internal IP addresses to valid external IP addresses. This has three important consequences for Internet access:

- Several workstations can share a single Internet access. You do not require a LAN access, only a single account with the Internet service provider.
- The IP addresses used in the LAN are translated into IP addresses valid worldwide. So you require no such addresses for your LAN.
- Only IP connections triggered from a workstation computer can be established. Consequently, while you can call up Web pages from a workstation, you cannot install a Web server visible in the Internet on a workstation.

Certain protocols cannot be used when NAT is being used. This affects protocols with the following properties:

- IP addresses are transported in the useful load, e.g. NetBIOS over IP or SIP.
- The protocol requires an active, inward-directed connection establishment, e.g. ICQ.
- The protocol will function without TCP/UDP port numbers, e.g. ICMP or IGMP.

The OpenCom 100 NAT has suitable processes for ensuring the functions of many important protocols affected by these rules. These are the protocols FTP (in “active” mode), CuSeeMe (“videoconferencing”), IRC (“chat”), ICMP errors (“traceroute”) and ICMP echo (“ping”).

Depending on the internet telephony protocol (VoIP, SIP) the required NAT extension (“Full Cone NAT”) or RTP-Proxy is activated on the Media Gateway Card.

Protocols which require inward-directed connection establishment can be configured in the **Network**: **Port Access** menu. For further information, refer to the online help of this menu.

9. Voice over IP (VoIP)

The term “Voice over IP” describes the usage of IP-based data networks for telephony. It is possible to distinguish between two different types of VoIP:

- Telephony via Internet provides cheaper charge-models for telephone services. For telephony directly via the Internet, only the cost of data transmission is incurred. Various Gateway providers can provide crossover into the PSTN (“Public Switched Telephone Network”) for a fee. As well as standardised protocols such as SIP and H.323, proprietary protocols such as the Skype network, are used. Voice and service quality via Internet is often indeterminable because they are dependent on the communication lines of various service providers. which have been optimised for data communication

- Telephony via Intranet enables joint usage of existing infrastructure for telephony and for data communication. Integrating the two communication networks into a single communications network can provide considerable savings. The OpenCom 100 gives users all the features of system telephony through its use of an IP-based protocol. Furthermore, the standardised SIP protocol can also be used on the intranet. The control of the data connections used makes it possible to define exactly the voice and service quality.

VoIP telephony over the Internet using the OpenCom 130 / 150 provides you with the following options (see also *SIP Telephony* starting on page 115):

- You can use low-cost “SIP trunk lines” with your existing Internet connection

- You can use the services of a SIP gateway service provider to access the public telephone network (PSTN)

- Automatic fallback (bundle overflow) to ISDN connections in case of the breakdown or over-occupancy of the SIP connection

- With a M100-AT4 interface card (4 analogue trunk lines, also available for OpenCom 131) you can also use the external IP Gateway 1 for SIP telephony

VoIP Telephony via intranet with the OpenCom 100 offers the following possibilities:

- Use of IP-based system telephones and of SIP telephones connected to Cat5 twisted-pair ethernet cables

- Use of IP-based system telephones and of SIP telephones via VPN, RAS, Branch or WLAN connections
- Using voice-data compression with compressing codecs, it is also possible to make multiple IP-based telephone calls simultaneously on a 64 kbit/s ISDN line
- Use of PC-supported system telephones (so-called “Softphones”) without extra hardware costs
- Operation of SIP-capable telephony software (see also *SIP Telephony* starting on page 115)
- TC system networking using Q.SIG-IP via VPN connections (see also *PBX Networking* starting on page 153)
- Setting up a DECToverIP network lets you use existing Ethernet cabling to set up a DECT network. The special DECT base stations designed for this purpose, can be handled using OpenCom 100’s Web interface (see *DECT over IP®* starting on page 138).

Integrating voice and data communication within the Intranet can provide savings possible and a range of new possibilities. However joint usage of existing network infrastructure may also cause conflicts, with IP address configuration via DHCP for example (for details see *Start Procedure* starting on page 129). You should therefore always plan the use of VoIP in the Intranet together with your network administrator. In order to avoid possible conflicts please also note the information under *Fundamentals* starting on page 107.

9.1 Quick Start

9.1.1 IP System Telephony

VoIP system telephony can be quickly and easily set up using the OpenCom 100.

1. To increase performance, install an M100-IP Media Gateway card (not applicable to OpenCom 131).
2. Call up the Configurator and go to the page **Telephony**: **Ports**: **Slots**. Click on the corresponding slot. Under **Configured,** select **MGC VoIP.** Optional: enter

an IP address from the OpenCom 100 IP network which is not being used, such as 192.168.99.253 under **IP Address Configured**. Click on **Apply**.

3. Install a memory card (MMC or MMCPlus). Call up the **System: Components** page. Select the “Firmware Addons” option. Click on **Browse** and select a “fwaddons_*.cnt” file from the installation CD. Click on **Load** to load the operation software for the VoIP system telephones to the memory card.

4. Go to the page **Telephony**: **Devices**: **VoIP Telephone** and click on **New**. Enter the **MAC Address** printed on the underside of the IP system telephone. Select the **Type** and enter an internal **Number**. Option: enter an IP address from the OpenCom 100 IP network which is not being used. Click on **Apply**.

5. Connect the IP system telephone’s LAN connection to the LAN and connect the phone to the mains power using the power supply unit provided.

Once your IP system telephone has been successfully started, you can set it up and use as you would any other U_{pn}-based system telephone.

Note: Use a shielded CAT-5 Ethernet cable (STP cable, Shielded Twisted Pair cable) to connect an IP telephone to a Local Area Network (LAN).

9.1.2 External SIP Line

If your OpenCom 130 / 150 has access to the Internet, you can an easily and quickly set up an SIP line.

Note: The OpenCom 131 cannot be operated with a Media Gateway card. This is why SIP telephony is not supported.

1. Request at least one SIP account from an SIP provider.

2. Install a Media Gateway card (see *MGW Interface Card* starting on page 114).

3. Call up the Configurator and go to the page **Telephony**: **Ports**: **Slots**. Click on the corresponding slot. Under **Configured,** select **MGC VoIP.** Optional: enter an IP address from the OpenCom 100 IP network which is not being used, such as 192.168.99.253 under **IP Address Configured**. Click on **Apply**.

4. Call up the Configurator and go to the page **Telephony**: **Trunks**: **SIP Provider**. If your SIP provider is not listed, click on **New**. Otherwise select the preconfigured SIP provider. Enter the **Name** and **Domain** (DNS name of the SIP ID). Enter the SIP server's IP address under **Proxy/Registrar** and an IP address under **STUN Server** and **STUN Port** where necessary. You can obtain more information on this from your SIP provider. Click on **Apply**.

5. On the **Telephony**: **Trunks**: **SIP Trunks** page, click on **New**. Activate the **Status** and enter a name for the account under **Name**. Select the **SIP Provider**. Enter the relevant account information under **User name**, **Password**, **Phone No.** and **SIP ID**. Click on **Apply**.

   The **SIP ID** setting will be used while logging in to the SIP provider. Die **Phone No.** setting denotes the external phone number used within the public phone network. You can enter this number here to support system administration.

6. Call up the **Telephony**: **Trunks**: **Route**: **New** page once again. Enter "SIP", for example, under **Name**, under **Code** the number "8" and select the SIP account that was just configured for **Bundle/SIP trunks 1**. The SIP account is now available with the dialling prefix "8". To use the SIP account by default, call up the page **Telephony: Trunks: Route** and select the route **External trunk.** Under **Bundle/SIP trunks 1,** select the SIP account you have just set up. Click on **Apply**.

Note: The **Telephony**: **Trunks** menu page is only displayed if you activate the **Level**: **Expert** option on the opening page of the **Configurator**.

7. Check that the SIP connection is active on the **System Info**: **Telephony**: **Trunks** page. Also check the SIP licence count on the **System: Licences** page.

Check the functionality by making an external call. You should assign the relevant external number of the SIP account to the internal numbers on the page **Telephony: Call distribution: Incoming**.

9.1.3 Internal SIP Telephony

SIP telephones connected via LAN or SIP telephony software on LAN workstation computers can also be operated with the OpenCom 130 / 150.

Note: The OpenCom 131 cannot be operated with a Media Gateway card. This is why SIP telephony is not supported.

1. Install a Media Gateway card (see *MGW Interface Card* starting on page 114).

2. Call up the Configurator and go to the page **Telephony**: **Ports**: **Slots**. Click on the corresponding slot. Under **Configured,** select **MGC VoIP.** Optional: enter an IP address from the OpenCom 100 IP network which is not being used, such as 192.168.99.253 under **IP Address Configured**. Click on **Apply**.

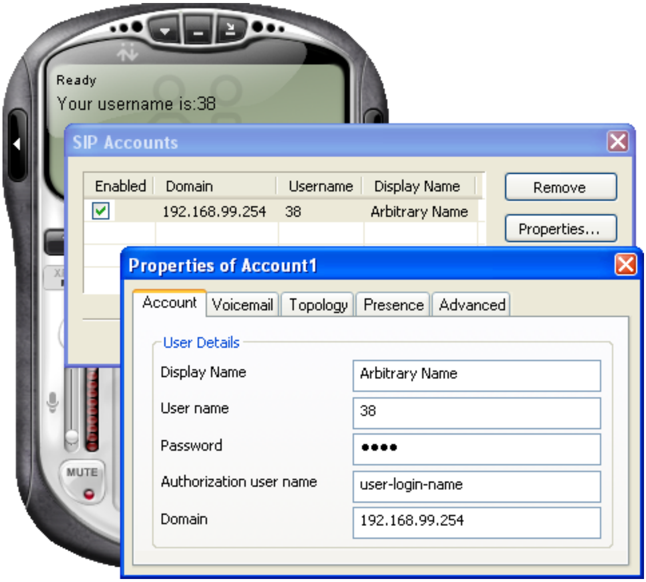
3. Call up the **Telephony**: **Devices**: **VoIP Phones** page in the Configurator. Click on **New**. Select the "SIP" option under **Type** and enter an internal **Phone No.** Click on **Apply**.

4. Call up the **User Manager**: **User** page. Assign the new internal call number to a user.

Tip: Internal SIP telephones can also be operated by users without passwords. If you do not assign the call number of the SIP telephone to a user, you can only configure a "Guest" user account on the SIP telephone.

5. An internal SIP telephone can be operated with a dynamically assigned IP address. If the SIP telephone has its own Web interface, for example, a static IP address can be practical. Click on **New** on the **Network**: **DHCP** page. Enter the **MAC address** of the SIP telephone and an available **IP address** and click on **Apply**.

6. Configure the SIP telephone or the SIP telephony software. Please also refer to the configuration help on the **Telephony: Devices: VoIP Phones** page. For the desired call number, click on **(Help)** and select a suitable help page under **Type**.



A configuration dialogue of SIP telephony software

7. You can only conduct a certain number of telephone calls simultaneously with internal SIP telephones. The number licenced can be viewed on the **System**: **Licences** page. The number of SIP telephones currently licenced can be determined on the **System Info**: **Telephony**: **SIP phones** page. If you click on **Reset licences**, the available licences will be reassigned with the next incoming or outgoing calls.

9.2 Fundamentals

VoIP makes the transmission of voice and telephony signalling via IP ("Internet Protocol") possible. After a connection is established, the terminal collects voice data (PCM data), which is then sent to the receiver using an IP packet. PCM data can also be compressed to save bandwidth.

9.2.1 Propagation Delay and Bandwidth

IP-based data networks are generally not able to guarantee a specific minimum bandwidth and defined propagation delay. A synchronised 64 kbit/s ISDN line guarantees a fixed data rate as long as the connection exists. In an IP-based data network, the data rate and propagation delay can vary. Short-term bottlenecks or retransmission due to errors may be the cause. A data flow interruption of a few seconds is barely noticeable when fetching a Web page, but it can be seriously interfere with a telephone call.

A modern Intranet normally offers enough performance reserves and reliability to make good-quality VoIP telephony possible. Specific components can also be optimised; for example by using a modern switch which evaluates the TOS byte of IP packets, by replacing unreliable connections, or by using a separated VLAN for VoIP.

9.2.2 Latency and Packet Length

For technical reasons, there is always a delay ("latency") between the recording of voice data via the microphone and playback via the receiver. Voice data is recorded for a short period so that it can be sent in an IP packet. The IP packet also has a signal-propagation delay before the receiver can begin playback. For these reasons, the extra time required for voice-data encoding and decoding may be neglected.

An IP packet consists of protocol data and user data. Sending shorter voice-data packets causes the ratio between the user data and the protocol data to become unfavourable and increases the bandwidth required. Sending longer voice-data packets increases latency.

The length of the voice-data packets must therefore be adjusted to the requirements of the transmission medium. Shorter voice-data packets can be sent if a direct ethernet connection exists. If an 64 kbit/s ISDN line is to be used for transmission, then longer voice-data packets should be used.

Longer voice data packages are generally used for SIP telephony over the Internet.

The following table provides an overview of the required bandwidth for a telephone connection with various parameter settings. The values apply to half-duplex ethernet; for full-duplex the values can be halved.

Required bandwidth (kbit/s) with respect to Packet Length and Codec

Packet Length (ms)	G.711 (not compressed)	G.729A approx. 8 kbit /s	G.723.1 6.3 kbit/s	G.723.1 5.3 kbit/s
20	180.8	68.8		
30		51.2	48.0	45.9
40		42.4		
50		37.12		
60		33.6	30.4	28.3
70		31.09		
80		29.2		

You will require a licence with which you can activate the G.729 codec for operation (in the **System**: **Licences** menu).

Note: To ensure SIP compatibility, the older system telephones OpenPhone 63 IP and OpenPhone 65 IP do not support the G.723 codec any more.

9.2.3 Voice Quality

The achievable voice quality depends on various factors. It is possible to optimise voice-data transmission on an existing network using the available configuration settings. Measuring the network quality may also help. The following comparison provides a guide to voice quality with specific quality levels:

Quality Levels for Voice Transmission with VoIP

Level	Voice Comprehensibility	Comparable to
1	Very Good	ISDN
2	Good	DECT

Quality Levels for Voice Transmission with VoIP

Level	Voice Comprehensibility	Comparable to
3	Satisfactory	GSM
4	Limited	Defective GSM
> 4	Unacceptable	No Connection

When a call is set up, the terminals involved negotiate the voice-data compression ("codec") that will be used. This is the first factor that determines the achievable quality level:

- **G.711** A-Law or µ-Law (Level 1, uncompressed): The audio data of a PCM channel (64 kbit/s) is adopted one-to-one. Every VoIP terminal must support this codec. This codec can not be used with an ISDN data connection.
- **G.729A** (Level 2): Reduction to approximately 8 kbit/s.
- **G.723.1 6.3** (Level 3): Reduction to 6.3 kbit/s.
- **G.723.1 5.3** (Level 3): Reduction to 5.3 kbit/s.

Unfavourable packet length selection may reduce voice quality. The duration of the recording and not the data packet's byte count is relevant in making this selection:

- Duration <= 30 ms: optimal transmission
- Duration 40 - 60 ms: one quality-level depreciation
- Duration > 60 ms: two quality-levels depreciation

The achievable voice quality also depends on the packet propagation delay and the packet loss between the terminals involved. These parameters can be determined using the "ping" programme.

Note: Measurements made with "ping" are round-trip propagation delays. Divide the maximum value displayed by two.

Packet Propagation Delay and Packet Loss

Value	Quality Level	Value	Quality Level
Propagation delay < 50 ms	Optimal	Loss < 1 %	Optimal
Propagation delay 50-100 ms	0.5 level depreciation	Loss 1-2 %	0.5 level depreciation
Propagation delay 100-150 ms	1 level depreciation	Loss 2-3 %	1 level depreciation
Propagation delay 150-200 ms	2 level depreciation	Loss 3-4 %	2 level depreciation
Propagation delay 200-300 ms	3 level depreciation	Loss 4-6 %	3 level depreciation
Propagation delay > 300 ms	4 level depreciation	Loss > 6 %	4 level depreciation

9.2.4 Optimisation

If you detect a large fluctuation in the propagation delay during measurement, this may also cause the voice quality to deteriorate. This may indicate a defective or overloaded line caused by bit-error or collision correction resulting from retransmission by the transmission procedure.

An existing star-topology ethernet-network may uses a Hub as the central distributor of ethernet packets. A Hub repeats all ethernet packets received on all connected lines. This can cause substantial collisions and result in a high fluctuation in the propagation delay.

If this is the case, use a modern switch component. Selective forwarding of ethernet packets (“Layer 2 switching”) avoids collisions. Modern switch components also evaluate the TOS byte of IP packets, thereby providing the optimal prerequisites for VoIP telephony.

Note: The OpenCom 100 uses a TOS byte (“Type of Service”) value of 0xB8 for IP packets with VoIP data. This requests “Minimise Delay” and “Maximise Throughput” for IP packets marked with this value.

9.2.5 Call Set-up

Various IP-based protocols are used for system telephony via the Internet protocol ("IP") (see also *Start Procedure* starting on page 129). Multiple TCP connections are made between an IP telephone and OpenCom 100 for the telephone's start procedure, registration and signalling. All voice data are directly exchanged between IP telephones using the RTP ("Realtime Transport Protocol") protocol.

Channels on a *Media Gateway (MGW)* are allocated for making a telephone connection with an ordinary terminal or for dial tones. The MGW converts IP voice data into PCM data streams used with conventional telephony and vice versa. For this, IP voice data are exchanged between the IP telephone and the gateway.

Tip: Switching between voice data channels may cause a slight delay in some circumstances. For example: when accepting a call on an IP telephone, headset users should wait about one second before answering.

9.2.6 Useful Services

The type of data compression used for VoIP prevents these types of connections from using certain services. Take these notes into account especially if you want to use connections made via Q.SIG-IP or SIP:

- ISDN data services can not be used
- Faxes can only be sent using the uncompressed G.711 codec
- DTMF dial tones are only received by the other party if the uncompressed G.711 codec is used. Alternatively, DTMF dial tones can be transferred using the Internet standard RFC 2833/4733. For this, the "DTMF (RFC4733)" codec needs to be activated for the codec configuration on the **Telephony**: **Extended**: **VoIP profile** page.
- Analogue modems can not be used

Tip: Configure the actual usage for the a/b ports, e.g. set them to **Fax** or **Data (analogue)**. Connections from and to these a/b ports will then be made using uncompressed or ISDN connections where possible.

9.3 Media Gateway (MGW)

The Media Gateway transforms VoIP voice data into PCM audio data. This function converts voice data between VoIP telephones and all other terminal types. Without the Media Gateway, VoIP telephones can only exchange call data directly with other VoIP telephones. Media Gateway functionality is also required for producing dial tones and making external phone calls with a VoIP telephone. A Media gateway card makes 8 channels available. One Media gateway channel should be available for a maximum of 3 VoIP terminals.

The Media Gateway also takes over the routing function for external SIP connections, making 8 external SIP connections possible.

9.3.1 Software MGW

The system software of the OpenCom 100 provides one limited Media Gateway function. Depending on system process utilisation and available system memory, up to four MGW channels are available. The software MGW function will be automatically activated when no Media Gateway card is present.

Using the OpenCom 131 no Media Gateway cards can be used. Here only the software MGW function can be used for VoIP.

The Media Gateway function implemented in system software has the following limitations:

- Up to four channels can be used simultaneously.
- The MGW channels are not compressed, so only the G.711 codec is available.
- There is no echo suppression.
- Voice quality may be reduced during high system utilisation.

For optimal call quality and high availability, you should consider using a MGW interface card (see also *MGW Interface Card* starting on page 114).

9.3.2 MGW Interface Card

A Media Gateway card (M100-IP) is available for the OpenCom 130 and the OpenCom 150. This interface card implements eight simultaneously usable Media Gateway channels.

Technical Data

- The MGW interface card is connected via the slot to the internal ethernet Switch. There are no external ports via pressure terminals.
- With an OpenCom 130 the MGW interface card can be operated in slot 2 or slot 3.
- With an OpenCom 150 up to two MGW interface cards can be operated in slots 3 and 4.
- The MGW interface card supports: all the codecs, as well as the silence detection, echo suppression and DTMF tone detection used by VoIP telephones.
- The MGW interface card can not be operated in a slave system.
- The MGW interface card contains the required software in its own Flash memory. This software is updated using a separate file in the **Configurator** via the **System: Components**.

Operational Information
The MGW interface card must be correctly inserted and configured (see also *Installing Interface Cards* starting on page 30).

Each MGW interface card requires its own IP address. This can either be assigned statically or obtained via DHCP.

1. In the **Configurator**, open the **Telephony: Ports: Slots** page.
2. Click on the slot number in the table column containing the desired interface card.
3. Enter the desired static IP address in the **IP address configured** field. Enter “0.0.0.0” to obtain an IP address via DHCP.

4. Confirm with **Apply**.

The configuration page displays the MGW interface card's MAC address. You will need this for the static assignment of an IP address via a DHCP server.

Note: During the commissioning of a MGW interface card, the software MGW function is automatically switched off.

Updating the interface card's software
Whenever a newer version of the M100-IP Gatewaymodul interface card's software becomes available, you can use the **System**: **Components** to load it (**MGC VoIP** option, "mgw_xxxx.tar.gz" file). The file size of a software update is approx. 2 megabyte and it takes approx. 5 minutes to load it via a LAN.

Version display
The current version of the of the M100-IP Gatewaymodul interface card's software is displayed on the **Telephony**: **Ports**: **Slots** status page.

9.4 SIP Telephony

The SIP Internet (Session Initiation Protocol) protocol provides you with a low-cost, standardised option for telephoning via IP-based networks. The OpenCom 130 / 150 enables you to use external SIP telephone connections ("SIP trunk lines"). Furthermore, internal SIP subscribers, SIP telephones or SIP telephony software are also supported (see also *Quick Start*: *External SIP Line* starting on page 104 and *Internal SIP Telephony* starting on page 106).

9.4.1 External SIP Connections

The **Telephony**: **Trunks**: **Route** menu gives you the option to configure a bundle overflow, which automatically occupies a second line in case of a breakdown or over-occupancy of the SIP connection. You can also set up your system to route certain types of calls, such as international calls, to an SIP connection.

Note: You will need a Media Gateway card for SIP telephony.

You will also need a fast Internet connection such as DSL for SIP telephony.

You will also usually need the services of a SIP provider. A SIP provider operates a special server (the SIP Registrar) to handle connections. The SIP provider also operates a gateway to the ordinary telephone network which users pay to use and which enables the SIP provider to provide calls to the telephone network. A SIP connection can also accept incoming calls from the telephone network.

The same voice transmission techniques as those explained in *Fundamentals* starting on page 107 are used for SIP telephony. SIP telephony has the following distinctive features:

- Subscribers are identified through an e-mail-like “SIP ID” such as 12345@domain.net or name@sip-provider.com.
- SIP transmits dialling numbers always in a single data package (block dialling). Dialling can therefore be concluded with the hash key # on the system terminal, or the end of the number will be indicated by a time-out. The value for this time-out can be defined for each SIP provider separately.
- You must log on (“Login”) to the SIP registrar before you can use SIP telephony. Use the OpenCom 100 to manage important information for the registration (user name and password) of one or more SIP accounts. It is possible to make several calls simultaneously using a single SIP account.
- A SIP connection causes constant Internet data traffic, so do not use SIP with Internet access which is paid for according to the time used.
- RTP call data is also exchanged directly between terminals for SIP telephony, so different codecs can be used for sending and for receiving. It is also possible to change codecs dynamically during a call. You should use every codec available in the VoIP profile at least once, because this will enable you to establish connections with as many SIP subscribers as possible.
- Fairly large packet lengths are quite normal on the Internet. They compensate for the longer packet propagation delay.
- A bidirectional RTP data stream with a dynamically-assigned UDP port number is used to set up calls between subscribers. For this reason, incoming RTP calls often fail to get past the Firewall or NAT configuration of the Internet gateway product used. If you do not use the OpenCom 100 as the Internet gateway, the product should be compatible with SIP telephony. These products provide a “Full Cone NAT” setting for this application.

- To enable the use of multiple devices on a single Internet connection, the IP addresses used in a LAN (often 192.168.x.x) are translated to a valid IP address using address translation (NAT - Network Address Translation), but no status information is available for NAT on an incoming RTP connection. To avoid this problem, the IP address of a workstation computer or telephone visible on the Internet is determined using a STUN server (STUN: Simple Traversal of UDP over NAT). You can ask your SIP provider for the STUN server's IP address and port number. If you don't need a STUN server, leave the **SIP Provider** field empty.

- For direct SIP telephony using OpenCom 100, only SIP IDs consisting of numbers for identifying subscribers registered with the SIP provider specified can be addressed

- You can integrate an external SIP connection in the **Telephony**: **Trunks**: **Route** menu into the route configuration. You can use a network provider rule to specify the routing of numbers within a specific range to use SIP telephony as a preference (see also *PBX Networking*, under *Configuration* starting on page 159).

You can configure SIP connections in the **Configurator** on the pages **Telephony**: **Trunks**: **SIP provider** and **Telephony**: **Trunks**: **SIP trunks**. Enter the technical attributes of a specific SIP provider, such as the IP addresses for the registrar and the STUN server under **SIP provider**. Under **SIP trunks** enter the information for an existing SIP account, such as the user name, password, assigned call number and the maximum number of simultaneous calls possible.

9.4.2 Internal SIP Subscribers

The OpenCom 130 / 150 becomes available as the SIP server for internal SIP subscriber telephony switching services. SIP telephones connected via LAN or SIP programmes installed on workstation computers can thus establish connections to all other devices or trunks connected to the OpenCom 100. For operation as a SIP server a *MGW Interface Card* is required.

Licence Assignment
The number of possible SIP subscribers is determined by the number of licences purchased. In order to provide you with the greatest possible flexibility regarding usage of available licences, licence assignment is dynamic via the "Floating licence". Using a user/password combination ("SIP log on") you can have several SIP subscribers under the same call number. Only every new SIP log-on occupies a new licence. The technical log-on process of a SIP subscriber with a valid user

name and correct password is always successful. Only when a connection is established is there an attempt made to occupy a licence under the SIP log-on. If all licences are occupied currently, the SIP subscriber can only make emergency calls.

Note: If the technical log-on is not successful due to an incorrect user name or incorrect password, the SIP subscriber cannot establish any connections – no emergency calls either.

When a SIP subscriber logs off, when terminating the programme, for example, the associated licence will become available immediately. A licence is also made available when a SIP subscriber's regular status query is not conducted. The internal clock for automatic log-out is determined by the **Profile** assigned under **Telephony**: **Devices**: **VoIP Phones**. The (**Keepalive**) internal clock setting is located on the **Telephony**: **Extended**: **VoIP profile** page.

Detailed information on the current licence assignment and on logged-on SIP subscribers is located on the **System Info**: **Telephony**: **SIP phones** page. This page is where you can reset licence assignment at any time by clicking on **Reset licences**.

Technical Notes

The names of settings for the various SIP telephones or SIP programmes are not uniform unfortunately. Please refer to the **(Help)** on the **Telephony**: **Devices**: **VoIP Phones** page and the following notes when configuring SIP subscribers:

- The "REGISTER" SIP message must be sent to the IP address of the OpenCom 100 using the 5060 destination port. For SIP subscribers, this setting is frequently located under "SIP Server" or "SIP Settings" with the terms "Domain", "Server IP" and "Server Port".

- The "REGISTER" SIP message must contain a valid user name and the appropriate password (the **User name** and **Password** fields in the **Configurator** under **User Manager**: **User**). For SIP subscribers, this setting is frequently located under "SIP User Settings" or "SIP Account" with the terms "Authorization User" and "Password".

- The "REGISTER" SIP message also contains a SIP-URI in the spelling for e-mail addresses, for example `"Displayname" <sip:123@192.168.99.254>`. The text portion of the SIP-URI ("Display Name") is not evaluated at log-on from the OpenCom 100. The series of characters before "@" is the "User Name" or "SIP Username". The internal call number of the user must always be used here (the **Ph.No.** field in the **Configurator** under **User Manager**: **User**). The series of characters after “@” is the “Domain Name” or the “SIP Domain”. The IP address of the OpenCom 100 must always be used here.

- A STUN server (Simple Traversal of UDP over NAT) or a SIP proxy is not required because internal SIP subscribers on the LAN are directly connected to the OpenCom 100. Switch these functions off if possible.

- With a SIP terminal, you can enter an international phone number with a leading plus. When you enter a call number in the E.123 format, the plus char is substituted by the “00” number sequence and the immediate line seizure via the standard route is activated for the call. If you prefer to dial in this number format, you should activate the international call number conversion (see *E.164 conversion* starting on page 163).

Features
SIP subscribers can establish connections to all other terminals and trunks. The SIP protocol generally works with block dialling. This is why the selected call number is only activated after an internal clock has expired or activated immediately via the hash key (“#”) when dialling. This is why code number procedures without the hash key and code number procedures with a concluding hash key can be used. An overview of code number procedures that can be used is located in the Configurator on the **System Info**: **Codes** page**.** Activate “SIP phones”. Please also note the corresponding information in the “OpenCom 100, Operation on Standard Terminals” user guide.

Alongside code number procedures, SIP subscribers can also use a series of functional features realized via the SIP protocol. The OpenCom 100 is the ending for all SIP connections as opposed to what is usually the case on the Internet. This enables SIP subscribers to use OpenCom 100 features. Direct data exchange is thus not possible between two SIP subscribers. The following table shows the possible features.

Features	Notes
Incoming and outgoing calls with call number display (CLIP)	A SIP telephone requires a call number display for CLIP.
Parallel connection of multiple SIP subscribers	SIP subscribers must be logged on under the same user identification.
Enquiry, toggling, call waiting, three-way conference, reject	Operation or feature must be available on the SIP telephone or in the SIP software.

Features	Notes
Call transfer	before and during a call; operation must be available.
Blind Transfer	SIP only: forwarding an incoming call without accepting the call; feature must be supported by SIP telephone or by the SIP software.
Keypad as “INFO” message	DTMF tones cannot be securely transferred “in band” via compression codecs. Digital “out band” transferral as “INFO” SIP message or via RFC 2833/ 4733 is supported. This feature must be available and activated on a SIP telephone or in the SIP software.

9.4.3 Aastra 673xi/675xi SIP Telephones

You can operate the SIP telephones of the Aastra 673xi and Aastra 675xi product families at the OpenCom 100 communications system. The firmware of the communications system already includes the matching firmware files for the following SIP telephones:

Aastra 673xi and Aastra 675xi SIP telephones

Model	Short description
Aastra 6730i	Basic level version of the Aastra 673xi products, 5 volts input via wall power supply, display with 3 lines, 8 programmable keys, 2 line keys
Aastra 6731i	Similar to Aastra 6730i, but with 802.3af (PoE) power supply, two Ethernet ports for PC and LAN, separate 48 volts wall power supply available
Aastra 6751i	Basic level version of the Aastra 675xi products, display with 3 lines, 802.3af (PoE) power supply, two Ethernet ports for PC and LAN
Aastra 6753i	Similar to Aastra 6751i, but with headset connector, 6 programmable keys, 3 line keys, up to 3 key extensions with function keys

Aastra 673xi and Aastra 675xi SIP telephones

Model	Short description
Aastra 6755i	Model with illuminated comfort display, 802.3af (PoE) power supply, two Ethernet ports for PC and LAN, 6 programmable keys and 6 softkeys, 4 line keys, up to 3 key extensions with function keys or 3 key extensions with softkeys
Aastra 6757i	Similar to Aastra 6755i, but with illuminated premium display which provides space for 12 softkeys

The Aastra 673xi/675xi SIP telephones provide, besides the VoIP telephony, additional system telephony features which you can configure conveniently and safely with the help of the Web configurator. The Aastra 673xi/675xi SIP telephones support G.711 codec (a-Law and µ-Law), the G.729 codec and the transmission of DTMF signals according to RFC 2833. An *MGW Interface Card* is required to operate Aastra 673xi/675xi SIP telephones with the OpenCom 100 communications system. As an additional requirement, a memory card with 256 Mb or more must be installed. This will provide the necessary memory space to store the firmware files for the varying Aastra 673xi/675xi SIP telephones.

The general commissioning takes place in the **Configurator** of the OpenCom 100 communications system with the following steps:

1. Under **System**: **Licences**, activate the SIP system devices (see also *Internal SIP Subscribers* starting on page 117).

2. Under **System**: **Components** load the firmware addons to the memory card.

3. Add a new entry under **Telephony**: **Devices**: **VoIP Phones**. Enter a call number and select the telephone type as Aastra 673xi/675xi. Optionally enter the MAC address and the IP address of the Aastra 673xi/675xi (see *Aastra 673xi/675xi DHCP* starting on page 124).

4. Under **Telephony**: **Devices**: **System phones** configure the common settings, the programmable keys and the softkeys for the Aastra 673xi/675xi (see *Aastra 673xi/675xi Setup* starting on page 122).

5. Connect the Aastra 673xi/675xi to your network and plug in the power supply. Details about this can be found in the installation manual which is provided with the Aastra 673xi/675xi.

If the Aastra 673xi/675xi has the factory default configuration, it will request an IP address configuration from the DHCP server of the OpenCom 100 communications system. The IP address and the location for loading files from the TFTP server of the OpenCom 100 is received as a part of the DHCP answer.

Now the Aastra 673xi/675xi reads in the configuration files from the TFTP server. This includes a generic configuration file ("aastra.cfg") and a device specific configuration file for the given MAC address. During this process the OpenCom 100 communications system transfers all settings and by this it configures the system telephony features for the Aastra 673xi/675xi. If necessary, The firmware stored in the Aastra 673xi/675xi as well as the diverse language modules are updated also.

Details about the boot procedure, how to program extensions ("XML keys") and about the manual DHCP-/TFTP configuration can be found in the english manual "IP Phone Admin Guide", which you can download from the Aastra web site.

9.4.3.1 Aastra 673xi/675xi Setup

You can change the configuration and function key settings for each Aastra 673xi/675xi individually:

1. Call up the **Telephony**: **Devices**: **System phones** menu page in the **Configurator**.

2. In the **Devices** list, select the desired Aastra 673xi/675xi.

   The menu page shows the current configuration, the function key assignment and a device graphic.

3. Click on the **Change** button to determine general settings, e.g. the display language. Confirm the setting on the following page with **Apply**.

4. Depending on the terminal type, different lists with function keys will be shown:

   – **Programkeys**: You can label this function keys on the device using a paper strip.

   – **Topsoftkeys** (Aastra 6757i only): You can label this function keys on the device display. You can switch between 2 layers of function keys using one of the function keys on the device.

– **Softkeys** (Aastra 6755i /6757i only): You can label this function keys on the device display. You can switch between 4 layers of function keys using one of the function keys on the device.

Click on the function key heading to call up the configuration dialogue for this function key. Select a function in the **Type** setting and optionally enter a label in the **Labelling** setting. Confirm the setting with **Apply**.

5. To transfer the settings to the Aastra 673xi/675xi, click the **Apply** button on the **Telephony**: **Devices**: **System phone** page.

   The Aastra 673xi/675xi restarts and thereby takes over the new configuration.

You can configure the following function keys for an Aastra 673xi/675xi.

Function keys for Aastra 6730i / 6731i and Aastra 6753i / 6755i / 6757i

Key type	Parameter	Function
empty	–	none
Speed dial	Call number	Direct dialling of a call number or code procedure
BLF	Call number	Busy Lamp Field; LED indicator shows when caller is busy
Ph.Book	–	Shows phone book managed by the OpenCom 100 communications system
Answered Calls	–	Shows the list of missed calls which is managed on the OpenCom 100 communications system; LED indicates available calls
Answered Calls	–	Shows list of answered calls managed by the OpenCom 100 communications system
Voice message	–	Shows the list of recorded voice messages; LED indicates available messages
Voice box	–	Calls your voice box
Presence	–	Displays and changes your presence status (“Messenger”)
Call forwarding	–	Shows the “call forwarding” menu; LED indicator shows when call forwarding is active

Function keys for Aastra 6730i / 6731i and Aastra 6753i / 6755i / 6757i

Key type	Parameter	Function
do not disturb	Calling internally / externally	Switches call protection on or off; LED indicator indicates active function
Pick-up	–	Picks up a call from another device in your pick-up group
Take	–	Transfers an active call from another device operated on you internal call number
Phone Lock	–	Locks or unlocks the telephone
Logout	–	Logs out the telephone and shows the Login Display
XML	URL	Calls the defined URL from a Web server. Programmable function control via XML data retrieval (please refer to the “IP Phone Admin Guide” administrator guide)
transfer	–	On the Aastra 6753i only: starts a transfer call
conference	–	On the Aastra 6753i only: starts a three-party conference

Details on using the Aastra 673xi/675xi SIP telephones as well as an overview about the possible function key assignments can be found in the “Telephony with SIP Phones on the Aastra 800/OpenCom 100 communications system” user guide.

9.4.3.2 Aastra 673xi/675xi DHCP

The address and device configuration is transferred from the OpenCom 100 communications system to the Aastra 673xi/675xi SIP phone using the DHCP protocol. You can configure the necessary address settings while adding a device entry under **Telephony**: **Devices**: **VoIP Phones** differently:

- You can enter the MAC address of the Aastra 673xi/675xi manually. Depending on the operation mode of the DHCP server, you can also enter the IP address manually. For this, change the **Status** setting on the **Network**: **DHCP** page to “Static address assignment”. Optionally, you can use the dynamic IP address assignment from the DHCP server. For this, change the **Status** setting on the **Network**: **DHCP** page to “Dynamic address assignment” and also change the

Devices selection to the "All" setting or to the "with configured MAC only" setting.

- By using the "Easy Configuration / Hot-Desking" feature you can add a device entry without entering a MAC address and without entering an IP address. An unregistered Aastra 673xi/675xi will show a login page after startup. A user can enter the desired call number and the matching user PIN which will establish the association of the user account to the device entry. For this, change the **Status** setting on the **Network**: **DHCP** page to "Dynamic address assignment" and also change the **Devices** selection to the "All" setting or to the "with configured MAC only and all SIP system devices" setting.

If you want to use both features concurrently, change the **Status** setting on the **Network**: **DHCP** page to "Dynamic and static address assignment".

If you operate another DHCP server in your network, the Aastra 673xi/675xi can distinguish the different DHCP answers and will normally take its configuration data from the OpenCom 100 communications system nevertheless. In a conflict situation, you may need to prevent the configuration of the Aastra 673xi/675xi SIP phone by a foreign DHCP server. For example, add an exception rule for all MAC addresses starting with 00:08:5D.

The TFTP server of the OpenCom 100 communications system will serve up to 4 concurrent data transfers. To avoid overloading the TFTP server (e.g. after a power failure), additional devices may get an IP address from the DHCP server with a delay. The allocation of IP addresses will restart if the TFTP server indicates the end of the overload situation.

9.4.3.3 Aastra 673xi/675xi Hot Desking

The Aastra 673xi/675xi SIP phones support the "Easy Configuration / Hot desking" feature. This feature allows a user to operate his personal configuration on any desired Aastra 673xi/675xi of equal type.

To prepare a Aastra 673xi/675xi for Hot Desking, you can configure a function key with the "Logout" function. If this function key is pressed, the currently assigned device entry will be marked as "Logged Out" and the Aastra 673xi/675xi restarts. You can indicate logged out device entries in the Configurator on the **Telephony**: **Devices**: **VoIP Phone** page by their shortened MAC address.

After restart, the logged out Aastra 673xi/675xi will show a login page. A user can enter his call number and his user PIN. If a device entry with this call number exists and has the same device type, the login procedure continues. If the device entry is currently active on another device, the associated remote Aastra 673xi/675xi will be logged out automatically and will show a login page on its part. The now logged in Aastra 673xi/675xi starts and will get its configuration from the newly assigned device entry.

Note: When adding the device entry, it is also possible to activate the **Logged out** option.

9.5 VoIP System Telephones

The following telephones and software packages are available for VoIP system telephony:

- Aastra 6773ip (OpenPhone 73 IP): This is a VoIP-enabled edition of the Aastra 6773 (OpenPhone 73) system telephone. This system telephone can be extended with up to three key extensions Aastra M671.
- Aastra 6775ip (OpenPhone 75 IP): This is a VoIP-enabled edition of the Aastra 6775 (OpenPhone 75) system telephone. This system telephone can be extended with up to three key extensions Aastra M671 or Aastra M676.
- Aastra 277xip (OpenPhone 7x IPC): This VoIP software offers the functionality of a system telephone using Windows XP/Vista executable software (see *Aastra 277xip (OpenPhone 7x IPC)* starting on page 135). This software also provides local answering machine functionality and can be integrated into CTI applications.
- The older VoIP system telephones OpenPhone 63 IP and OpenPhone 65 IP can still be used. With the OpenPhone 63 IP and OpenPhone 65 IP telephones no postdialling via DTMF is supported for ISDN or SIP lines.

9.5.1 Device Properties

The VoIP-enabled versions of the system telephones Aastra 6773ip (OpenPhone 73 IP) and Aastra 6775ip (OpenPhone 75 IP) offer the same features as the corresponding system telephones. Using VoIP system telephones is therefore not much different from using standard system telephones. The following differences exist:

- Two RJ45 connector ports are available for ethernet connection. The ports are connected to one another via the telephone's internal switch. The switch supports 10 Mbit/s or 100 Mbit/s full-duplex with priority given to VoIP data transmission.

  LAN Port: Allows the telephone to connected to the LAN. Use a non cross-over RJ45 patch cable to connect to a Hub or Switch.

  PC Port: Allows the telephone to be connected to a workstation computer. Use a non cross-over RJ45 patch cable to connect to the PC's network port.

- The VoIP system telephone's power supply is provided by an extra plug-in power supply. It is also possible to provide a power feed via PoE ("Power over Ethernet"). PoE requires special devices for power feeds, as well as a completely wired RJ45 connection line.

- You can also connect a standard headset via RJ45 sockets (DHSG standard) to VoIP system telephones.

- VoIP system telephone's audio signals are generated by the telephone itself. DTMF dial tones and Music on Hold are produced by the Media Gateway function.

- A VoIP system telephone can also be operated without a permanent connection to the communications system, for example via an on-demand RAS connection.

- Signalling data for call control, call data during three-way conferences, connections to conventional terminals and external connections is exchanged between the VoIP system telephone and the communications system. During a call between two VoIP system telephones, call data is exchanged directly between the two VoIP system telephones.

- During the device's start procedure, the IP address is configured and the device software is requested via the DHCP and TFTP network protocols.

9.5.2 VoIP System Telephone Configuration

The VoIP system telephones Aastra 6773ip (OpenPhone 73 IP) and Aastra 6775ip (OpenPhone 75 IP) obtain the required IP address configuration and operating software via the DHCP, BOOTP and TFTP IP protocols. After the power supply is assured, the device's internal boot loader is started which controls the further start procedure.

Standard operating procedure is to contact the OpenCom 100's DHCP server so that the start procedure can be concluded without problems. To register a new VoIP system telephone, proceed as follows:

1. Temporarily remove the VoIP system telephone's ethernet connection. Switch on the VoIP system telephone's power supply. Note the MAC address shown in the display, for instance "MAC: 00:30:42:00:00:00". Switch off the power supply.

2. In the **Configurator**, open the **Telephony: Devices: VoIP Phones** page. Click on the **New** button.

3. Select the VoIP system telephone's **Type** and enter the previously noted MAC address. Assign a **Name** and **Phone No.** Confirm with **Apply**.

4. Connect the ethernet connection with the VoIP system telephone's RJ45 connector. Switch on the power supply. Verify the correct start procedure on the display.

9.5.3 LAN DHCP Server

If the LAN already uses a DHCP server to configure workstation computers, there are various options for correctly responding to VoIP system telephones' DHCP, BOOTP and TFTP requests. A comparatively simple approach is described here.

1. Configure the LAN's DHCP server to ignore DHCP requests from the VoIP system telephones. With a Linux DHCP server programme, you must, for example, include the following lines in the system file "/etc/dhcpd.conf":

```
group {
 deny booting;
 host 192.168.11.12 {
  hardware ethernet    00:30:42:00:11:22;
 }
}
```

Every DHCP service programme has similar options. You may need to reserve a free IP address for each VoIP system telephone. You will find more details in your DHCP service programme's online help or handbook. The MAC address of all VoIP system telephones always begins with 00:30:42.

2. Configure a fixed IP address for the OpenCom 100. To do this, call up the **Configurator** and open the **Network: LAN** page. Click on the **Change** button.

3. Enter the current IP address configuration in **IP address** and **Network mask**. Confirm with **Apply**.

4. Configure the OpenCom 100's DHCP server to assign IP addresses. To do this, call up the **Configurator** and open the **Network: DHCP** page. Click on the **Change** button.

5. From **Status**, select the **address assignment** option. Confirm with **Apply**. The **DHCP** page is re-displayed.

6. Add the configured VoIP system telephones to the list of IP addresses. Click on the **New** button.

7. Enter the VoIP system telephone's **IP address** and **MAC address**. Enter the IP address reserved by the DHCP service programme. Confirm with **Apply**.

Restart the OpenCom 100 and all connected VoIP system telephones.

9.5.4 Start Procedure

It may sometimes be useful to understand a VoIP system telephone's start procedure. Examples:

- A complex DHCP address assignment prevents the operation of the OpenCom 100's DHCP server within the LAN.

- A VoIP system should be operated with a non-broadcast-capable IP connection. This may be an RAS connection, a VPN connection or another type of routed connection.

An external DHCP server can also control a VoIP system telephone's start procedure. In this case, system software matching the type of VoIP system telephone must be transferred via TFTP.

The file name is determined by the telephone type. If a Media Gateway card is installed, you can also specify the TFTP server on the card which has a higher performance.

Telephone type	**OpenCom 130/131/150**	**Media Gateway card**
Aastra 6773ip (OpenPhone 73 IP)	opi7x.cnt	/ram/ip_tel/opi7x.cnt
Aastra 6775ip (OpenPhone 75 IP)	opi7x.cnt	/ram/ip_tel/opi7x.cnt
OpenPhone 63 IP	opi63.cnt	/ram/ip_tel/opi63.cnt
OpenPhone 65 IP	opi65.cnt	/ram/ip_tel/opi65.cnt
RFP 32/34/42	ip_rfp.cnt	/ram/ip_tel/ip_rfp.cnt

After the VoIP system telephone has been connected to the mains power supply, the start procedure is as follows:

1. The boot loader starts and shows the VoIP system telephone's MAC address in the display. A DHCP request is sent simultaneously via broadcast on the 255.255.255.255 broadcast address.

2. An IP address, network mask and the default gateway for the start procedure are sent from the DHCP server. Via the "Next server" option, the DHCP server also provides the TFTP server's IP address and the operations software's file name. The DHCP server uses the MAC address to select the operations software file which matches the type of device.

3. The boot loader loads the approximately 2 MB operations software file from the specified TFTP server. The TFTP server's IP address and the file's name are shown in the display. The loaded operations software is started.

4. The operations software sends a DHCP request on the broadcast address 255.255.255.255. The VoIP system telephone now receives an IP address, network mask and default gateway for operations from the DHCP server. Using “Option 43”, which is reserved for this purpose, the DHCP server also provides the IP address of the communications system and port number 8100 for registration.

5. The VoIP system telephone creates a TCP connection to the supplied IP-address/port-number combination and sends a registration query. The OpenCom 100 checks the MAC address sent with the registration and confirms the registration request if the VoIP system telephone is listed in the menu **Telephony: Devices: VoIP Phones**. The keep-alive time, port number (8101) for telephony signalling and the value to use for the TOS byte are also communicated in the registration answer.

6. The VoIP system telephone creates a second TCP connection using the signalling port number 8101 and sends a registration analogue to the U_{pn} system telephones.

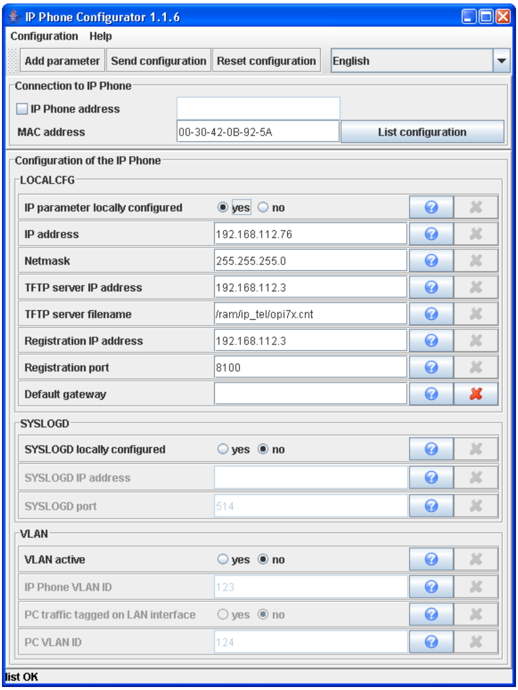
7. Extra connections are created using the IP protocol RTP (“Realtime Transport Protocol”) for call data when a call is created. For calls between two VoIP system telephones, port numbers above 8200 are used. For transmission to a Media Gateway card, a port in the range 1024 – 1087 is used.

If you wish to operate a VoIP system telephone via a routed IP connection (for example VPN or RAS) it may be necessary to configure an external DHCP server accordingly. Please note the selection of the codec and keep-alive time for RAS connections. This can be done by selecting the default profile **RAS** in the **Telephony: Devices: VoIP Phones** for the VoIP system telephone. The operations software provided via TFTP must match the type of device and communications system. You may also need to configure BOOTP, DHCP and TFTP servers for the VoIP system telephone.

9.5.5 Local Configuration

In addition to automatic configuration via BOOTP/DHCP, it is possible to manually configure an Aastra 6773ip (OpenPhone 73 IP) or an Aastra 6775ip (OpenPhone 75 IP). This can make sense, for example, when you wish to connect a VoIP system telephone at a distant location via router. This local configuration is saved permanently in the non-volatile memory of the VoIP system telephone. To change the local configuration, use an additional programme, the Java-based “IP Phone Configurator”.

Note: Java programmes can be run on all common operating systems. To execute Java-based programmes, you must install a suitable Java runtime environment on your operating system (JRE). This can be downloaded under the following web address: http://www.java.com/.



1. The “IP Phone Configurator” can be started directly from the Product CD. Start Windows Explorer. Navigate to the Product CD. Double-click the “Aastra\IpPhoneConfigurator.jar” file.

The “IP Phone Configurator” dialogue opens. Select the desired language setting (“English” or “German”)from the drop-down menu.

2. Enter the network address of the VoIP system telephone. You have two connection types to choose from under **Connection to IP Phone**:

– Deactivate the **IP Phone address** option to establish a broadcast connection via “UDP-Broadcast”. You have to select this type of connection if the VoIP system telephone has not yet been assigned an IP address. IP broadcasts cannot be transmitted via router. The VoIP system telephone thus has to be directly connected to your PC via a hub or via a switch.

– Activate the **IP Phone address** option to establish a point-to-point connection via “UDP-Unicast”. Enter the IP address of the VoIP system telephone into the entry field. You can select this type of connection if the VoIP system telephone has already been assigned an IP address.

3. Enter the **MAC address** of the VoIP system telephone. You will find the MAC address on the underside of the device. Click on **List configuration**. The status bar at the bottom edge of the programme window displays “list OK”.

4. Change the desired settings under **Configuration of the IP Phone**. Click on **Reset configuration** to activate the standard settings for all entry fields.

5. Click on the **Send configuration** command to transfer the currently shown configuration to the VoIP system telephone. The status bar at the bottom edge of the programme window displays “send OK”.

Note: The VoIP system telephone receives the configuration and sends a response. The new configuration is only saved and activated once this has happened. This can result in the “IP Phone Configurator” not receiving the response of the VoIP system telephone.

Please note: If you are operating multiple network cards with active IP configuration in your PC, this may mean that the loading of configuration data fails. First you deactivate additional network cards or use a point-to-point connection. Sending configuration data with a broadcast connection functions even without a response from the VoIP system telephone.

You can implement the following settings:

IP parameter locally configured: Select the **yes** option to activate manual IP address configuration. Select the **no** option to activate automatic IP address configuration via BOOTP/DHCP.

IP address and **Netmask**: Enter an available IP address and the network mask to be used by the VoIP system telephone.

TFTP server IP address and **TFTP server filename**: Enter the IP address and the complete file name for the operating software of the VoIP system telephone (see table on page 130).

Registration IP address and **Registration port**: This is where you usually enter the IP address of the OpenCom 100 and the port number 8100.

Default gateway: Click on the **Add parameter** command to have this optional entry field displayed. Then enter the IP address of the router ("Default Gateway"). Click on the Delete button to remove the optional parameter.

SYSLOGD: For monitoring purposes, VoIP system telephone messages can be sent to a Syslog server. Activate the **yes** option and configure the **SYSLOGD IP address** and **SYSLOGD port** settings to activate this function.

VLAN (Expert option - do not change the "no" setting in standard cases): To improve transmission security or to enforce security guidelines, PC data transmission and VoIP data transmission can be separated using this method. Activate the **yes** option and enter the desired **IP Phone VLAN ID** for the VoIP data transmission. Enter a value ranging from 1-4094. Data will always be transmitted without a VLAN tag at the VoIP system telephone's PC access. If you activate the **yes** option for **PC traffic tagged on LAN interface**, PC data will be labelled with the **PC VLAN ID** at the LAN access. Please note that to change the VLAN settings, the VoIP system telephone has to be restarted.

9.6 Aastra 277xip (OpenPhone 7x IPC)

Besides the hardware VoIP system telephones, PC software for VoIP telephony can also be deployed. This software can be used with the operating systems Windows XP and Vista.

Software VoIP system telephone Aastra 2775ip (OpenPhone 75 IPC) with one key extension

As well as VoIP system telephony from workstation computers, the Aastra 277xip (OpenPhone 7x IPC) includes the following features:

- Usage via Mouse/PC keyboard
- “Drag & Drop” call number selection
- Integrated answering machine / recording function
- Terminal control for the sight-impaired
- Selectable user interface (“Skins”)
- Display language modification

The workstation computer requires a full-duplex-enabled sound card as well as a suitable headset for audio recording and playback.

You will not need a licence to install Aastra 277xip (OpenPhone 7x IPC) but you will require a licence to operate it with the OpenCom 100. Unit licences, enabling the simultaneous operation of a certain number of Aastra 277xip (OpenPhone 7x IPC)s, are available.

The licences are activated in the OpenCom 100's **Configurator** in the Menu **System: Licences**. The system software includes a licence for a demo version for temporary use (60 days). Please contact your local dealer or Aastra sales department if you wish to purchase a permanent licence. Licences can be combined. Each licence can be activated only once.

9.6.1 Installation

Installation is done using a setup programme. The Aastra 277xip (OpenPhone 7x IPC) can also be installed without a user interface. The programme can then be used via a CTI application (Net-TAPI or OpenCTI).

Start the installation programme Aastra 277xip (OpenPhone 7x IPC) from the product CD and follow the installation assistant's instructions.

9.6.2 Configuration

Analogue to the VoIP system telephones, the Aastra 277xip (OpenPhone 7x IPC) creates multiple IP connections to the OpenCom 100. When you start the programme for the first time, the **Options** dialogue is automatically opened. Here you must configure the following values:

1. Enter in the **VoIP IP Address** field the OpenCom 100's IP address.

2. Enter six hexadecimal-digits into the **Device ID** field. This device ID is not a MAC address, so overlapping with existent MAC addresses is possible. The device ID is configured in the **Configurator**, on the **Telephony: Devices: VoIP Phones** page.

3. Confirm with **OK**.

Notes
VoIP system telephony requires an active IP connection to a workstation computer. If a Firewall is installed for your workstation computer, you may need to explicitly allow this connection.

If you log on to the workstation computer using a different user name, you must reconfigure these values.

You can use any arbitrary sequence of digits not already in use in the LAN for the device ID. Select a random device ID to secure telephone usage. The device ID can only be read on the Web console.

The displayed menu texts and parts of the operations software are elements of the Aastra 277xip (OpenPhone 7x IPC) installation, but they can be loaded from the OpenCom 100 via TFTP where necessary.

10. DECT over IP®

In order to achieve optimal network coverage, a DECT network with several DECT base stations can be operated. A DECT network is comprised of DECT terminals connected with the next respective base station (network cell). For users of a DECT terminal, the handover from DECT base station to base station is completely transparent. Even during a conversation, users are switched from one network cell to the next without any interruption. Administration of DECT terminals is done centrally via the OpenCom 100 Configurator in the **Telephony: Devices: DECT Phones** menu.

Note: DECT over IP® is a registered trademark of Aastra Telecom Schweiz AG.

10.1 Properties

10.1.1 DECT Base Stations

DECT base stations can be connected to the OpenCom 100 via U_{pn} accesses or via network (TCP/IP). These DECT base stations are available for the type of access selected:

- RFP 22: Access via U_{pn} with lines up to 1000 metres in length; integrated antennas; 4 voice channels (8 when using 2 U_{pn} accesses)
- RFP 24: like the RFP 22; mounted outside enclosed areas (IP55); external antennas

Note: The newer DECT base stations RFP 22 and RFP 24 can be simultaneously operated with the older DECT base stations, RFP 21 and RFP 23. Fax transmissions (group 3 with ECM) and SARI (roaming with Secondary Access Rights Identification Broadcasts) can be done using the newer DECT base stations. Data transmission via DECT is not available with the newer DECT base stations.

- RFP 32: Access via shielded CAT5 Ethernet cable (STP cable, Shielded Twisted Pair cable) with up to 100 metres of cable from the last Ethernet switch, integrated antennas; 8 voice channels

- RFP 34: like the RFP 32; mounted outside enclosed areas (IP55); external antennas
- RFP 42: Access via shielded CAT5 Ethernet cable (STP cable, Shielded Twisted Pair cable); offers simultaneous function of a WLAN Access Point conforming with the IEEE 802.11b/g protocol; external antennas; 8 voice channels

Note: The DECT base stations RFP 32, RFP 34 and RFP 42 support the DECT encryption function. This feature is however, only available if all the DECT base stations support it.

Note: When started, the operating software for the DECT over IP base stations is transmitted via TFTP protocol from the OpenCom 100. The configuration for the start sequence is transmitted by the DHCP server of the OpenCom 100 to a DECT over IP base station for the start sequence.

If VoIP telephony is already being used, Ethernet cable access makes good sense. Transmission of telephony signalling and voice data via TCP/IP also offers usage of existing network infrastructure and an increase in range using suitable methods. VPN connections, for example, can be used for data links to provide service to remote or hard-to-reach locations.

10.1.2 Features

All DECT over IP base stations can be connected to a CAT5 Ethernet cable with a 10/100 Base T. Power is supplied either via Power-over-LAN (IEEE 802.3af) or via an additional power supply unit.

Please note: The WLAN function of the RFP 42 is activated only when connected to the 100 Base T.

DECT terminals offer all system telephony features. DECT telephones supporting the GAP standard can also be operated. Transparent GAP device handovers are supported. DECT encryption of calls can be deactivated for the RFP 32, RFP 34 and RFP 42 if desired.

VoIP audio communication between the DECT over IP base station and the OpenCom 100 is made via the RTP/RTCP protocol. RTP voice data are directly converted into DECT voice data by the base station. The base stations support the following VoIP codecs:

- G.711: uncompressed
- G.723: compressed
- G.729: compressed

10.2 Configuration

One of the DECT over IP base stations that is installed assumes coordination and configuration of the DECT over IP functions ("DECT over IP Manager", OMM). Select a base station that has a dependable data link to the OpenCom 100.

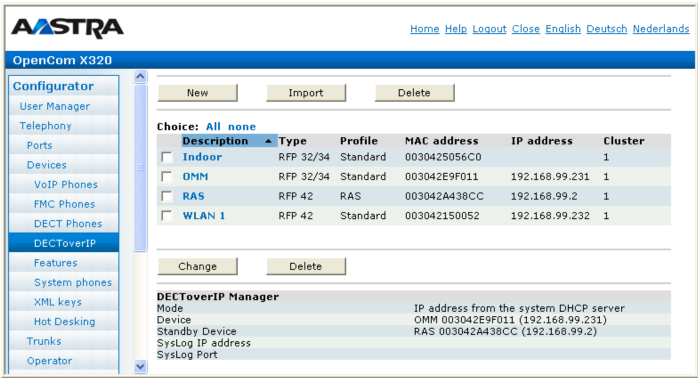
Note: You can determine a second base station as additional DECT over IP Manager ("Standby Device"). If the first DECT over IP Manager fails, the second base station takes over this critical function after some minutes and a reset of the DECT network.

Go to the **Telephony: Devices: DECT over IP** page in the **Configurator**. Click on **New**, to add a DECT over IP base station. Enter the **MAC address** of the base station that you have selected to be the DECT over IP Manager. The MAC address of the base station is located on its type label. Enter an **IP address** for this base station. Confirm with **Apply**. Now click **Change**, to determine the DECT over IP Manager. In standard cases, leave the **Mode** in the "IP address from the system DHCP server" setting. Select the desired DECT over IP base station with the **MAC address (IP address)** setting. Confirm with **Apply**.

If the OpenCom 100's DHCP server is not configured for static address assignment, you first need to configure the IP address of the DECT over IP Manager with the help of an additional programme (see *Local IP Address Configuration* starting on page 146). Also change the **Mode** setting on the **Telephony: Devices: DECT over IP** page to "IP address configured local" and enter there the configured IP address for the DECT over IP Manager. All other base stations can be operated using either a fixed IP address or an IP address assigned dynamically via DHCP. Please refer to the information given in the chapter entitled *LAN DHCP Server* starting on page 128.

Note: A base station cannot be operated as a DECT over IP Manager and a WLAN-Access Point simultaneously.You should therefore use a DECT over IP base station which does not have WLAN function as your DECT over IP Manager.

Create a separate entry for each DECT over IP base station and for the DECT over IP Manager on the **Telephony: Devices: DECT over IP** page. You use these entries to determine the VoIP data compression ("Profile").



Configurator: Telephony: Devices: DECT over IP

User administration and set-up of DECT terminals is done in the Configurator of the OpenCom 100 as well.

The DECT over IP Manager offers a separate web user interface to manage the settings of devices with WLAN functions. Therefore at least one WLAN-RFP has to be configured. If everything is configured correctly, you will see the **WLAN Config** link. Login as the user "Administrator" with the currently set administrator's password of the OpenCom 100.

10.2.1 Dual Operation

Simultaneous operation of base stations via U_{pn} access and base stations via Ethernet access is possible with the OpenCom 100. Transparent handovers, for example, are only possible when between DECT base stations using the same access technology. When switching over to a DECT base station using different access technology, a DECT terminal automatically re-establishes a connection (roaming).

Be sure to keep the PARK ID of your OpenCom 100 in mind. The PARK ID is displayed under **System Info**: **Versions** in the **Configurator**. If the PARK ID starts with 31, there may be an attempt at a handover between DECT base stations using different access technology. In this case, make sure that the signal areas of the DECT base stations using different access technology do not overlap. If the PARK ID starts with a different value (e. g. 30), prevention of unintentional handovers is effective.

10.2.2 Synchronisation

Transmissions of all DECT base stations at a single location must be synchronised in order that DECT terminals are able to receive multiple DECT base stations simultaneously. Synchronisation can be conducted via U_{pn} access. It cannot be conducted via an Ethernet/IP connection. DECT over IP base stations are thus synchronised via wireless connection.

When planning a larger sized DECT network, it is advisable to take the following points into consideration:

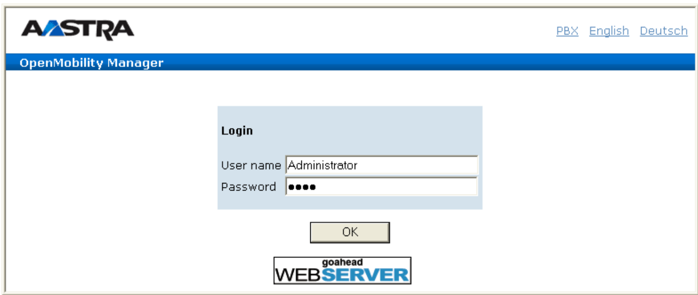
- All DECT over IP base stations at a single location must be able to receive at least one, or even better, two neighbouring base stations. Synchronisation requires less signal strength than a voice connection does.
- Synchronisation range is increased using multiple base stations. To decrease the probability of a connection breakdown, base stations should not be arranged in chain formation. The signal should be distributed with a network that is as extensive as possible and where each base station is supported by multiple synchronisation partners.
- To re-synchronise, first wait for all current connections to be terminated.

You can operate a DECT network consisting of several remote locations (“clusters”). A cluster is a number of DECT base stations that operate synchronously with each other. No handover is possible between DECT base stations from different clusters. You should configure a second cluster for DECT base stations of a second location.

10.2.3 Setting up the WLAN Function

The RFP 42 DECT over IP base station provide the additional function of a Wireless LAN Access Point (WLAN-AP). WLAN refers to data transfer by means of radio waves in accordance with the IEEE 802.11b/g standard. This standard enables a wireless connection to be made to an Ethernet network (LAN) using suitably equipped user terminals. Data transfer via radio waves is very fast. Depending on the conditions of the operating environment, it can reach speeds of up to 54Mbit/s (gross).

WLAN settings are configured centrally for all Access Points using a separate Web configurator, which can be found at the IP address of the DECT over IP Manager (OMM, OpenMobility Manager). You can reach this address by entering the IP address of the DECT over IP Manager directly into the address bar of your Web Browser. Alternatively, you can also go to the **Configurator**, to the page **Telephony: Devices: DECT over IP** and click on the **WLAN Config.** button. Log in under the **User Name** "Administrator" and enter the same password as for the OpenCom 100.



DECT over IP/OpenMobility Managers Login Page

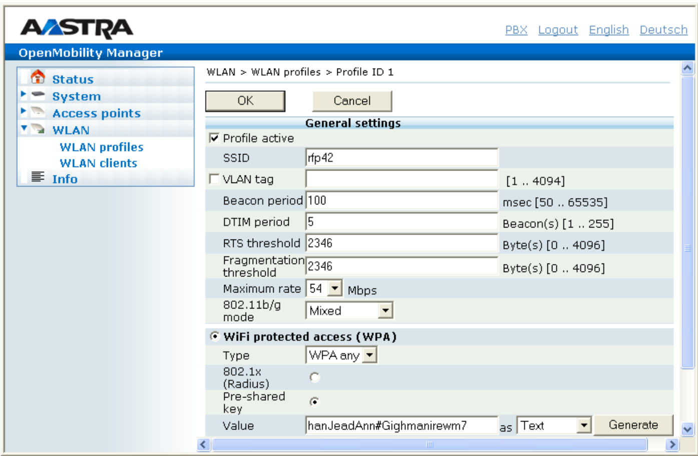
The WLAN function and the function of the DECT over IP/OpenMobility Manager cannot be used simultaneously on the same DECT over IP base station, so you will always need at least two DECT over IP base stations. The WLAN settings are then made as follows:

1. Set up the existing DECT over IP base stations in theOpenCom 100's **Configurator**. Go to the DECT over IP Manager's Web Configurator.
2. On the **WLAN Profiles** page, configure at least one set of settings (see below under: *Setting up a WLAN Profile*). Note down the password you have used ("Pre-Shared Key"), so that you will be able to use it again later when setting up wireless user terminals or notebooks.
3. Assign the desired WLAN profile on the **Radio Fixed Parts** page. Click on spanner symbol ( ) on the left next to the relevant DECT over IP base station. Under **WLAN Settings** select the number of the configured **WLAN Profile**. Confirm your settings with **OK**. You can use one profile for multiple DECT over IP base stations.

You can now use the WLAN function of your WLAN-enabled DECT over IP base stations and set up the user terminals as required.

Setting up a WLAN Profile

The WLAN function of the RFP 42 DECT base station also includes such rarely-required features as networks for large company premises or airports. In this guide we will, for the sake of brevity and clarity, describe only those features required for secure standard operation.



DECT over IP/OpenMobility Manager: WLAN Profiles

Use the following settings for standard operations.

General Settings

- Select the desired **WLAN Profile** and activate the **Profile Active** option.
- Enter a **SSID** (Service Set Identifier, wireless network identification) to identify a network. This network identification is transmitted at regular intervals, making it easier to find the networks you're looking for, using the "View available wireless networks" function in Windows XP, for example.
- For standard operation you should leave the following settings at their default values: **VLAN Tag** at 0 (Off), **Beacon Period** at 100 ms, **DTIM Period** at 5, **RTS Threshold** at 2347 (Off), **Fragmentation Threshold** at 2346 (Off), **Maximum Bitrate** at 54 Mbit/s, **802.11b/g Mode** at "Mixed" and **Interference Avoidance** on "Off".

Tip: If you are using only modern WLAN cards with 802.11g, you can further speed up data transfer by configuring the setting **802.11b/g Mode** to “802.11g only”.

- You can prevent the transmission of wireless network identification (SSID) with the **Hidden SSID Mode** setting. This will however make network identification difficult and does not generally increase data security, so it is preferable to leave this on the default setting of “Off”.

Security Settings
On no account should you use **Open System** or **Wired Equivalent Privacy (WEP)** settings**,** whether out of convenience or in order to avoid configuration problems, unless of course you want to start up an Internet Cafe!

- Activate the **Wifi Protected Access (WPA)** option.

- Under **Type** select the “WPA v.1”setting. If you are running the Microsoft Windows XP operating system from ServicePack2 or higher on your computer, you can use the “WPA v.2” setting.

- For standard operation select the **Pre-Shared Key** option. Enter a password in the **Value** input field and leave it set to **Text**. Use a password with the following characteristics:

  – No words or names that can be found in a dictionary

  – At least 8 characters long

  – It should also include numbers, a mixture of upper and lower case and special characters

  You could also use the **Generate** button to generate a password. Some WLAN configuration software does not convert text into hexadecimal values as a standard procedure. If this is the case, go to the **Hex Value** setting and select the **Generate** button.

- Leave the **Cipher Length** setting at 256 Bit and the **Distribution Interval** setting at 120 seconds. You will not usually need the settings for **WME** or for configuring **Multiple SSIDs** for standard operation.

Tip: If you are running an Internet-Cafe without using powerful encryption, you should, for the sake of your customers’ security, prevent them from being able to access each others' computers. Activate the **BSS Isolation** option. You can also stop unpleasant guests from using the system with a **MAC Address Filter** – but this will not hold up users who know about this function for long.

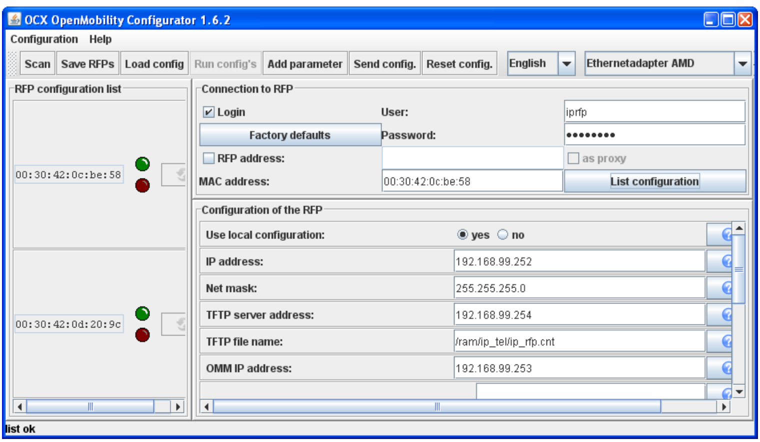
10.2.4 Configuring for a Remote Location

If you are using a DECT over IP base station in the same LAN as the OpenCom 100, the IP address configuration and software loading procedure which are run when a DECT over IP base station is started are handled by the OpenCom 100 using the DHCP and TFTP protocols.

For the DHCP function to be available, the DECT over IP base station must be able to reach the OpenCom 100 via a "Broadcast". In the case of a remote location this kind of access - via a VPN connection for example - will not be possible. As in the case of an IP system telephone, you will have to acquire the necessary system software for the DECT over IP base station with the help of a TFTP server.

Local IP Address Configuration
The IP address configuration can be set up as a "local configuration" with the help of an additional programme.

1. Call up Windows explorer. Browse to the communications system's product CD. In the "Aastra" directory, double-click the "OM_Configurator.jar" file.



OpenMobility Configurator

Note: You need a Java runtime library to be able to run Java programmes. If you don't have one installed, you can download the installation programme from the http://www.java.com/ web site.

2. To log in to the dialogue, you enter:

User
Password

3. Enter the **MAC address** of the DECT over IP base station. The MAC address will be printed on the label on the DECT over IP base station's casing. Click on **List configuration**.

The DECT over IP base station's current configuration will be displayed.

4. Change the DECT over IP base station's IP address configuration. Activate the **Use local configuration** option ("yes") and enter the required details:

– **IP address**: Static IP address of the DECT over IP base station

– **Net mask**: Subnet mask of the DECT over IP base station

– **OMM IP address**: IP address of the DECT over IP Manager. For the actual DECT over IP Manager simply repeat the entry from the **IP address** entry field.

– **OMM port number;** Leave the default setting on "16321".

– **PBX IP address**: IP address of the OpenCom 100

– **PBX port**: Leave the default setting on "8099".

5. Under **TFTP server address** enter the IP address the operating software is to be downloaded from. This will usually be the Media Gateway card's IP address (see *MGW Interface Card* starting on page 114). Leave the **TFTP file name** setting on the default setting ("/ram/ip_tel/ip_rfp.cnt").

6. For a remote location, the OpenCom 100's LAN will usually be accessed via a (VPN) router. Click on **Router addresses**: **[+]**. Enter the router's IP address ("default gateway"). Confirm by clicking on **Add**.

7. Click on **Send config** to activate the desired IP address configuration.

11. PBX Cascading

As requirements grow, the OpenCom 100 can be operated together with other PBX installations. If you merely need a larger number of connections, it is easy to link a second PBX (PBX Cascading). If you want to operate the OpenCom 100 at several locations with different PBXs, this is possible by PBX Networking (see page 153).

Please also refer to the short installation guide “OpenCom 100 cascading set” which is available as PDF file.

Note: It is not possible to cascade an OpenCom 131 with another communications system.

11.1 Variants of PBX Cascading

You can combine two PBXs in order to increase the number of terminals that can be connected. A master PBX and a slave PBX are connected to one another by means of two cables. The two PBXs essentially function like a single PBX with a higher number of ports. The master PBX controls the slave PBX. The following PBXs from the product family can be used for cascading:

Master system	**Slave system**
OpenCom 130	OpenCom 130
OpenCom 150 Rack	OpenCom 150 Rack

For the PBX Cascading you will need a licence. The licence agreement provides you with the necessary steps to activate this function.

11.2 Functionality of PBX Cascading

PBX cascading requires two twisted-pair leads with RJ45 plugs between the PBXs:

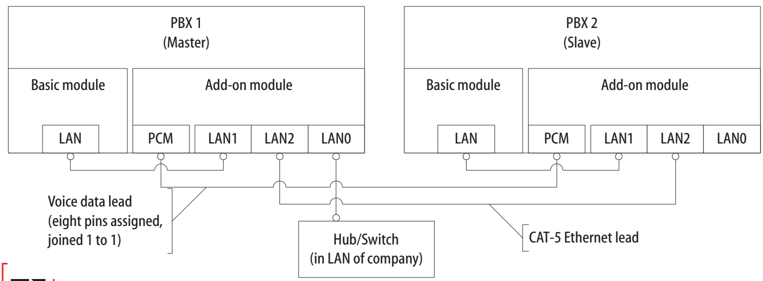
- Voice data: one lead with all eight pins wired 1 to 1. Connect this to the PCM ports of the PBXs. The shielded CAT-5 lead may be up to three meters long.

The PCM port is on the add-on module of the OpenCom 130. You must therefore install an add-on module in each of the two PBXs before they can be cascaded.

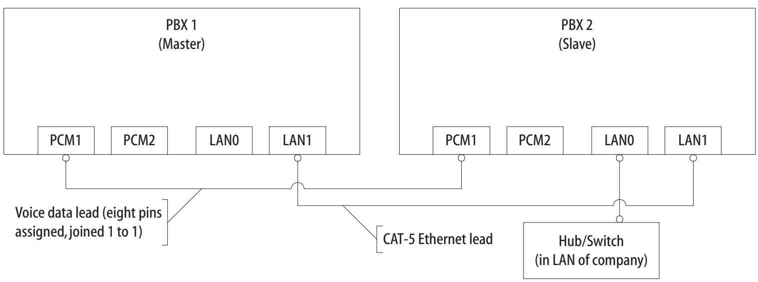
- Administration data: one CAT-5 Ethernet lead.

  – In the case of an OpenCom 130, connect the LAN2 port of the master system's add-on module with a LAN2 port of the slave system's add-on module.

  – In the case of an OpenCom 150, connect the LAN1 port of the master system with a LAN1 port of the slave system.



aded PBX system (OpenCom 130)



Cascaded PBX system (OpenCom 150)

11.3 Putting a Cascaded PBX into Operation

Proceed as follows to put a cascaded PBX system into operation:

1. Take the additional slave system out of its packaging and place it in immediate proximity to the master system. Connect a system telephone to the slave system for a later performance check. Use the U_{pn}1 press-fit terminal of the basic module (only OpenCom 130) or one of the U_{pn} ports of an interface card.

2. Back up the master system data. For further information, refer to the online help topic **System: Data Backup**.

3. Switch off the master system if it is operating. Disconnect the module from the power supply by pulling out the mains plug.

4. If necessary, install the add-on module for the master system. If you use an OpenCom 130 as the slave system, you also have to install an add-on module.

5. Connect the two modules by means of two suitable cables as described above.

6. Power on the two PBXs. The order in which you do this does not matter.

7. In the **Telephony: Ports: Slots** dialogue of the master system’s Configurator, click on **Slave**. Select the slave **Type** in the **Slave: Change** dialogue.

   If a possible slave system was detected when the system was started, there is an additional entry in the **Type** field (“Online: PBX type”). If you select this entry, the settings for **Type** and **MAC address** are applied automatically.

Please note: If you change the type of slave system later on, the port settings that have been made will be deleted.

   The master system then initialises the slave system. This may involve transfer of firmware (operating software) from the master system to the slave system. The transfer process is only executed for two OpenCom 130 / 150 PBXs. This can take a few minutes.

8. Configure the system telephone connected to the slave system for testing purposes in the **Configurator**. In the **Telephony: Ports: Upn** dialogue, click on one of the additionally displayed entries of the type **Upn 1/0/n** (1: slave system, n: U_{pn} port number).

Note: Changes to the configuration while initialising the slave system may trigger error reports referring to the ongoing initialisation.

You can see that the initialisation has been completed from the display on the system telephone connected to the slave system.

11.3.1 Notes

Observe the following when operating a cascaded PBX system:

- All U_{pn}, S_0 and a/b ports of the slave system can be used with appropriate telephones. All features of system telephones on U_{pn} ports are available without restriction.
- The S_0 ports of the slave system can also be used for trunk lines or for PBX networking (see *PBX Networking* starting on page 153).
- It is not possible to operate a DECT base station on one of the U_{pn} ports of the slave system.
- The COM, actor/sensor and LAN ports of the slave system cannot be used. The LAN ports on the add-on module of slave system can be used without restriction.
- The two communications systems must have a direct Ethernet connection or be connected via a hub in order to exchange data. They cannot be connected through a router.
- The OpenCom 150 is equipped with the PCM2 port in order to realize cascading of a third PBX with a future release. Use one of the two LAN0 ports, to connect the OpenCom 150 to the Local Area Network.
- The slave system cannot be addressed directly through a LAN. For configuration, always use the Web console of the master system.
- A memory card installed in the slave system (OpenVoice, OpenAttendant) cannot be used.
- To operate the slave system again normally, you must reset it to its factory settings (refer to *Resetting the System Data* starting on page 87).

11.4 Licensing Information

A licence is required for certain additional OpenCom 100 programme packages, for example for the internal voice-mail system called **OpenVoice**.

The following information is for customers who have already obtained licences for an OpenCom 100 and would like to cascade or network their existing system with another OpenCom 100.

PBX Networking
If you network two OpenCom 100 systems with each other, you needn't generate any new activation keys but can continue to use the corresponding functions on the existing OpenCom 100. The “disadvantage” of this alternative is that you have to administer a separate configuration on each OpenCom 100.

PBX Cascading
In the case of PBX cascading, the master system administers the overall configuration.

If you have already installed activation keys on an existing OpenCom, they have to be ported to the master system.

In this case, new activation keys for the use of additional programme packages must be generated on the Aastra licence server. The licensing confirmation for the cascading contains all the information you need to carry out this procedure.

Note: You will need the serial numbers of both infocom systems for porting the activation keys. The **serial numbers** can be found in the Web console's **Configurator**, in the **System Info: Versions** menu.

12. PBX Networking

OpenCom 100 provides all the features necessary for PBX networking. You need PBX networking in the following cases:

- To operate the OpenCom 100 as a subsidiary system on another PBX. This will also allow you to use the OpenCom 100 as a DECT server, for example.
- To network the OpenCom 100 with an OpenCom 1000. In this way you can use the OpenCom 100 as a PBX for a branch office, for instance.
- To network several OpenCom 100s into a PBX system.
- To use flexible configuration possibilities of trunk lines for a OpenCom 100.

All settings that affect the configuration of PBX networking can be found in the Configurator menu **Telephony: Trunks** and in the **Telephony: Settings** dialogue under **QSIG linking**. Refer also refer to the corresponding help topics in the OpenCom 100 online help.

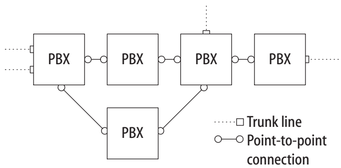
You can use ISDN point-to-point connections (Q.SIG or DSS1 protocol) or IP connections (Q.SIG-IP protocol) for TK system networking.

Note: If you do not need the features of PBX networking, the simplified configuration is sufficient in most cases. For this purpose, assign the preconfigured bundles (bundles) **Multi-terminal access** or **System access** to the ports. The preconfigured route called **External trunk** now makes it possible to seize an external line immediately or by first dialling the prefix “0”. You can rename the preconfigured bundle and the preconfigured route if required, but you cannot delete them.

12.1 Connections

Networking two or more TK systems means interconnecting them. The OpenCom 100 allows you to use the following connections:

- ISDN trunk lines
- ISDN point-to-point connections (Q.SIG) on external S_0 ports or on the S_{2M} port
- IP network connections (Q.SIG-IP)



Example of a PBX network

Various line types and transmission protocols can be used for point-to-point connections. The required network topology (distance, connection capacity) determines which type of point-to-point connection is most suitable.

12.1.1 Protocol: Q.SIG or DSS1

The Q.SIG protocol, designed for ISDN point-to-point connections, is the preferable choice as the transmission protocol; alternatively, the DSS1 protocol, designed for ISDN dial-up connections in the Euro-ISDN, can be used. Certain PBX networking features can only be used with the Q.SIG/Q.SIG-IP protocol, however. In particular, the identifier indicating whether a call is internal or external cannot be transmitted using DSS1.

Both protocols implement communication on several protocol layers:

- L1: Layer 1 defines the physical line properties and the electrical coding of signals.
- L2: Layer 2 enables communication via individual error-protected channels that are independent of each other.
- L3: Layer 3 defines the administration of the individual channels and implements the features designed for ISDN.

Note: All layers of the Q.SIG-IP protocol are symmetrical. The following are unnecessary: a Master/Slave setting, clock and synchronisation settings (please refer to *Connection via Q.SIG.IP* starting on page 158).

12.1.2 Master/Slave

For an ISDN connection, it is possible to determine which PBX is the protocol master and which the protocol slave. This relationship can be determined for all three protocol layers independently of one another.

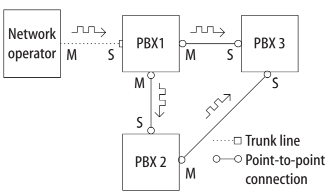
For each protocol layer, the PBX at the other end always has to be suitably configured. If one PBX is the protocol master for a layer, the other PBX must be the protocol slave for this same layer. Normally all three protocol layers are configured identically. In the case of a trunk line, the network operator is the protocol master for all three layers.

Note: In the case of an S_{2M} line, it is also possible to determine for each useful channel which end can administer a channel (master = internally seized or slave = externally seized). On S_0 lines this setting is determined by "L3 master" for both B-channels.

12.1.3 L1 Clock

To enable PBXs in the ISDN network to communicate with each other, they must be "clock-aligned". The L1 protocol master sets the clock for layer 1, and the L1 protocol slave adopts (synchronises to) this clock.

When planning a PBX networking scheme, you must make sure that the L1 clock propagates from a master via a number of PBXs.



Example: propagation of the L1 clock

If more than one port with the setting **L1 Type** = “Slave” is configured on an OpenCom 100 and the setting **L1 sync possible** has been activated, then one of the ports is automatically defined as the L1 clock source. The OpenCom 100 will automatically switch the clock source to another port configured as an L1 clock source (if a line fails, for example).

Please note: Reciprocal or circular application of the L1 clock is not allowed.

Example: In the above case you could reverse the L1 slave/master setting for the connection between PBX 1 and PBX 3. However, if you then activate the setting **L1 sync possible** for the port of PBX 1, this may cause parts of the PBX network to stop functioning temporarily.

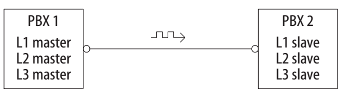
When applying the L1 clock of trunk lines, you can assume that the public network is “clock-aligned”. So, in the above example, you can connect additional trunk lines to one of the PBXs.

12.2 Types of Point-to-Point Connections

There are different types of connection available for an point-to-point connection between two PBXs, depending on the distance between them.

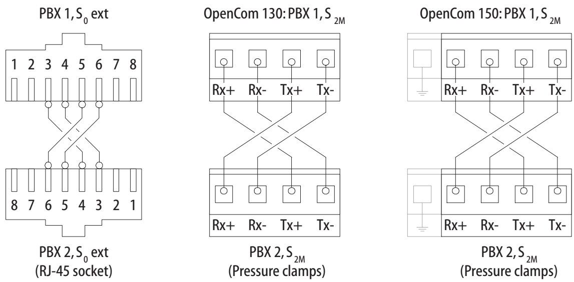
12.2.1 Direct Connection

This type of ISDN point-to-point connection joins the two systems directly to each other using a crossover twisted-pair cable. An S_0 connection can be used for distances up to 1,000 metres, while an S_{2M} connection can span up to 250 meters. Normally one PBX is the protocol master for all three layers, and the other PBX is the protocol slave for all three layers.



Direct connection

- Use the RJ45 jacks on one of the external S_0 ports for an S_0 connection between two OpenCom 100s. You can use the corresponding pressure terminals for S_0 ports on interface cards.

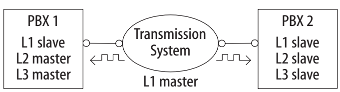


Wiring of a direct connection

Note: If you use an S_0 port on an interface card (pressure terminal) and an S_0 port with an RJ45 jack for the direct connection, make sure you make the necessary changes to the port assignment (see *S0 Ports on Interface Cards* starting on page 49).

12.2.2 Connection via an Active Transmission System

For distances exceeding the range of a direct connection, an active transmission system can increase the range to up to 50 km. Normally the L1 master is the transmission system for the two connected PBXs. For the protocol layers L2 and L3, one PBX is normally the protocol master and the other PBX is the protocol slave.

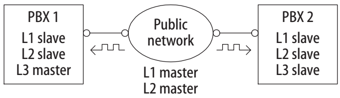


Connection by an active transmission system

Note: The active transmission system itself gets its L1 clock either from the network operator or from a clock generator connected by wire.

12.2.3 Connection via the Public Network

Point-to-point connections via the public network of a network operator can be used for bridging distances beyond 50 km. Due to the long distance involved, for technical reasons it is not possible to synchronise the L2 protocol. Consequently, the public network is normally the protocol master for protocol layers L1 and L2. One PBX is therefore the L3 master and the other PBX the L3 slave.



Point-to-point connection via a public network

12.2.4 Connection via Q.SIG.IP

If you are operating a fast and continuous internet connection at two or more locations, you can establish the TK system networking via internet connection as well. The OpenCom 100 uses the Q.SIG protocol, for use with ISDN point-to-point connections and transports the protocol and voice data via IP connections.

The number of simultaneous conversations possible will depend on the capacity of the internet connection and the compression method used. A multiple S_{2M} point-to-point connection is simulated for each Q.SIG-IP bundle. This means that 5 virtual D channels and up to 120 voice channels are available. Both Media Gateway Card channels and the Media Gateway software function can be used for Q.SIG-IP.

Q.SIG-IP connection data are subject to codec compression (please refer to *Voice over IP (VoIP)* chapter regarding *Fundamentals* starting on page 107). Q.SIG-IP also transfers the voice data directly from terminal to terminal via the RTP protocol. In certain cases, for example, when an incoming external call is placed via multiple TK systems, one or more RTP proxies may be used to forward the connection.

Currently, there are no standards for the necessary extensions to the Q.SIG protocol. This means that you can only use Q.SIG-IP between Aastra 800 and OpenCom systems.

Networking two OpenCom 100 systems using Q.SIG-IP requires 2 licences – one licence per system. The number of possible voice connections is not restricted by the licence.

Go to the **Telephony**: **Trunks**: **Bundle** page in the **Configurator** to set up a Q.SIG-IP connection. Create a new bundle and select the **Access type** “System Access”. Select “Q.SIG-IP” under **Protocol**. Configure the IP address of the other system, the port numbers to be used, the number of possible voice connections and select a VoIP profile for the codec selection. Please refer to the relevant help topics in the Online Help for the OpenCom 100 as well.

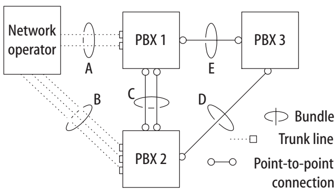
Note: Q.SIG-IP cannot be operated using a connection with NAT. For a Q.SIG-IP connection, a branch connection or another VPN connection is required.

12.3 Configuration

The possible configurations described below can be set up in the Web console using the **Telephony: Trunks** menu.

12.3.1 Bundles

A **bundle** is a group of lines of the same type and direction. A line can only be assigned to one bundle.



Example of a PBX network with bundles

In the above example, the following bundles are configured for PBX 1:

- Two S_0 lines in a multi-terminal configuration to the network operator which are assigned to the “A” bundle.
- Two S_0 point-to-point connections to PBX 2 which are assigned to the “C” bundle.
- One S_0 point-to-point connection to PBX 3 which is assigned to the “E” bundle.

Note: A line or a bundle cannot be seized directly. It is always performed indirectly via a route.

12.3.2 Routes

A **route** is a group of bundles enabling a connection in one direction. If the first bundle of a route is fully utilized, the next bundle is seized (“bundle overflow”). One bundle can also be used for different routes.

In the above example, a route set up for PBX 1 allows a connection to PBX 2. Bundle “C,” “E” and “A” are assigned to this route. If a user connected to PBX 1 wants to reach a party in PBX 2, lines will be seized in the following order:

- PBX 1 first searches for a free channel in the “C” bundle.
- If all the lines in bundle “C” are busy, the system tries to set up a connection via bundle “E”. PBX 3 switches the connection through, provided it is appropriately configured (refer to *Numbering* starting on page 160).
- If it was not possible to set up an indirect connection via PBX 3, the system tries again via bundle “A”. The “prefix” necessary for this can be configured with the route.
- The user does not get a busy signal until the attempt to set up an indirect connection via the network operator has also failed.

Note: If an internal connection is switched via a network operator, the call is signalled using the external number of the calling PBX.

For each route you can define a randomly selectable code digit for seizing the route. You can also configure whether a user is authorised to seize a particular route, whether LCR is to be used for one of the bundle and the criteria (business or private call, booking numbers) for evaluating call data.

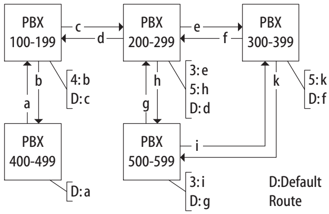
12.3.3 Numbering

A user can seize a particular route by pre-dialling a specific code digit. With this “open numbering”, a user must always dial this code digit and then the telephone number in order to reach a party in another PBX.

If none of the telephone numbers in your PBX network occur twice, you can also configure “closed numbering”, allowing the same telephone number to be used for reaching each user within the PBX network.

With closed numbering, the OpenCom 100 determines which route to seize from the telephone number dialled. The information needed for routing a call can be configured in a numbering table containing up to 100 entries. You use this table to assign telephone numbers and/or ranges of telephone numbers to a particular route.

A **default** entry in the numbering table makes it possible to seize a “default route” for all remaining unassigned numbers. In particular, this simplifies configuration of the OpenCom 100 as a subsidiary system: the only entry you assign to the **default** entry is the route to the host system



Example of closed numbering tables

The automatic switching of call requests (i.e. routing) by means of bundle overflow or default numbering can lead to “circular switching”.

To avoid this, a “transit” counter is incremented whenever a connection is switched through on Q.SIG lines. When the configured maximum value is reached, further switching stops.

12.4 Technical Details

A different PBX number must be set for each OpenCom 100 in a PBX network. This setting can be found in the Web console, in the menu **Telephony: Settings** under the heading **QSIG linking**. You can also set the maximum value for the transit counter there. This value depends on the topology of the PBX network and should allow the system to have the maximum number of further connections possible.

You can display the connection status of the lines at any time in the Configurator menu **System Info: Telephony: Trunks**. You should check this in particular after making changes to a configuration to see whether all the lines used for system networking are operable.

Some of the features possible in Q.SIG are not supported by OpenCom 100 with all their options, for example callback on busy within the Q.SIG network. The call categories defined in Q.SIG (e.g. Emergency Call, Operator, Normal) and the Q.SIG name transmission feature (“user names”) are fully supported.

The code digits to be used for seizing a route with open numbering are not transmitted to the destination PBX and thus cannot be evaluated by it. To reseize a route (for example for a callback), you must set the appropriate digit prefixes in the bundle configuration for the routes to be reseized.

Tip: If, for example, you are configuring a route which can be seized using routing code "5" and have selected one or more bundles for this route, change the **Prefix for dest. call number at incoming internal** setting to "5" for this bundle in order to enable the route to be reseized.

Owing to their hardware properties, not all S_0 ports of the OpenCom 100 can be used for PBX networking without restrictions. Depending on the type of system, some ports can only operate in the L1 master mode or L1 slave mode. The external S_0 and S_{2M} ports can be set according to the following table.

	S_0 1	**S_0 2**	**S_0 3**	**S_0 4**	**S_{2M}**
OpenCom 130	S	M/S	-	-	M/S
OpenCom 150	-	-	-	-	M/S

Legend
S = Slave
M/S = Master/Slave
M = Master

Note: The S_0 ports on add-on cards can be operated in both L1 master and L1 slave mode.

13. Telephony

13.1 E.164 conversion

The OpenCom 100 communications system supports two different types of call numbers when dialling external call numbers. Usually you enter the code for a route, e. g. a “0” for the “external line” route. Then you enter an external call number. The external call number can be either a local area code or a country area code.

The additionally configurable “E.164 conversion” feature enables you to enter the entire international call number. Using the international call number makes sense in the following applications:

- When using the “Fixed Mobile Conversion” (FMC) feature as the call number is usually dialled from the local directory of the mobile telephone being used.
- When dialling via a computer programme connected with TAPI where call numbers are often already in the international format due to synchronisation with a mobile telephone.
- When importing or comparing directory data when directory entries are in the international format.
- When networking communications systems with locations in different local areas or in different countries.

The “E.164 conversion” analyses an international call number. The analysis divides the call number into multiple parts: the international area code, the local area code, the access call number and if necessary the extension. The call number is respectively abbreviated, eliminating any unnecessary area codes. The abbreviated call number is then used, e. g. to execute a call.

13.1.1 Configuration

You can configure the “E.164 conversion” feature for each bundle separately. This is possible for point-to-point configured bundles (“system access”), for point-to-multi-point configuration (“multi-terminal access”) and for SIP trunks:

1. Call the OpenCom 100 communications system **Configurator**. On the introductory page, change the **Level** option to **Expert**.

2. This step is optional as the country code is preallocated due to the **Country** setting under **System**: **Common**.

   Call the **Telephony**: **Settings** page. Click on the **Change** button. Enter the country code without a zero in front into the **International area code** field, e. g. “49” for Germany. The setting in the **Own area code** field is not relevant for the “E.164 conversion” feature. Confirm with the **Apply** button.

3. Call the **Telephony**: **Trunks**: **Bundle** page or the **Telephony**: **Trunks**: **SIP trunks** page. Click on the desired bundle or desired SIP trunk. Activate the **E.164 conversion** option.

   The following setting is only relevant if the local area code is not part of the ISDN-MSN (Germany and Austria) in your national ISDN. If you have selected “Germany” or “Austria” under **System**: **Common** as the **Country** setting, you also have to enter the local area code into the **Area code** field for the bundle. The prefixed zero is not necessary.

   Confirm with the **Apply** button.

4. Call up the **Telephony**: **Trunks**: **Route** page. Check whether the **Type** setting for the used routes is set to “Private” or “Business”. For routes of the type “Internal” the “E.164 conversion” is not active.

The Q.SIG bundle used for networking communications systems cannot be used with the “E.164 conversion” feature. Please keep in mind that the differentiation between access call number and extension only takes place with system access or DDI-capable SIP trunks and also only when using direct extensions. With a call number allocated via call distribution there is no automatic differentiation between external and internal call numbers.

Note: For incoming calls via a bundle with the “E.164 conversion” feature each external call number (“CLIP”) appears converted into the international format and is also saved in this format e. g. in the caller list. Keep this in mind when entering call numbers used for authenticating (CLIP-Auth).

13.1.2 Example

The following example explains the “E.164 conversion” function on a terminal which is operated on a system access with the following configuration:

Attribute	Number
International area code (Country Code, CC)	49 (Germany)
Local area code (National Destination Code, NDC)	30 (Berlin)
Access call number (Subscriber Number, SN)	6104 (Aastra Berlin)
Extension or internal call number (Direct Dialling, DDI)	4666 (Sales Support)
Code for the “External trunk” route	0

Various call numbers are now dialled from this terminal:

Number dialled	Number actually used
0003311234567 (foreign country, Paris)	003311234567 The code for the route is not transmitted via ISDN.
00049401234567 (domestic, Hamburg)	0401234567 The international area code is replaced with a “0”.
00049301234567 (domestic, Berlin)	0301234567 The international area code is replaced with a “0”. One's own local area code is not removed.
000493061042007 (Internal)	2007 The international area code and the access call number are deleted. The destination is called internally without using the ISDN connector.
+493061042007	2007 You can only enter the plus sign (after E.123) with a SIP telephone (see *Internal SIP Subscribers* starting on page 117). There is a “E.164 conversion” in this case also.

Number dialled	Number actually used
003061042007	03061042007 There is no “E.164 conversion” without an international area code.

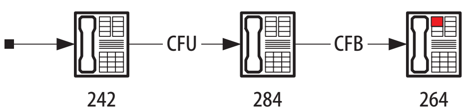
13.1.3 Further Information

Please note the following information when using the “E.164 conversion” feature:

- When all external lines are occupied, the “Congested” state is only indicated later on during the dialling process.
- Emergency calls are always executed without “E.164 conversion”.
- Call number assignments in the call distribution are not evaluated for the “E.164 conversion” feature. For an assigned MSN, e. g. no automatic internal dial is executed even when the destination could be reached this way.
- Depending on the telephony provider, you can possibly also use the area code of your own country without “E.164 conversion”. Differentiating between external and internal call numbers can only be done using the “E.164 conversion”.

13.2 Call Forwarding

The current version of the OpenCom 100 communications system also offers configuration of multi-level call forwarding. When you forward a call number that has already been forwarded, this results in a call forwarding chain.



Call Forwarding Chain

Multiple call forwarding is executed independent of the call forwarding type. An overview of the various call forwarding possibilities is nonetheless helpful for the following explanations.

Call Forwarding

Name	Description
Call forwarding immediately (CFU)	Immediate and unconditional call forwarding
Call forwarding on busy (CFB)	Call is forwarded only if user is busy
Call forwarding after time (CFNR)	Call forwarding is only executed after a definable time interval
Call Diversion (CD)	Is manually executed upon an incoming call from the user
Virtual call number	A virtual call number is always diverted to a destination call number
Call forwarding of a hunt group	Users of a hunt group can also configure respective call forwarding
Call forwarding to external	Call forwarding to an external call number or via remote-controlled dialling (Call Through)
Call forwarding by a system user	Call forwarding via OpenAttendant (with the **Connect to phone number** and **Connect to voicebox** function) or via OpenVoice (with the secretarial function)

13.2.1 Attributes

A call forwarding chain can contain any call forwarding types and call forwarding users. There is no limit to the number of successive call forwarding instances.

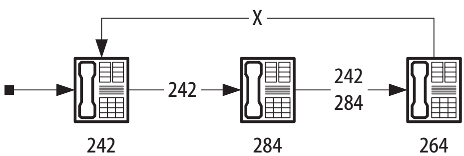
If the call forwarding destination is a system telephone, an incoming call is additionally indicated with the display `via...`. The caller list of a system telephone can also determine both the call number of the caller as well as the call number of the user doing the forwarding.

In the case of multiple forwarding, a setting in the user group of the call forwarding destination determines which of the “Via” call numbers is displayed. You can have either the last call forwarding user (default) displayed or the first call forwarding user in a chain displayed. The “Via” call number is, however, only displayed when the call number display is activated for the forwarding user.

Note: When call forwarding to the **OpenVoice** voicebox programme, the “Via” call number is evaluated in order to determine the owner of a voicebox. The last call forwarding user is used no matter what the user group setting. When call forwarding using the secretarial function of the voicebox programme, the call number of the voicebox owner is also shown as the “Via” call number.

13.2.2 Loop Detection

Loops can generally occur during a call forwarding chain, e. g. when the call forwarding destination refers back to the call forwarding source. This is why a forwarded call has a call forwarding history. When the next call forwarding destination is already included in the call forwarding history, a loop is detected and any further call forwarding is prevented. If no parallel call signalling takes place, e. g. by the **Indicate call forwarding after time parallel** setting in the user group, the call is cleared when there is a loop.



Loop Detection

A loop is also detected during call diversion. When you wish to divert an incoming call to a destination call number which is already part of the call forwarding chain, the display shows `NEG.` and call signalling is continued.

Note: The call forwarding history cannot be transmitted via Q.SIG connections. Chain detection is also deactivated when forwarding via the voice portal programme **OpenAttendant**.

13.2.3 Virtual Call Numbers

A virtual call number is not assigned to any terminal. You always also enter an internal or an external destination call number directly when configuring a virtual call number. When the virtual call number is called this destination call number is signalled. This behaviour is handled as an immediate call forwarding and is thus the first call forwarding in a possible call forwarding chain.

You can include a virtual call number e. g. in call distribution. Using the possibility of multiple call forwarding, you can also use a virtual call number as an exchange ("Operator"). Furthermore, a user with the **Call forwarding for other user** authorisation can configure additional call forwarding for a virtual call number also.

13.2.4 Hunt Groups

A hunt group is an internal call number which can reach multiple users. An incoming call is signalled to all users of a hunt group. A hunt group is configured in the **Configurator** in the **Telephony**: **Groups**: **Hunt groups** menu. Call signalling is independent of the **Type** setting in the hunt group:

- **Parallel** setting: all users of the hunt group are called simultaneously.

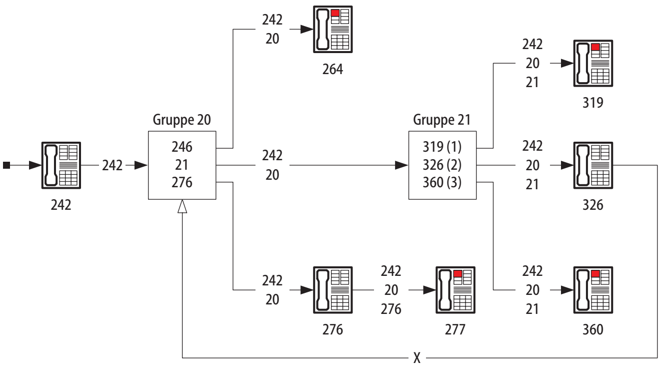
- **Linear**, **Cyclical** or **Statistical** setting: the users of a time-dependent hunt group are call successively. The user called first is determined by the respective setting.

The users of a hunt group can also configure their own respective call forwarding. For this, you need to activate the **Call forwarding of members possible** option for the desired hunt group on the **Telephony**: **Groups**: **Hunt Groups** page. The call forwarding history is forwarded to multiple users and continued separately respectively. When a call forwarding loop is detected for a user, no further call signalling takes place for this user. This user is not called in a parallel hunt group. This user is skipped in a time-dependent hunt group.

Using a call forwarding chain, you can include a further hunt group as a user in a parallel hunt group.

Note: Time-dependent hunt groups do not permit including any additional hunt groups.

In addition, you can also configure call forwarding on busy (CFB) for hunt groups. This call forwarding can be used for an internal or external destination call number. In addition, you can enter a back-up destination for time-dependent hunt groups when they cannot be reached. Using the **Forward after time** setting, you can instead configure a call number and a specific time interval in the **Telephony**: **Groups**: **Hunt Groups** menu.



Forwarding hunt groups

13.2.5 External Call Forwarding

You can also forward calls to external destinations without restriction. However, the call forwarding history cannot be continued with external destinations. When multiple users of a parallel hunt group configure call forwarding to external destinations, a single call to the hunt group can also occupy multiple external lines.

Special rules apply for call numbers which are displayed upon an external call forwarding destination:

- If the call is originally initiated by an internal user, the external call number of the internal user is transmitted.
- If the call is originally initiated by an external user, this user's external call number can be transmitted using the“CLIP no screening” feature. If “CLIP no screening” is not available, the external call number of the last forwarding user is transmitted.

13.2.6 Information on the Update

It is possible to convert the configuration of an earlier firmware version to the current version within the context of an update. This happens automatically when you load a saved configuration after an update. Please note the following points:

- The previous behaviour is preserved with the conversion. When restoring a former configuration, the **Call forwarding one level only possible** option is switched off automatically on the **Telephony**: **Settings** page. Additionally, the **Call forwarding of members possible** option is deactivated for all restored hunt groups.
- Restored call forwarding instances can behave differently as call forwarding chains of earlier firmware versions (before the 9.0 release) are not supported. If necessary, use the trace function for call forwarding (the **Diagnosis**: **Trace** menu, **CF tracking** option in the **Configurator**).
- The call forwarding interval, previously only centrally definable, is adopted for all users as the default value.
- The “Show hunting group no. as via” option for hunt groups has been omitted. Instead, use the **Display: Call forwarding via** option in the **User Manager**: **User groups** menu.
- The “Courtesy Service” function used to be for directly determining whether a user was currently busy or not and the corresponding announcement played. The **Announcer at free** is always played when it is configured with the firmware version (9.0 release). Then **Announcer at busy** is played if applicable.

13.3 PIN Code Telephony

The users in a company normally use the existing terminals primarily for company-related communication. Now the users would like to also be able to make private calls in some cases. Private calls require:

- seizure of special external lines,
- changed and valid authorisations,
- recording separate charges and

- that destination call numbers not be saved for redialling.

This function can be realised using the “PIN Code Telephony” feature. To initiate a special call, the user on any terminal uses a special menu function or code procedure. After the user enters his/her own internal call number and associated user PIN, the desired attributes for the next call are activated.

When the call is finished, the previously active telephone configuration is restored. This external call number dialled is not recorded in a redial list.

13.3.1 Configuration

You can configure the “PIN Code Telephony” feature separately for each company. Configuration is done with the following steps:

1. Call the OpenCom 100 communications system **Configurator**. On the introductory page, change the **Level** option to **Expert**.

2. Open the **User Manager**: **User groups** menu page. Click on the **New** button. Enter a designation into the **Group name** field and select an existing user group as a template. Confirm with **Apply**.

3. Then click on the **Change** button. Select the desired **Company**. Activate the **PIN dial** option for the time groups intended. Configure either external authorisation or recording of **Connection data**. Confirm with **Apply**.

4. Open the **Telephony**: **Extended**: **Companies** menu page. Click on the headline of the company desired (default: “Company 1”).

5. Under the **PIN dial** header, select the desired **Route**. Select the user group created by the previous steps for the **User group** setting. Confirm with **Apply**.

You can also use the “Standard” user group or the “External trunk” standard route for the “PIN Code Telephony” feature. If you configure a special route, you must also enter a seizure code for this route (**Telephony**: **Trunks**: **Route** menu, setting: **Code**).

Activate the **PIN dial** user group option for all terminals which are to have the “PIN Code Telephony” feature available to them. This can also be the “Guests” user group. Furthermore, you can also use the “PIN Code Telephony” feature for users who have no terminal assigned to them (“virtual users”).

13.3.2 Implementation

You would like to use the “PIN Code Telephony” feature from any telephone. To do so, the telephone belonging to another user is switched to your personal user account (“identity change”). Carry out the following steps:

1. Call the main menu on a system telephone. Select the `6 Connections`: `7 PIN dialling` menu entry. Enter your call number and your user PIN. If the entry is correct, `<PIN dialling>` now appears. Now select the desired external call number including the seizure route for the “External Trunk” standard route.

   You can also initiate PIN dialling with a code procedure:
   # 4 6 (your no.) * (user PIN) # (external no.)

   The route configured for the “PIN Code Telephony” feature is used to make a call in that the seizure code for the standard route is also replaced by another seizure code if applicable. Furthermore, the authorisations are activated for the user group designated for the “PIN Code Telephony” feature.

2. Make your call. Please note that the call number identity being used is displayed to the external caller even when you are calling from a different terminal. In addition, the call number identity being used is indicated as busy during the call. This is why the corresponding trunk key also lights up during this time on the respective system telephone.

The external call number dialled is not saved in any redial list: neither on the telephone used, nor one's own telephone.

13.4 Switch authorisation

The “Switch authorisation” feature enables a user to switch the user group of another terminal for a single call. The user switching can receive a charge notification at the end of the call.

A typical application is the guest telephone in a hotel: the concierge on duty activates external dialling for a guest when desired. When the call is finished, the concierge is informed of call duration and relevant charges with a brief information display.

13.4.1 Configuration

You can configure the “Switch authorisation” feature separately for each company. Configuration is done with the following steps:

1. Call the OpenCom 100 communications system **Configurator**. On the introductory page change the **Level** option to **Expert**.

2. Open the **User Manager**: **User groups** menu page. Click on the **New** button. Enter a designation into the **Group name** field (e. g. “Guest external seizure”) and select an existing group as a template. Confirm with **Apply**.

3. Then click on the **Change** button. Then select the desired **Company**. Configure the external authorisation and recording to **Connection data** as desired. Activate the **Immediate External line seizure** option as desired. Deactivate the **Switch authorisation** option to prevent authorisation from being activated for an additional call in an unauthorised manner. Confirm with **Apply**.

4. Click on the user group for the user who is to execute the “Switch authorisation” feature. Activate the **Switch authorisation** option. Confirm with **Apply**.

5. Open the **Telephony**: **Extended**: **Companies** menu page. Click on the header of the company desired (default: “Company 1”).

6. Under the **Switch authorisation** header, select the **User group** temporarily active due to switching (e. g. “Guest external seizure”). Activate the **Notify at end of call** option as desired. Confirm with **Apply**.

Devices to be switched (“guest telephones”) are generally configured in a user group without external authorisation. If desired, you can decline to assign these terminals to a user. In this case, the “Guests” user group is automatically assigned.

13.4.2 Implementation

A hotel guest would like to make an external call. You fulfil this wish with the following steps:

1. Call the main menu on your system terminal. Select the `6 Connections: 6 Switch auth.` option. Enter the internal call number of the terminal where a temporary authorisation switch is to take place. Confirm with the `OK` key. Select the `On` option and confirm with the `OK` key.

If the terminal to be switched is currently in the call state, the authorisation switch only takes place once the call is finished.

2. The next outgoing call on the switched terminal then is made with the changed authorisations. If, e. g. the **Immediate External line seizure** option was activated to do so, the external dialling tone is audible and an external call can be made without entering a code.

   If the next outgoing call is not made within 60 seconds, the authorisation switch expires automatically.

At the end of the call – when configured in this way – you receive a brief message regarding the call duration and the relevant charges incurred.

14. Fixed Mobile Convergence

The “Fixed Mobile Convergence” (FMC) feature offers operation of mobile telephones on the OpenCom 100 communications system. Connections to and from the mobile telephone are directed via the communications system. This is done by the communications system managing the mobile telephone as an internal terminal with its own internal call number. This is how all features of the system telephony can be used on the mobile telephone. Connection expenses billed on the mobile network are limited to the connection to the communications system. When the mobile telephone user establishes an external connection (e. g. an international call) via the communications system, the expenses for the international call are apportioned to the system.

Of particular interest is parallel operation of a stationary system telephone and a mobile telephone that can be reached under the same internal call number. There is no difference in how both terminals can be reached by callers: an incoming call is signalled on the system telephone and on the mobile telephone at the same time. When a call is accepted on the mobile telephone, the terminal of the caller does not display the mobile call number, rather the office call number administered on the OpenCom 100 communications system is displayed, which can be used to reach the system telephone as well. It is possible to take an ongoing call from the mobile telephone on the system telephone (and vice versa). System features such as e. g. enquiries and forwarding are available with calls via the mobile telephone just as with calls via the system telephone. An outgoing call using the mobile telephone is executed via the same call number as a call from the system telephone.

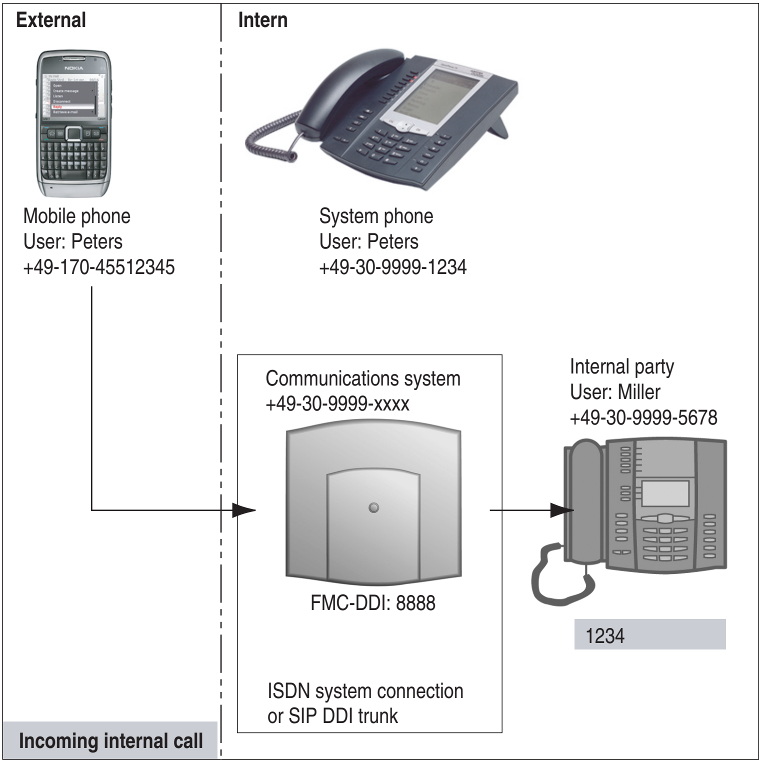
The FMC feature requires no additional hardware. The technical solution comprises the following:

- Each mobile telephone (= FMC telephone) is assigned an internal call number which is used for the configured mobile call number. The radio technology of the mobile network (GSM, UMTS or CDMA) is not relevant.

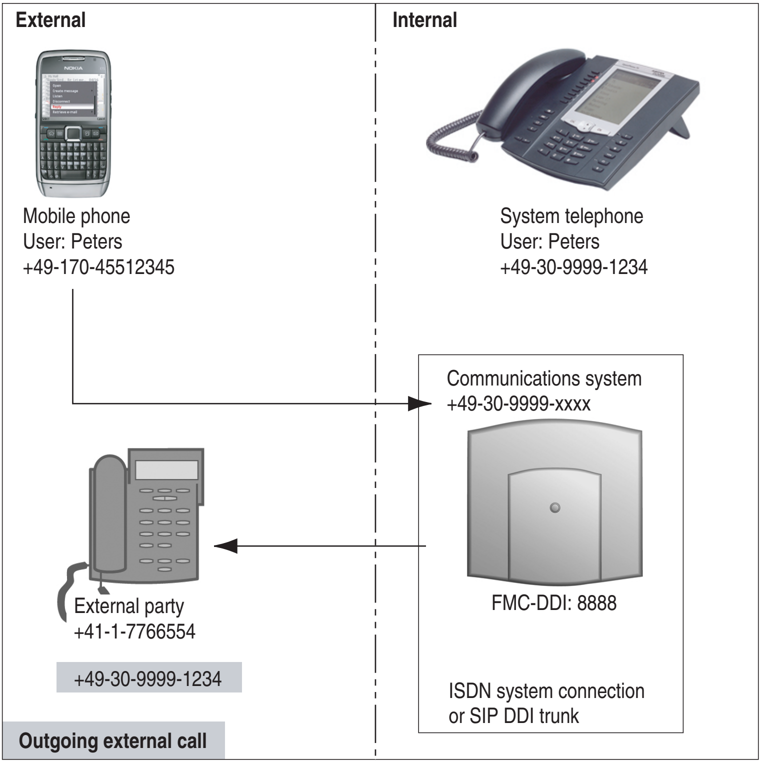
- A mobile telephone is generally integrated with the communications system via an ISDN system connection or via a DDI-capable SIP line. An internal FMC call number is configured for the connection between communications system and mobile telephone. A further internal FMC call number can be additionally configured for transmission of MWI such that the mobile telephone user can be informed of waiting messages (e. g. voicebox messages).

- Call number information such as e. g. the call number of an external caller can be transmitted using the “CLIP no screening” feature available via the system connection and thus be displayed on the mobile telephone.
- There is an information exchange via DTMF tones during a voice connection between the communications system and the mobile telephone. This means that e. g. the “R” key function is transmitted via a DTMF sequence with three star signs.
- A mobile telephone call with the FMC call number is considered the beginning of the log-in procedure. After authentication via the call number (CLIP) or via a special DTMF sequence, there is an audible internal dial tone on the mobile telephone. The mobile telephone user can now dial any internal or external call number and use system features.
- There is special software available for a series of mobile telephones: “Aastra Mobile Client”. This software makes it convenient to use FMC telephony functions.

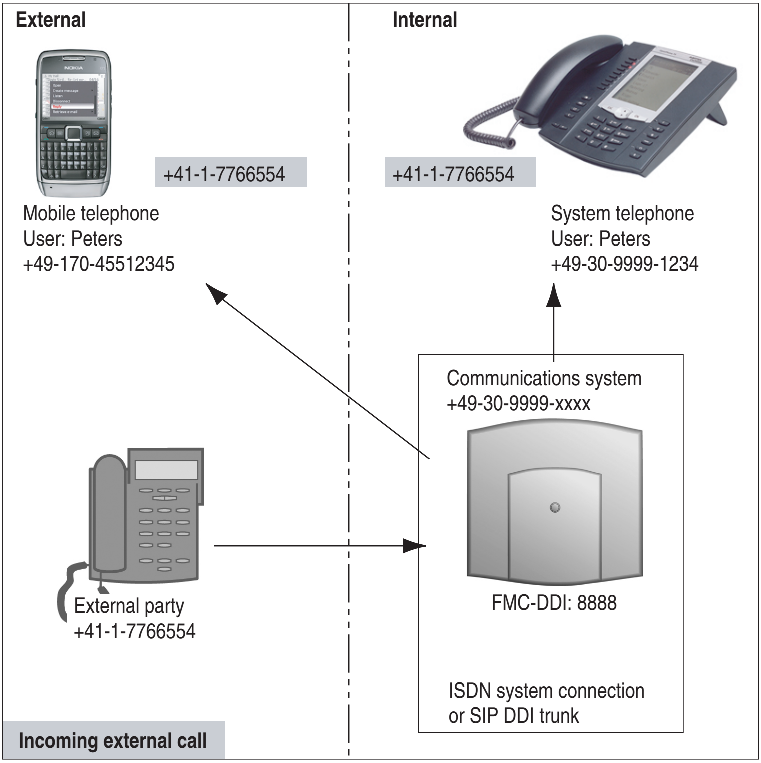
The following diagrams illustrate the integration of mobile telephones with the OpenCom 100 communications system via FMC using three sample call situations:



Outgoing internal call: Call by an internal user from a mobile telephone



Outgoing external call: Call by an external user from a mobile telephone



Incoming external call: Call by an external user which is signalled on both the mobile telephone and on the system telephone

Licence

Configuring FMC telephones on the OpenCom 100 communications system requires a licence. With this licence you can also install a corresponding number of FMC clients ("Aastra Mobile Client"). For more detailed information, please consult the licence confirmation you received.

Documentation

The following sections provide information on configuring the "Fixed Mobile Convergence" (FMC) feature and on configuring FMC telephones. During configuration you can also use the online help of the OpenCom 100 communications system as a source of information.

For users whose mobile telephones have the "Aastra Mobile Client" software installed on them, there is a separate user guide available: "Fixed Mobile Convergence – Using mobile telephones on the Aastra 800 / OpenCom 100 communications system”. This manual is available as a PDF file on the OpenCom 100 communications system product CD.

14.1 Configuring FMC Telephones

The following steps are for activating the FMC feature and to configure FMC telephones:

1. In the **Configurator** of the OpenCom 100 communications system, call up the menu page **System**:**Licences**.Check to see whether the **Aastra Mobile Client** entry has a green checkmark next to it, indicating there is a valid licence. Otherwise you have to authorise the feature first.

2. Call the **Telephony**: **Devices**: **FMC Phones** menu page. Click on the **Change** button. Enter an internal call number into the **FMC Phone No.** field which a mobile telephone uses to dial into the communications system. Or you may enter another internal FMC call number into the **MWI Signaling phone number** field for signalling calls from the communications system. Confirm with **Apply**.

   These call numbers can usually be reached as extension numbers from external sources or be called from external sources via the system connection. On the **Telephony**: **Call Distribution** menu page, check the settings for the **Incoming DDI** and **Outgoing DDI** call distribution, and assign extension numbers to the FMC call numbers as needed.

Note: You should create an “FMC” user in the **User Manager** and assign the two FMC call numbers to this user. MWI signalling requires the user belong to a user group with external dialling authorisation.

3. Click on the **New** button to create a new FMC telephone. Enter a new internal call number into the **Phone No.** field. You have to then assign this call number to another existing user or create a new user. You can also select a previously configured internal call number for parallel operation of an existing terminal and mobile telephone under the same internal call number. Enter the desired mobile call number with the prefixed line seizure code into the **Mobile Phone No.** field.

The authorisations of the respective FMC telephone are used for call forwarding to the mobile telephone. The assigned user thus has to belong to a user group with external dialling authorisation.

4. Activate the **Auto-Login (CLIP)** option to use the accelerated log-in procedure via the mobile telephone call number. Keep in mind that the **Mobile Phone No.** field has to correspond to the CLIP transmitted by the mobile telephone. If you use the “E.164 dialing” feature you have to enter the international area code prefix in the **Mobile Phone No.** field.

Note: Mobile telephone calls using the “CLIP no screening” feature are rejected by the communications system.

Alternatively, mobile telephone log-in can be done using a password procedure. This requires entering a user PIN.

Note: For security reasons users of FMC telephones should never retain the default “0000” as their user PIN, rather create their own individual PIN.

5. Activate the **MWI Signaling** option, if new voice box messages should trigger a signaling call. The signaling call will be rejected from the “Aastra Mobile Client”, which in standard cases will generate no extra charges. Deactivate the **MWI Signaling** option, if you do not want to use this function for the new FMC telephone.

6. To create the new FMC telephone confirm with **Apply**.

7. If you also wish to use the “Aastra Mobile Client” software on the mobile telephone, you can manage this software on the configuration platform available on the internet (see *Configuring “Aastra Mobile Client” Software* starting on page 183).

Tip: As mobile telephones manage all call numbers in the international format, you should also configure the “E.164 dialing” feature on the OpenCom 100 communications system for the system connection used or the SIP line used (see *E.164 conversion* starting on page 163).

14.2 Configuring “Aastra Mobile Client” Software

System telephony features (e. g. enquiry, forwarding) can be conveniently used on the mobile telephone (FMC telephone) when you install the “Aastra Mobile Client” software on the device. Comparable to a system terminal, this makes many features available on the mobile telephone via a separate menu. The “Aastra Mobile Client” software runs under Symbian, a widespread operating system for mobile telephones. Specific models by the manufacturers LG, Nokia, RIM and Samsung are supported. Please contact your retailer if you have questions or if you desire further information on the supported devices.

You manage the installation, settings and licensing of the “Aastra Mobile Client” software on an internet-based configuration platform:

1. Open a Web browser and call the internet address provided in the licence confirmation.

   Logging in to the configuration platform requires the information also provided in the “Aastra Mobile Client” licence confirmation. You then use a licence key to activate a specific number of administrator and software licences.

Note: The configuration platform language depends on the language setting of your Web browser.

2. Create a new entry for each FMC telephone. This entry includes the FMC call number of the OpenCom 100 communications system and the mobile call number of the FMC telephone. The following table explains the various settings.

Setting	Explanation
GSM number	the mobile call number assigned to the FMC telephone (E.123 notation with prefixed plus sign)
Automatic start (Only in the **Settings** menu of the “Aastra Mobile Client” software)	Provides automatic start of the software when the mobile telephone is switched on

Setting	Explanation
Auto login	Activate this when the **CLIP Authentication** option is being used. Deactivate when the DTMF login procedure (password procedure) is to be used.
User call number	Internal call number of the FMC telephone (required for the DTMF login procedure)
PIN	User PIN for the DTMF login procedure (this corresponds to the user PIN of the user which is assigned to the FMC telephone)
PBX dialling in no.	Extension number of the **Phone No.** (E.123 notation with prefixed plus sign)
Voice mail no.	Internal call number for the OpenCom 100 communications system voicebox
MWI CLIP	Extension number of the **Phone No. (MWI Signaling)** (E.123 notation with prefixed plus sign)
Min. ext. number length	Determines how long a call number has to be for the route code to be automatically prefixed for external line seizure
Exchange access business	Route code for external line seizure
International area code	Area code replacing a prefixed plus sign (generally: “00”)
Extended DTMF	**Off:** *** for R function **On:** DTMF “A” for R function The DTMF “A” can be more quickly detected by the DTMF receiver
DTMF delay [ms]	Delay for DTMF tones (may be necessary for some mobile telephones)

Setting	Explanation
Hide settings menu	Prevents access to the **Settings** menu of the “Aastra Mobile Client” software on the mobile telephone
Time Control	Allows access to the **Time Control** function of the “Aastra Mobile Client” software on the mobile telephone

3. Upon conclusion of configuration, send multiple text messages (“SMS”) to the mobile telephone in the following order:

a) The **Send licence** button sends licence information via SMS to the mobile telephone.

b) The **Send configuration** button sends all configuration settings via SMS to the mobile telephone.

c) The **Send download link** button sends the mobile telephone an SMS with the internet address for downloading the “Aastra Mobile Client” software.

4. On the mobile telephone, open the SMS for the software installation. You can download the software from the internet via a mobile network connection. Depending on the device, it may be possible to (indirectly) download via WLAN or Bluetooth.

Please refer to the user guide of the mobile telephone for more details on downloading, installation of software and managing software packages.

5. Start the “Aastra Mobile Client” software on the mobile telephone. Upon starting, the software searches the list of SMS received and automatically saves the configuration and licence data. Both the configuration and licence SMS are automatically deleted after commissioning of the client.

6. You may configure a quick launch option for starting the “Aastra Mobile Client” software. Depending on the type of mobile telephone, you can configure a special key, a mode entry or a desktop link. Please refer to the mobile telephone user guide for details on this also.

Operating FMC functions via the software is described in the “Fixed Mobile Convergence – Using mobile telephones on the Aastra 800 / OpenCom 100 communications system” user guide.

Note: When you have configured the “Aastra Mobile Client” software for the DTMF log-in procedure (see page 184) and the FMC telephone user changes his/her PIN, you must change the configuration setting and re-send a configuration SMS to the mobile telephone. The you may have to re-start the “Aastra Mobile Client” software.

15. Team Functions

15.1 Introduction

With the team functions you can manage your telephone communication tasks by assigning lines with separate call numbers to the keys of different terminals. The terminal users, or team members, can thus pick up one another's calls or telephone each other using the configured keys.

Team functions can only be configured on the corded system telephones Aastra 677x (OpenPhone 7x) and their softphone variants because only these have the required features.

15.1.1 Explanation of Keys

The team functions are programmed on the call keys of the Aastra 677x (OpenPhone 7x) telephones. Depending on the terminal, different numbers of call keys are available:

Number of available call keys

Telephone	Number of keys
Aastra 6771 (OpenPhone 71)	One key with a display, five keys without a display
Aastra 6773 (OpenPhone 73) and Aastra 2773ip (OpenPhone 73 IPC)	Three keys with a display, five keys without a display
Aastra 6775 (OpenPhone 75) and Aastra 2775ip (OpenPhone 75 IPC)	Nine keys with a display
Aastra 6773ip (OpenPhone 73) and Aastra 2773ip (OpenPhone 73 IPC) or Aastra 6775 (OpenPhone 75) and Aastra 2775ip (OpenPhone 75 IPC) with an additional key extension Aastra M671	36 additional keys without a display Up to three of these key extensions can be used with an Aastra 6773 (OpenPhone 73) and Aastra 2773ip (OpenPhone 73 IPC)/ Aastra 6775 (OpenPhone 75) and Aastra 2775ip (OpenPhone 75 IPC).

Number of available call keys

Telephone	Number of keys
Aastra 6775 (OpenPhone 75) and Aastra 2775ip (OpenPhone 75 IPC) with an additional key extension Aastra M676	20 additional keys with a display Up to three of these key extensions can be used with an Aastra 6775 (OpenPhone 75) and Aastra 2775ip (OpenPhone 75 IPC).

Note: Only one function or call number can be programmed for each call key.

The following keys can be used:

- **Trunk key**: Calls (for the programmed call number, e.g. 11) are signalled to this key, and you can make internal and external calls via this number. A trunk key can be programmed with a substitute function (with another team member acting as the substitute). Calls for you are then signalled to the terminal of another team member. A trunk key also provides functions for managing calls. For example, you can configure call protection if you do not want to be disturbed, or call diversion to another telephone.

- **Team key**: As with a trunk key, a team key can be used to receive or make calls. However, this key cannot be used to change the settings for managing calls; it is not possible, for example, to configure call diversion to another telephone. Calls made via a team key are signalled to all terminals with a trunk key that has been programmed with the same number. For example, the team key with the number 11 calls all trunk keys with the number 11.

- **Busy key:** The purpose of a busy key is to make the busy status of other subscribers visible. An incoming call for a busy team member is signalled on the other team member's busy key. That team member can take this call by pressing the busy key, which seizes his own terminal's trunk key. Calls taken via the busy key are not entered in the call list of the team member who was originally called. In addition, it is possible to call the respective team member via his busy key when his terminal is idle. You set up a call to this team member by pressing your own trunk key.

- **Direct call key**: Only outgoing calls can be made with a direct call key; they are signalled to all terminals with the same number programmed to a trunk key. Calls via a direct call key are signalled to the destination terminal even if that terminal has been programmed with a substitution function or call protection.

If the destination terminal has been configured for call diversion, the direct call is not diverted.

Which Key is Suitable for Which Purpose?

- **Trunk keys** can be assigned call numbers for managing central communication tasks, for example, customer support. If the call numbers of the support department are assigned to trunk keys on all of its terminals, then all members of the support department can receive and manage calls and use the substitute function.

- **Team keys**, for example, can be used to create a project group within a department. Calls from customers of this group can then be answered by any team member who is not busy. The team members can call each other by the team keys.

- A **busy key** can be used to configure an enquiry station showing the status of the individual users. The enquiry station sees the status of the users and can put calls through by simply pressing the key.

- **Direct call keys**, for example, can be configured at a terminal in a conference room to call the secretary.

15.1.2 Team Configuration

You can create teams and programme call keys in the **Configurator** of the OpenCom 100 (**Telephony: Groups** and **Telephony: Ports: Upn** menu).
Call key 1 is preset as a trunk key on all system telephones. This setting can be changed by the system administrator.

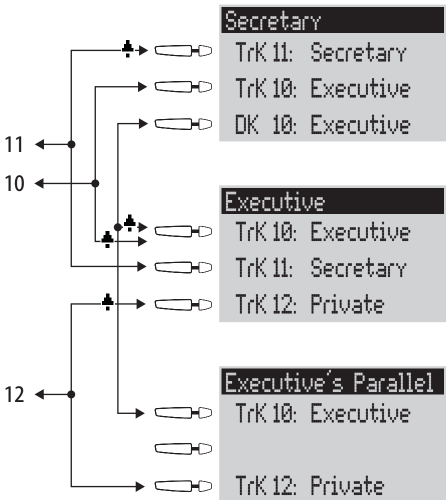
15.2 Examples of Use

The following examples illustrate the various uses of teams and team functions.

For information on the display texts and how to use the individual functions, refer to the chapter “Managing Calls in a Team” in the “Aastra 6771 / 6773 / 6775 (OpenPhone 7x)” user guide.

15.2.1 Executive/Secretary Team

In this example, the executive/secretary team comprises two members: the executive and the secretary. The secretary has one Aastra 6773 (OpenPhone 73) system telephone, and the executive has two, one of which is used as a parallel telephone in a sofa suite.



Example: executive/secretary team

Line Seizure
The secretary can be reached on the call number 11 (trunk key TrK 11: secretary's office).

The executive can be reached on the call number 10 (trunk key TrK 10: executive's office). He can also answer calls from his parallel telephone. In addition, a private line is configured for both of the executive's telephones (trunk key TrK 12: private).

Call numbers 11 and 10 are both configured as a trunk key on the executive's and the secretary's terminal respectively. Thus the executive and the secretary can use either call number (for answering as well as making calls). Each can act as a substitute for the other.

The secretary's terminal also has the executive's call number configured as a direct call number (DK 10: executive's office). The secretary can therefore reach the executive and put through calls even if the executive has programmed a substitute.

Line Busy Indication
If a line is busy, e.g. TrK 11 secretary's office, the other terminal will indicate this. The executive's private calls via TrK 12 are not indicated on the secretary's terminal since no appropriate trunk key is configured on the latter's telephone.

Call Signalling
In this configuration example, calls to one's own call number are signalled acoustically on the following telephones:

- Call number 11 on the secretary's telephone
- Call numbers 10 and 12 on the executive's telephone.

Calls for the other team member's call number are indicated by an optical signal on one's own telephone (flashing trunk key LED).

The parallel telephone will indicate calls only by an optical signal.

Time-delayed acoustic signalling can be configured for TrK 10 on the secretary's telephone. If the executive, for example, does not answer a call within 10 seconds, the secretary's telephone will start to ring.

If the executive activates a substitute function with the secretary as the substitute, calls for call number 10 will be indicated on the executive's telephone by an optical signal only, but signalled acoustically on the secretary's telephone. The secretary can also activate a substitute function. Calls for call number 11 are then signalled acoustically on the executive's telephone, and indicated by an optical signal on the parallel telephone and the secretary's telephone.

15.2.2 Three-member Team

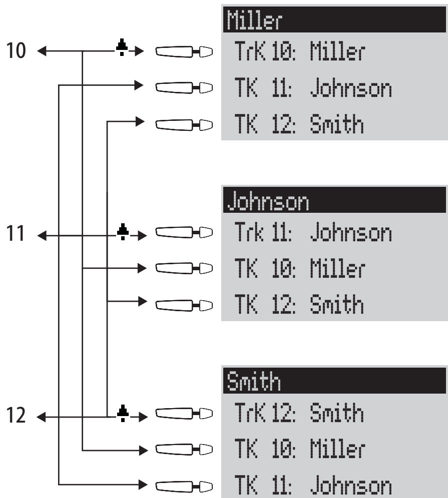
The three-member team described here is an example of a team configuration within a project group, e.g. export sales.

Each team member has one Aastra 6773 (OpenPhone 73) system telephone with all call keys programmed as trunk and team keys.

Line Seizure
Each team member's call number, e.g. call number 10 for Miller, is programmed as a trunk key on his telephone.

On the other telephones in the team, this call number is programmed as a team key (e.g. TK 10 on Johnson's and Smith's telephones). The team members can thus see which number a call is for and can answer it by pressing the appropriate team key.

The team members can call each other via the team keys. For example, Miller can call number 12 by pressing TK 12; the call is then signalled to Smith's telephone on TrK 12.



Example: three-member team

Line Busy Indication

If a line is busy, e.g. TrK 11 Johnson, the team keys 11 on Miller's and Smith's telephones will indicate this.

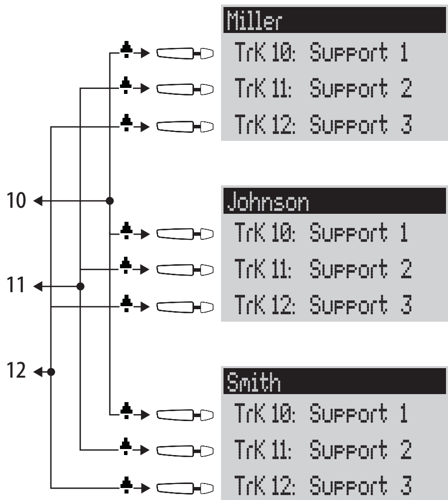
Call Signalling

In this example, calls via the trunk keys are signalled acoustically. Calls via the team keys are indicated by a visual signal (the team key LED flashes).

15.2.3 Unified Team

The unified team described here is an example of a team configuration within a department in which calls are to be managed quickly (e.g. support department).

Each team member has one Aastra 6773 (OpenPhone 73) system telephone with all call keys programmed as trunk keys.



Example: unified team

Line Seizure
Call numbers 10, 11 and 12 are programmed as trunk keys on each team member's telephone (TrK 10 to TrK 12.

All team members can use these numbers for answering as well as making calls.

Tip: In this team configuration it is useful to programme one of the function keys on each telephone with the "Hold" function. A call, e.g. for TrK 11, can then be put on hold by pressing the function key. If another team member then presses trunk key TrK 11 on his telephone, he can accept the call. For further information on function keys, refer to the "Aastra 6771 / 6773 / 6775 (OpenPhone 7x)" user guide.

Line Busy Indication
If a line is busy, e.g. TrK 11 Johnson, the trunk keys on the other team telephones will indicate this.

Call Signalling
In this example, calls via all trunk keys are signalled acoustically.

15.2.4 Toggle Team

The toggle team described here illustrates how a large number of call numbers can be managed efficiently with the help of team functions.

Each team member has one Aastra 6775 (OpenPhone 75) system telephone with all call keys programmed as trunk and team keys.

Line Seizure
Each team member is assigned seven call numbers, each programmed as a trunk key (TrK 10 to TrK 16 and TrK 20 to TrK 26). For each member, these trunk keys are programmed either as support numbers or hotline numbers.

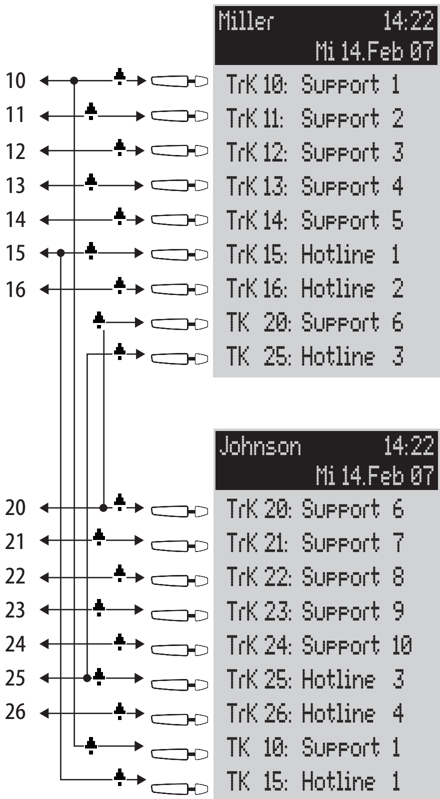
The first support number and the first hotline number of each team member is programmed as a team key on the other member’s telephone, e.g. TrK 10 and TrK 15 on Miller’s telephone as TK 10 and TK 15 on Johnson’s telephone. The assumption here is that most calls will go to the respective first call numbers, and team members can thus help each other out by answering one another’s calls.

On each telephone it is possible to toggle between the calls on individual lines, e.g. TrK 10 and TrK 11, by pressing the appropriate key (toggling).

Every call on a trunk key can be transferred to any other party by means of the R key. For more information, refer to the chapter entitled “Consultation, Toggling, Transfer and Conference” in the “Aastra 6771 / 6773 / 6775 (OpenPhone 7x)” user guide.

Line Busy Indication
If a line is busy, e.g. TrK 10 on Miller’s telephone, the appropriate team key will indicate this, e.g. TK 10 on Johnson’s telephone.

Call Signalling
In this example, calls via trunk keys are signalled acoustically. Calls via team keys are indicated by a visual signal (the team key LED flashes).



Example: toggle team

16. Call Queue

16.1 Introduction

A queue can be activated for the telephone numbers of any type of telephone, i.e. for system, analogue, ISDN and DECT telephones. If a call number with a queue is busy, calls to this number enter the queue. The caller first hears an announcement (if function “Announcer at busy” is configured) and then a dial tone.

Calls which remain in the queue for too long are cleared from the queue. The caller then gets a busy tone. If all the positions in the queue are taken then any further calls also hear the busy tone.

The time until an external call is cleared from a queue is defined by the network operator. In Germany this is usually two minutes and in other European countries usually three minutes.

If more than one telephone number (e.g. trunk or team keys) has been configured for a telephone, separate queues are used for each number.

On the Aastra 6775 (OpenPhone 75) or OpenPhone 65 system telephone, additional calls are signalled by a brief tone in the loudspeaker and in the display. If calls are in the queue, a number at the beginning of the second line of the display on the Aastra 6775 (OpenPhone 75 / OpenPhone 65 indicates how full the queue is. If more than one telephone number with a queue is configured on the telephone, the total number of entries are displayed.

Calls in a queue are handled by the OpenCom 100 in the following order of priority: instant connection, door calls, automatic recalls, VIP calls, then other internal and external calls. Sensor calls thus have priority over other calls, for example. Calls of the same priority level are switched in the order of their arrival.

The system administrator sets the number of calls that can be placed in a queue individually for each user group. The value can lie between “0” and “99”. The “0” value deactivates the ”Call queue” function for a user group. When the maximum number of calls in the queue is reached, further callers hear a busy tone.

Only calls which have a “voice” service indicator are administered in a queue.

Note: As calling fax machines often operate with the “voice” service indicator (e.g. on analogue ports), you should assign ports for fax machines on the OpenCom 100 to a user group **without** a queue.

Queues can be combined with the “call forwarding,” “pickup” and “hunt group” functions, for example, in order to configure an enquiry station for an operator.

16.1.1 Activation of Queues

Queues can be activated on a per user group basis. On delivery the default set, for all preset groups, is **off**.

When using queues, it often makes sense to activate call waiting protection. For this purpose, **Call waiting protection** authorisation must be allocated to the user group, and call waiting protection must be activated on the terminal.

Furthermore it is sensible to combine queues with the **Announcer at busy** function. When a caller calls a subscriber who is busy then they will hear a “central welcoming text”, for example, “Here is company XYZ. You will be immediately connected”. The function **Announcer at busy** can be set in the **Telephony: Call Distribution: Incoming** or **Telephony: Call Distribution: Incoming DDI** menu. Central welcoming texts can be recorded using the programme package **OpenVoice**.

You should configure a new user group (e.g. “Operators”) and activate the authorisations **Call queue**, **Call waiting protection** and, if necessary, **Call forwarding**. If users belong to this group, a queue will be activated automatically for all telephone numbers assigned to them.

16.1.2 Call Forwarding

Forwarded calls of the forwarding type “Immediately” and “On busy” have priority over queues. The queue of the forwarding telephone is not used for forwarding calls in this manner.

During the configuration of this type of call forwarding, the contents of the queue are **not** transferred to the target terminal. If there are still calls in the queue when the call forwarding function is activated, these calls can only be accepted on the source terminal.

If a call is to be forwarded "After delay", it enters the queue. If the call has not been answered before the delay period expires, it will be forwarded to the target terminal and can then be answered there.

16.1.3 Pickup

The functions "Pickup from group" and "Pickup selective" can be used together with queues. A user who accepts a call using "Pickup from group" or "Pickup selective" picks up the next call from the queue.

16.1.4 Hunt Groups

Hunt groups of the "parallel" type are usually used together with queues, with the queues of each telephone in the group being synchronised to each other. When a call to the number of the hunt group arrives, the call enters all parallel queues. If a call from one of the queues is answered, it is removed from all other parallel queues.

16.2 Examples of Use

Note: In the following examples it is assumed that a U_{pn} interface card has been installed (in an OpenCom 130 or an OpenCom 150). Ports on a U_{pn} interface card are DECT enabled, so that DECT base stations can be connected to operate cordless system terminals.

16.2.1 Enquiry Station for an Operator with Two System Telephones

The operator switches all incoming calls and can either work on the Aastra 6775 (OpenPhone 75 / OpenPhone 65 or the mobile terminal, the Aastra 610d / 620d / 630d.

Configuration

- Configure the system access or access for multiple terminals under **Telephony: Ports: S_0**.

- Configure the Aastra 6775 (OpenPhone 75 / OpenPhone 65 and e.g. a base station (RFP) under **Telephony: Devices**.
- Configure a trunk key for the Aastra 6775 (OpenPhone 75 / OpenPhone 65 under **Telephony: Devices: System telephones**.
- Configure the Aastra 610d / 620d / 630d under **Telephony: Devices: DECT phones** and assign the Aastra 610d / 620d / 630d its own telephone number. Check in the Aastra 610d / 620d / 630d.
- Under **Telephony: Call Distribution: Incoming** or **Telephony: Call Distribution: Incoming DDI** route all incoming calls to the number of the Aastra 6775 (OpenPhone 75 / OpenPhone 65 trunk key.
- In the **Configurator**, create a new group called "Operators" under **User Manager: User groups**. Activate the **Call queue**, **Call waiting protection** and **Call forwarding** authorisations for this group and set the **Dial in (outgoing): External** option appropriately.
- Create a user called "Operator 1" under **User Manager: User**. Assign this user to the "Operators" user group. Assign the telephone numbers of the Aastra 6775 (OpenPhone 75 / OpenPhone 65 trunk key and the number of the mobile Aastra 610d / 620d / 630d to this user.
- Activate `Call wait. prot.` (call waiting protection) on both terminals in the `Protection` menu.
- Configure a feature key on the Aastra 6775 (OpenPhone 75 / OpenPhone 65 which activates / deactivates a "call forwarding immediately" to the telephone number of the mobile Aastra 610d / 620d / 630d (in the menu `Call diversion: Divert phone: Immediately`).

Use

Incoming calls are routed to the Aastra 6775 (OpenPhone 75 / OpenPhone 65 manned by the operator, who then puts the calls through. A queue is used so that callers do not get a busy signal. The display on the Aastra 6775 (OpenPhone 75 / OpenPhone 65 indicates how many calls there are in the queue.

If the operator wants to leave the workstation and take along the enquiry station, call forwarding to the Aastra 610d / 620d / 630d is activated by pressing a feature key. Calls which are in the Aastra 6775 (OpenPhone 75 / OpenPhone 65 queue must still be answered on this telephone. New calls are signalled on the mobile

Aastra 610d / 620d / 630d or enter its queue, allowing the Aastra 610d / 620d / 630d to be used as a mobile enquiry station.

On returning to the workstation, the operator deactivates call forwarding by pressing a feature key. Calls which are already in the queue are switched on the mobile Aastra 610d / 620d / 630d. New calls are signalled on the Aastra 6775 (OpenPhone 75 / OpenPhone 65 or enter its queue.

16.2.2 Group of Three Enquiry Stations

The enquiry stations switch all incoming calls. Incoming calls are administered in queues. Depending on the number of arriving calls, one to three enquiry stations in this group are manned. The enquiry stations are each equipped with an Aastra 6775 (OpenPhone 75 / OpenPhone 65.

Configuration

- Configure the multi-terminal access or the system access under **Telephony: Ports: S_0**.
- Configure the three Aastra 6775 (OpenPhone 75 / OpenPhone 65 telephones under **Telephony: Devices**.
- Configure a trunk key with its own telephone number for each of the Aastra 6775 (OpenPhone 75 / OpenPhone 65 telephones under **Telephony: Devices: System telephones**.
- Configure a hunt group of the **parallel** type under **Telephony: Groups: Hunt Group**, and include the three telephone numbers of the trunk keys in this hunt group.
- Under **Telephony: Call Distribution: Incoming** or **Telephony: Call Distribution: Incoming DDI** route all incoming calls to the number of the hunt group.
- In the **Configurator**, create a new group called “Operators” under **User Manager: User groups**. Activate the **Call queue** and **Call waiting protection** authorisations for this group.

- In the **User Manager**, configure a user for each of the three operators and assign these settings to the user group called “Operators”. Allocate each user the telephone number of the trunk key of their system telephone.

- Activate `Call wait. prot.` (call waiting protection) on all three terminals in the `Protection` menu.

- Programme a feature key with the function “Sign on / sign off from hunt group” on the three system telephones (in the menu `Calls: Hunt group`).

Use
Incoming calls are signalled in parallel to all signed-on enquiry stations. If the enquiry stations are busy, the incoming call joins the queue on each of the terminals in the hunt group. If one of the enquiry stations accepts a call from the queue, the call is removed from the queues of all the other enquiry stations. The display at each enquiry station (Aastra 6775 (OpenPhone 75 / OpenPhone 65) indicates how full the queue is.

If attendants leave the station, they sign off from the hunt group by means of a feature key. In contrast to Example 1, further calls do not have to be processed after the sign-off, as the calls are also registered in the queues of the other signed-on enquiry stations.

Note: When the last enquiry station remaining in the hunt group signs off, further callers will hear the busy tone.

17. Multi-Company Variant

Communications systems are frequently shared by several companies. These companies want to jointly use the existing infrastructure (e.g. the existing lines and features of the system), while at the same time they wish to organise and pay for their communication completely independently of one another.

This “multi-company variant” can be implemented using the OpenCom 100 within a shared office, for example.

In the multi-company variant, the companies are essentially completely independent of one another. This allows them to have their own trunk lines, which is useful for billing purposes. The OpenCom 100 hardware and software are used equally by all the companies, however. It is possible to configure the OpenCom 100 for each company and define the extent to which the features of the system may be used.

In brief, the features of the multi-company variant are as follows:

- Up to five companies can be configured at the same time.
- Every user of the OpenCom 100 is assigned to a company.
- Each available bundle (trunk group) or SIP trunk is uniquely assigned to a company so that incoming external calls can be transferred to the correct internal subscriber.
- For each company, every route can have its own code. For example, it is possible to activate different routes with the code “0” for different companies. This enables separate charging for outgoing external calls, for example.
- An individual exchange (“operator”) can be set up for each company.
- Each company can maintain the communication data of its business partners in its own company telephone book.
- The charges can be billed individually for each company.

17.1 Configuring the Multi-Company Variant

The multi-company variant can be commissioned and configured by the system administrator of the OpenCom 100 without any major effort. In the multi-company variant, the communications system behaves in exactly the same way as the single-company variant. This is particularly of interest to users who want to expand their own system and at the same time operate it in a group.

The process in brief:

1. The feature must be activated (see *Activating the Multi-Company Variant* starting on page 203).

2. The required companies must be set up (see *Configuring and Managing Companies* starting on page 204).

3. The users of the OpenCom 100 are assigned to the individual companies (see *Assigning Users* starting on page 204).

4. In order that the OpenCom 100 can transfer incoming calls to the corresponding company (or its staff) correctly, the existing bundles must be uniquely assigned to the companies (see *Assigning a Bundle/SIP Trunk* starting on page 205).

5. In the case of outgoing external calls, the lines via which the members of a company can make a call must be defined (see *Allocating Routing Codes* starting on page 205).

6. An exchange must be set up for each company so that the OpenCom 100 can correctly process statuses in which a call should be routed to the exchange (see *Configuring the Company Exchange* starting on page 206).

17.1.1 Activating the Multi-Company Variant

To be able to configure several companies in the OpenCom 100, the “Multi-company variant” (OpenCompany 45) programme package must first be activated. This is done in the **Configurator** on the Web console in the **System: Licences** menu. The licence confirmation you received with the programme package contains all the information you require about how to proceed.

Only when this package has been activated are the fields required to configure the multi-company variant available in the other menus of the Web console, for example in the **User Manager: User groups** menu or in the **Telephony: Trunks** menu.

17.1.2 Configuring and Managing Companies

Up to five companies can be configured in the OpenCom 100. By default, one company with the name "Company 1" is predefined. All configuration settings, e.g. in the user groups or in the bundle configuration, apply to this predefined default company if not other company has been selected.

Companies are set up and managed in the **Telephony: Extended: Companies** menu:

- A new company is created in this menu using the command **New**. Each company can be given a name up to 20 characters long. This name is then displayed in all configuration dialogue boxes in which company-specific settings can be defined.

- In this menu a company can be deleted again using the command **Delete**. If a company is deleted which is still used at other places (in the user groups, for example), the respective configuration is changed to the default company.

- The name of the default company can be changed, but the default company itself cannot be deleted.

17.1.3 Assigning Users

For each user you must define the company to which they belong. This assignment determines, for example, which company telephone book the user has access to and which company-specific configuration data apply to them.

As the OpenCom 100 manages users in groups, the assignment "user > company" is also established this way. The company to which each user group belongs must be defined for each group. A user group can only belong to one company, i.e. not to several. However, a company can have several user groups. It is therefore possible, in the same way as in the entire system, to allocate a range of authorisation rights for the use and configuration of features for each company.

When setting up a new **User group** (in the **User Manager** menu), you will find that the default company is predefined; another company can be assigned as long as no other companies have been set up.

17.1.4 Assigning a Bundle/SIP Trunk

Connections of the same type and in the same direction are arranged in a bundle (e.g. S_0 multi-terminal connections). To be able to correctly transfer incoming calls to the members of the configured companies (the users) via the lines of a certain bundle of the OpenCom 100, each of the available bundles must be assigned to one of the companies. This is necessary to be able to transfer incoming external calls to the correct company exchange in cases where the called internal subscriber cannot be reached (“Connection to Operator”), for example. SIP trunks can also be assigned to a company.

The assignment of bundles to companies is done in the **Telephony: Trunks: Bundle** menu. The assignment of SIP accounts to companies is done in the **Telephony: Trunks: SIP trunks** menu.

For outgoing external calls which users set up via the lines of their company’s bundle/SIP trunk, the assignment of the bundle to the company is irrelevant: the charges are assigned according to the “source” principle.

Charges are billed to the company to which the user belongs who set up the connection. The OpenCom 100 recognises this on the basis of the assignment between user groups and companies and on the basis of the routing code with which a line of the bundle/SIP trunk was seized. For more information, please see the following section.

17.1.5 Allocating Routing Codes

Routes are used for automatic and selective seizure of bundles or connections for external calls. It is possible to seize a route by predialling a code.

In the **Telephony: Trunks: Route** menu, you can define which company can seize each route. An individual **code** for the seizure is allocated per route for each company. The OpenCom 100 ensures that during configuration no seizure code is allocated twice (for two different routes) for each company. If during configuration of a route no code is allocated for one of the configured companies, the route concerned cannot be seized by the members (user groups) of this company.

17.1.6 Configuring the Company Exchange

An internal telephone number must be set up for each company which represents the exchange, i.e. “the operator”. The calls to specific extensions arriving at the exchange are routed to this number, for example, as are all external calls where the called subscriber (a user who belongs to this company) cannot be reached, as in the case of a timeout.

A company exchange is set up in the **Telephony: Operator** menu. In this menu, you can specify an internal telephone number for each company and time group which then represents the exchange for this company.

17.2 Working with the Multi-Company Variant

All the features of the OpenCom 100 which the users may already be familiar with from the single-company variant are available in the multi-company variant. These features can be used to the same extent and can be used in exactly the same way.

The following section describes the features additionally available to the users in the multi-company variant.

17.2.1 Company Telephone Book

An individual company telephone book can be created for each company. In addition to this, “personal” and “central” telephone books exist:

- A personal telephone book is available for each user.
- The central telephone book can be used across the companies by all users of the OpenCom 100.

The company telephone book is a central telephone book for the whole company. It is only available to the users/user groups who are assigned to this company. You can also define whether the members of each user group may edit the company telephone book or not.

The company telephone book is treated exactly the same way on the system terminals as the other types of telephone books. This means that the entries listed in the personal, central and company telephone books are displayed on the system phones at the same time.

Users can also use the telephone book of their company with the **OpenCTI 50** and **Phone Book** Web applications, assuming they are authorised to use these applications.

In addition, it is also possible to assign a user group with the authorisation to edit foreign company telephone books. This authorisation is useful if members of this group - e.g. the “Administrators” - service the entire system. Foreign telephone books can only be edited in the **Configurator** in the **Phone Book** menu.

The number of entries in a company telephone book is unrestricted. The OpenCom 100 can manage up to 2,000 entries in *all* telephone books (in the central, personal and company telephone books).

17.2.2 Making Calls Between Companies

All users of the OpenCom 100 can make internal calls to one another, irrespective of which company they belong to. Calls between users from the different companies are therefore not subject to any restrictions.

17.2.3 Billing Charges per Company

In the **Costs** Web application you can output the charges for each company. Users who are authorised to use this application can view the charges for each company.

18. Configuring the PC Software

Further possibilities of use can be implemented on a workstation computer with the Windows operating system by installing drivers and programmes. You can find the installation programmes required for this on the product CD that comes with the OpenCom 100.

Proceed as follows to install extra software:

1. Log on under Windows as the administrator.

2. Insert the product CD.

   If your PC is suitably configured, the CD will start automatically. Otherwise select **Run** from the Start menu. Click on the **Browse** button to look for the programme “cd_start.exe” on the CD. Confirm this with **Open** and **OK**.

3. Select the required option from the start interface. Follow the programme instructions.

Further instructions for various options that are available are given below.

18.1 PC Offline Configuration

The offline configurator is a reduced system software. With the offline configurator it is possible to create system configurations for various TC-system types – without a connection to the live system. You configure a virtual system and later, aided by the data backup, transfer the configuration to the running system.

Installing the Offline Configurator

1. Display the product CD contents in Windows Explorer. Search the “\OFC\” installation directory for the offline configurator.

2. Start the **StartCenter** installation programme by double-clicking on “Setup.exe”. Following the installation assistant’s instructions. Select the appropriate installation directory or apply the default.

3. On the product CD, in the offline configurator installation directory, you will find ZIP archives for various TC-system types. Copy the desired ZIP archives to the installation directory.

4. End the installation with a function test. With a double-click on the newly created desktop icon, start the **StartCenter** programme. Select from the fold-down selection the desired TC-system type. Available offline configurators are now displayed as symbols. Double-click on a configurator symbol and select from the dialogue **Start with Factory settings**. The offline configurator starts with an command window. Open a Web browser and in the address line enter: "http://localhost/". Confirm with enter.

5. Configure the virtual system as usual. Using the data backup, you can save the configuration.

6. Activate the command window. End the offline configurator with the window's menu command **Close**.

To maintain multiple different installations, you can copy further offline configurators to new directories of your hard-drive. Change between the directories by selecting the command **Extras**: **Setup** in the **StartCenter** programme. Select the corresponding **Working Directory** in the **Setup** dialogue. You will find further explanations in the **StartCenter** online help.

Notes

- The ITC-system serial number to which the configuration is later to be transferred is queried when starting the offline configurator. You will recognise this by the caution symbol in the left symbol-bar. Click on this symbol. Enter the serial number into the input field that opens.
  The serial number is required to activate programme packages requiring licences in the offline configurator (in menu **System: Licences**). The serial number can be obtained for the ITC-system's **Configurator** in the **System Info: Versions** menu.

- If there is a firewall installed on the workstation computer, then a warning will be displayed. This warning shows that the computer is now ready-to-receiver requests for the offline configurator's web-server service. You must permit the web-server service, for example by activating the option **For this program do not display this notification again**.

- If there is an existing web-server service installed on the workstation computer, then you must enter a new port number in the **Setup** dialogue of the programme **StartCenter**, for example 8080. In the Web browser, the address line input is then: “http://localhost:8080/”. When using different port numbers you can also start multiple offline configurators simultaneously on one workstation computer.

- If for the Web browser there is a proxy activated, then you might have to enter the computer name “localhost” into the exceptions list (**No proxy for …**).

- During deinstallation of the programme **StartCenter**, the entire installation directory with all its sub-directories is deleted. Backup the necessary offline configurators before uninstalling.

18.2 Setting up TAPI

With a TAPI (**T**elephony **A**pplication **P**rogramming **I**nterface) you can operate a CTI application (**C**omputer **T**elephony **I**ntegration). Here, the CTI application uses the services of the OpenCom 100 with the help of the TAPI driver installed on a Windows PC.

Many telephony functions, such as enquiry, toggling, three-party conference, pick-up, call protection and call forwarding can be controlled using appropriate TAPI-compatible software.

Requirements
You require an active IP network connection between the PC and the communications system. CTI functions can be used only in conjunction with system terminals.

You must therefore have configured at least one user for a system terminal. In addition, you require a TAPI 2.1-compatible CTI application, for example the **Phone Dialer** included in the Windows operating system.

Installing the TAPI Driver

1. Call up the start mask from the product CD (see *Configuring the PC Software* on page 208).

2. Select **Software**: **Install TAPI Service Provider** from the start mask.

3. Follow the programme instructions.

Configuring the TAPI Connection

Note: Under Windows you should log on as the user for whom you want to configure the TAPI connection.

1. In the Start menu, select **Settings**: **Control Panel**. Select the **Printers and other Hardware** (with Vista: **Hardware and Sound**) category. Double-click on the **Phone and Modem Options** icon.

2. Change to the **Advanced** tab.

3. From the list of installed driver software, select **Aastra 800/OCX/OpenCom Service Provider** and click on **Configure**.

4. In the following dialogue you will find a list with the configured connections for the user who is currently logged on. Click on **New**.

5. In the following dialogue you provide information for the new connection. In the **Connection name** box you can enter a descriptive name for the connection. In the **CTI server** box you must enter the DNS name or the IP address of the OpenCom 100. Using the **[…]** button you can search for this in the LAN. In the boxes **Username** and **Password** you enter the user data of one of the users configured on the OpenCom 100. This user must be allocated a system terminal. Confirm your entry with **OK**.

6. The new connection is now configured. Close the opened dialogues with **OK** and **Close**.

Testing the TAPI Function

1. In the Start menu, select **Programs**: **Accessories**: **Communication** and then start the programme called **Phone Dialer**.

   Under Windows XP and Vista, the **Phone Dialer** is started indirect by using the dialling function of the **Address book** (can be found in the start menu under **Programs**: **Accessories**). A manual start of the programme file “Dialer.exe” in the “C:\Program Files\Windows NT” (XP) or “C:\WIndows\System32” (Vista) folder is possible also.

2. In the **Tools** menu, select the item **Connect using...** to select the system terminal that is to use the CTI application. Under Windows XP you select the item

Options from the **Edit** menu. In the **Lines** tab you then select the system terminal from the **Phone calls** list.

3. Enter a telephone number in the **Number** box and confirm with **Dial**. Under Windows XP you first click on the **Dial** icon and in the subsequent dialogue activate **Phone call**.

4. The number you entered is displayed on the selected system terminal. Lift the receiver to start dialling.

Note: This note is not relevant to Windows XP or Vista. If the "Phone Dialer" programme is not installed, you will have to install it. To do this, you open the **Control Panel** and click on **Software**. In the **Windows Setup** tab you activate the **Connections** component.

18.3 Setting up NET CAPI

With a CAPI driver (**C**ommon **A**pplication **P**rogramming **I**nterface) Windows programmes are able to access services and functions of an ISDN card. With a network-based CAPI, the OpenCom 100 allows the use of ISDN functions also by workstation computers in which no ISDN card is integrated.

Requirements
You require an active IP network connection between the PC and the telephone system.

Please note: Before installing the CAPI driver for the OpenCom 100, any existent ISDN card must be removed and any CAPI drivers on your PC must be de-installed.

Installing the NET CAPI Driver

1. Call up the start mask from the product CD (see *Configuring the PC Software* on page 208).

2. Select **Software**: **NET CAPI Driver** from the start mask and follow the programme instructions.

Configuring the NET CAPI Driver

The NET CAPI driver requires an extra internal number so that the “virtual ISDN card” on the OpenCom 100 can be addressed:

1. Go to the **Configurator, Telephony**: **Extended**: **CAPI-ISDN** menu. Click on **Change**.

2. Activate the **Status** check box. Enter at least one unassigned, internal number in the boxes under **Parameters**. Confirm your entry with **Apply**.

3. Go to the **Configurator, User Manager: User** menu. Select one of the users shown. Enter the number just assigned in one of the boxes **No. 1** to **No. 10**. Confirm your entry with **Apply**.

4. If it is to be possible to call the “virtual ISDN card” externally, or if external calls are to be possible, the number must be included in call distribution (**Configurator**, **Telephony: Call Distribution** menu).

5. After installing the NET CAPI driver, you will find an extra icon on the right side of the Windows Start bar. Click on this icon with the right mouse key. Select the **Log-on** command from the menu.

Note: In the subsequent dialogue you must log on NET CAPI first with the user (user name and password) for which you configured the CAPI telephone number in the **User Manager** (see Step 3).

You will find further information on the working of the NET CAPI driver and CAPI application programmes on the product CD.

Note on Sending Faxes

The NET CAPI can not address an analogue Group-3 fax. Use a CAPI-compatible modem-simulation driver or connect an analogue modem or analogue modem card to one of the OpenCom 100’s internal a/b ports for sending faxes.

18.4 Browser for OpenCTI and OpenHotel

You can simplify the daily use of the **OpenCTI** and **OpenHotel** Web applications using the Web browser especially adapted for the OpenCom 100. Each time the workstation computer is restarted, this browser programme can automatically start and log you in. This means that the applications are always operational and can be accessed using the icon in the information area of the task bar.

Installing the Browser

1. Call up the start mask from the product CD (see *Configuring the PC Software* on page 208).

2. Select **Software**: **Install Browser for OpenCTI** or **Software**: **Install Browser for OpenCTI** from the start mask.

3. Follow the programme instructions.

After installing the browser, there is a new menu item in the Windows start menu under **Programs**: **OpenCTI Browser** respectively **Programs**: **Hotel Starter**.

Further information can be found in the online help of the browser programme. To view this, click the top left corner in the **OpenCTI**-browser's programme window on the system menu symbol or on the symbol in the information area of the task bar. Select the **Readme** command. You will find the **OpenHotel**'s readme in the installation directory of this browser programme.

Note: Both browser programmes can be used simultaneously.

18.5 Setting up Video Telephony

You can use the **OpenCTI** to switch on the video function during an internal call. To be able do this, the Microsoft NetMeeting 3.0 programme must be installed and set up on all participating workstation computers.

Note: NetMeeting is already pre-installed on the Microsoft Windows XP operating systems.

Setting up Microsoft NetMeeting 3.0

1. Connect a standard web cam to the workplace computer and install the driver.

2. In the Windows **Start** menu select **Run** and enter: “conf.exe”. Confirm your selection with **OK**.

3. Follow the instructions of the Install Wizard. Registration in an Internet directory is not necessary and is not recommended. Select the installed web cam and exit the Install Wizard.

4. Run a functionality test. To do this, start the **NetMeeting** programme. Click on the call button. Under **Address** enter the IP address or DNS host name of a external station. Confirm your selection with **Call**.

Note: If Firewall software is installed on the workstation computer, a warning will now appear. This shows that the computer is now ready to receive the NetMeeting. You must allow the NetMeeting, by activating the **Do not show this message for this program again** option for example.

18.6 Synchronising the PC Clock

With the network service SNTP (simple network time protocol) it is possible to synchronise the internal clock of a workstation computer with the time of the OpenCom 100.

Requirements
You must enter the time zone so that the OpenCom 100 can calculate the time of the internal clock back to the GMT (Greenwich Mean Time) required for SNTP:

1. Go to the **Configurator, System: Common** menu. Click on **Change**.

2. Enter the **Time zone** for which the time of the OpenCom 100 applies and whether summer time is allowed for. Confirm this with **Apply**.

Configuring SNTP
For various operating systems, you can use one of the numerous SNTP programmes offered for downloading on the Internet. Configure the OpenCom 100 as an SNTP server for such programmes.

Please note: In a Windows domain network, the PDC server (primary domain controller) should automatically assume the function of the timer.

SNTP with Windows XP
Here you configure the SNTP server by double-clicking on the time in the Start bar. Enter the OpenCom 100 as the **Server** in the **Internet time** tab.

18.7 Address Queries using LDAP

You can search the data of the central telephone book of the OpenCom 100 from a workstation computer in the LAN using LDAP (Lightweight Directory Access Protocol). When configuring an LDAP-enabled programme, specify the IP address of the OpenCom 100 as the address of the LDAP server.

LDAP with Outlook Express

You can configure and operate the LDAP directory service with Outlook Express™, a Microsoft™ e-mail programme, as follows:

1. Call up the **Accounts** command in the **Tools** menu.

   The **Internet Accounts** dialogue box will then open.

2. Click on **Add**. Select the **Directory Service** command from the pop-up menu.

   The **Internet Connection Wizard** dialogue box for Internet access will then open.

3. Under **Internet directory (LDAP) server, enter the address of the** OpenCom 100. It is not necessary to log in to the **LDAP server**. Click twice on **Next**. Then click on **Finish**.

4. Check the function. In the **Edit** menu, call up the **Find**: **People** command.

   The **Find**: **People** dialogue box will then open.

5. In the **Look in** list, select the entry with the OpenCom 100 address. Enter a user in the **Name** input field, Administrator for example. Then click on Find now.

   The list of entries found should now display the address from the central telephone book.

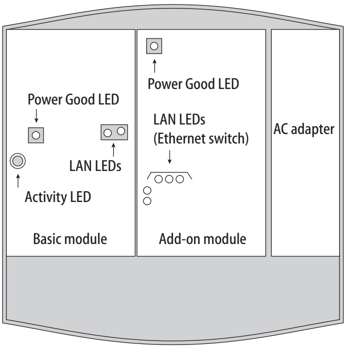
Note: Only users can be found for whom an internal telephone number has been configured.

19. Frequently Asked Questions

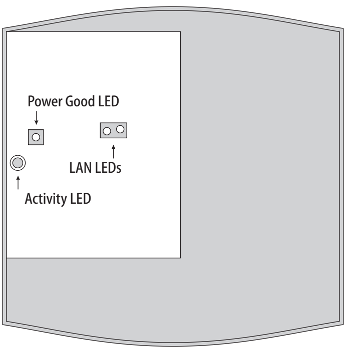
This chapter provides tips and information on how to deal with any malfunctions or faults you may experience with the OpenCom 100.

Please note: Repairs to the OpenCom 100 should only be carried out by qualified personnel.

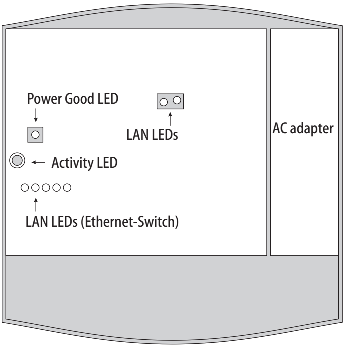
The following LEDs indicate that the OpenCom 100 is ready for operation:



Position of LEDs on the OpenCom 130



Position of LEDs on the OpenCom 131



Position of LEDs on the OpenCom 150

With the OpenCom 150 Rack, the LEDs for LAN0 and LAN1 are placed on the front panel.

19.1 General/Hardware

Question: The OpenCom 100 is not functioning.
Check whether the plug-in power supply is properly connected (OpenCom 130). If an add-on module is installed: Make sure the mains plug is properly connected.

Plug another device into the mains socket to check whether there is any voltage.

Question: The mains plug is connected, the mains socket is supplying output, but the OpenCom 100 still does not function.

DANGER!	**This device contains hazardous voltages. To make the system dead, remove the power plug and the plug-in power supply from the socket.**

Take the housing cover off. Is the “Power Fine” LED illuminated?

If not, contact your service centre or an authorised dealer. The AC adapter plug of the OpenCom 100 may be defective.

Question: After restarting the OpenCom 100, nothing is indicated on the displays of any connected terminals.
It takes a short while for the OpenCom 100 to start up.

After the restart, check whether the activity LED flashes at a rate of 10s / 1s. This flash cycle indicates that the OpenCom 100 has started up correctly and is ready for operation.

If the OpenCom 100 has not restarted properly, reset the OpenCom 100 to its original factory setting (refer to the chapter entitled *Resetting the System Data* starting on page 87).

19.2 Telephony

Question: It is not possible to make external calls.
Check the connection between the NTBA and the OpenCom 100.

In the **Configurator**, check whether the external S_0 ports are configured correctly (**Telephony: Ports: S_0** menu):

– Configuration of System- / Multi-terminal access OK?
– Port is connected to the NTBA?
– Faultless Cabling?
– Terminating resistors properly configured?

Check the status of the trunks. In the Configurator, open the **System Info**: **Telephony**: **Trunks** page. For the **Trunks** used for the “External trunk” route the **Status** indicator should display a small green hook symbol.

Question: The OpenCom 100 is connected to an NTBA with a multi-terminal configuration. Why is it not possible to establish external connections?
With the original factory setting, an additional external S_0 port is set for an NTBA in the communications system configuration; this additional port will be used first to seize a trunk line.

Deactivate the corresponding S_0 port in the **Configurator** (**Telephony: Ports: S_0** menu).

Question: One of the telephones is not functioning at all.
Make sure the telephone has been properly connected.

Check also whether the appropriate port has been configured correctly in the **Configurator** (**Telephony: Ports** menu).

Question: It is not possible to make external calls with one of the telephones.
Check whether a user is configured for the telephone. Otherwise the settings of the Guests user group are valid for the telephone. To standard, this user group has no external call authorisation.

Make sure the user configured for this telephone belongs to a user group with external line access (**Configurator**, **User Manager: User groups** menu).

Check also whether the internal call number of this telephone has been configured for incoming call distribution (**Configurator**, **Telephony: Call Distribution** menu).

Question: One of the features (e.g. call diversion) on one of the telephones cannot be used even though the feature has been configured in the Configurator of the OpenCom 100.
Make sure the user configured for this telephone belongs to a user group that has access to this feature (**Configurator**, **User Manager: User** and **User groups** menus). Some features cannot be used until the system PIN is changed.

Question: Nothing is indicated on the display of one of the connected ISDN telephones.
You have connected the ISDN telephone to an external S_0 port (RJ-45 socket). These ports are intended for connection to the NTBA only. Connect the telephone to an internal S_0 port (pressure terminal).

Question: Calls can be made but not received with one of the ISDN telephones.
The internal call number that has been configured for this ISDN telephone in the **Configurator** (**Telephony: Ports: S0** menu) must also be configured as an MSN on the ISDN telephone itself. For further information, refer to the User Guide of your ISDN telephone.

Question: An ISDN telephone always rings, if another telephone on the S_0 bus is being called.
This case also requires configuring the MSN on the ISDN telephone (see above answer).

Question: It is not possible to configure Call Distribution: Outgoing for multi-terminal access.
You have configured multi-terminal access and system access in parallel. All outgoing calls are therefore established via system access, and outgoing call distri-

bution can be configured for system access only in the **Configurator** (**Telephony: Call Distribution** menu).

A specific MSN can be seized for individual calls by means of a code number procedure. For further information, refer to the “OpenCom 100 / Aastra 800, Operation on Standard Terminals” user guide.

Question: What are some of the causes for problems when sending and/or receiving faxes?
In frequent cases, the reason may be found in a problem with the ISDN-L1 reference clock distribution. The L1 clock is delivered from the network provider. An unclean L1 clock distribution and the introduced signalling jitter is overheard by the human ear. Nevertheless, data and fax transmissions may be disturbed by the jitter. Please check, which ISDN lines will deliver the L1 clock. Details can be found under *L1 Clock* starting on page 155.

The fax data transfer possibly is routed via a compressing VoIP connection. Please verify, if the a/b port is configured with the “Fax” setting. For the VoIP connection, select a VoIP profile which includes the non-compressing G.711 codec.

19.3 PBX Networking

Question: Why is it not possible to answer calls from another communications system via the missed calls list?
You get a call e.g. via a direct Q.SIG connection. Which line seizure codes you will need to dial for the callback is not included automatically in the information transferred with Q.SIG. Therefore, you need to setup the callback in the bundle configuration. Call up the **Telephony**: **Trunks**: **Bundle** page. Change the desired Q.SIG bundle and enter the line seizure code for the callback in the **Prefix for source phone number at incoming internal** input field.

Question: Why is it not possible to reach a communications system indirectly via another communications system?
You have interconnected e.g. three communications system with two direct Q.SIG connections. While configuring the routes, you applied the default “Business” selection for the **Type** setting. Call-switching for internal calls is possible for the in-between communications system for internal routes only. Change the Type setting for all affected routes to “Internal”.

19.4 DECT

Note: In the following explanations it is assumed that a U_{pn} interface card has been installed (in an OpenCom 130 or an OpenCom 150).

Question: The LED of the RFP 22 / 24 base station is flashing, but none of the DECT devices is functioning.
Make sure the terminal setting for the corresponding U_{pn} port is set to an RFP 22 / 24 base station (**Configurator**, **Telephony: Ports: U_{pn}** menu).

If multiple RFP 22 / 24 base stations are installed, the blinking LED indicates that synchronisation is not finished.

Question: The LED of the RFP 22 / 24 base station is continuously lit up, but one of the cordless DECT devices is indicating "No connection".
You have not registered this DECT device. Configure a port in the **Configurator** and start the enrolment procedure (**Telephony: Devices: DECT Phones** menu).

Question: Is it possible to increase the time for the enrolment procedure?
You must manually enter the IPEI of the DECT device in the **Configurator**. The enrolment time is then increased to one hour (**Telephony: Devices: DECT Phones** menu).

Question: Another manufacturer's DECT device is not functioning.
Check whether the DECT device supports the DECT GAP standard. In the **Configurator**, also make sure **GAP** is set for this DECT device (**Telephony: Devices: DECT Phones** menu).

Question: The startup procedure of the DECT base stations take a long time? What is the reason?
This behaviour may indicate a problem with the reference clock. Refer also to *What are some of the causes for problems when sending and/or receiving faxes?* starting on page 221.

19.5 LAN

Question: Why is it not possible to establish a network connection with the OpenCom 100?
Check whether the LEDs for the switch and the PC's network card are indicating a connection.

Check the LEDs for the LAN functions of the OpenCom 100. The green LAN LED at the top indicates whether the network cable has been properly connected. The red LAN LED indicates whether there is any network traffic on the line.

If you have installed an add-on module (in an OpenCom 130), also check the LAN LEDs of the Ethernet switch. The Ethernet line between the basic module (LAN port) and the add-on module (LAN1 port) are properly connected if the centre LED lights up. The Ethernet line between the hub of the corporate network and the LAN0 port is properly connected if the right-hand LED lights up.

To check whether there is a network connection with your OpenCom 100, enter the “ping IP address” command in “Run” in the Windows Start menu (e.g. ping 192.168.99.254).

Question: How can I determine the IP address of the OpenCom 100?
To find out what the IP address is, enter the code number *182 on one of the connected system telephones.

The code-number procedure *183 also displays the network mask.

Question: The network connection is functioning, but nothing is displayed in the browser.
Enter the complete IP address of the OpenCom 100 along with the protocol identifier, for example http://192.168.99.254/.

Check whether the browser has been configured for connection through a proxy server. If so, deactivate the “Connect through proxy server” setting.

Question: You have just configured the OpenCom 100 via the network. Why is it not possible now to establish a remote data transfer network connection?
The network card and the communication (remote data transfer) adapter cannot be run with the same routing setting. Deactivate the network card before connecting via the dial-up network.

Question: Our network has grown over time, with several segments connected by one central router. How can PCs from all segments connect to the OpenCom 100?
If several routers are configured for your network in different segments, you can enter extra static routes in the **Network: Routes** menu.

Question: In our network the OpenCom 100 dynamically issues the IP addresses by DHCP. Can I firmly assign the IP address for our internal server PCs (mail, Web)?
You need a static address assignment for these PCs. Make the appropriate host assignment entries in the Configurator (**Network**: **Hosts** menu). Create a static DHCP entry for each host assignment in the **Network**: **DHCP** menu. Activate "Dynamic and static address" for the DHCP server.

19.6 Internet

Question: I cannot access our company Web site.
Outside your system, your company Web site is accessed at "www.firm.com", but in the **Configurator** you have entered "firm.com" as the domain. Your company's site URL thus counts as an internal URL and can only be accessed by entering the direct IP address. If required, change the domain setting in the **Network**: **LAN** menu.

Question: Why do some Internet services not work even though they can be used when dialling in directly via a modem?
Some Internet services require an active connection coming from the Internet. But the configured filter rules prevent this. Plus, it is not possible to establish incoming Internet connections with the PCs directly owing to the network address translation process.

It is possible to redirect incoming connections in the **Configurator**, menu **Network**: **Port Forwarding**. You should secure the redirection target (PC or server) with a suitable firewall software.

Question: A SIP connection only passes unidirectional voice. What is the reason?
You did not use the OpenCom 100 as internet router or the STUN server of the SIP provider is unavailable. You need to activate the SIP support at your internet router, such as "SIP-ALG" or "Full Cone NAT" functions. You can also use the OpenCom 100 for internet access. Correct the STUN setting in the **Telephony**: **Trunks**: **SIP Provider** menu.

Question: Is it possible to use Q.SIG-IP connections via an Internet access with dynamic IP address?

Q.SIG-IP connections require a fixed IP assignment for technical and security reasons. Therefore you need an Internet access with a fixed IP address. It is possible to tunnel a Q.SIG-IP connection through a VPN connection. A VPN connection offers the possibility to determine the peer's IP address with a DynDNS service during connection setup. VPN and DynDNS can be realized with external servers or routers.

20. Technical Specifications

OpenCom 130

System data	
Mains power supply	230 V ~ 50 Hz
Rated power	Basic module:25 VA Add-on module:80 VA
Safety class	2
Permissible temperatures, stationary, weatherproofed	+5 °C to +40 °C
Dimensions (W x H x D)	396 x 390 x 100 mm
Weight	Basic module and power supply unit: 1.9 kg Add-on module and power supply unit: 1.0 kg
S_0 ports	
Euro ISDN external (S_0 external) for basic access, DSS1 protocol	Basic module: 1 x Add-on module: –
Euro ISDN switchable (S_0 external / S_0 internal) for basic access, DSS1 protocol, or for ISDN terminals, DSS1 protocol	Basic module: 1 x Add-on module: up to 8 x; possible interface cards: – 4 x S_0 – 2 x S_0 and 6 x U_{pn} – 2 x S_0 and 6 x a/b
– Supply voltage	40 V ± 10%
– Supply power	2 VA for internal
– Range	150 m internal

OpenCom 130 (Cont.)

U_{pn} ports	
for system terminals and RFP 22 / 24 DECT base stations	Basic module: 3 x to connect system terminals, *not* DECT-enabled Add-on module: up to 16 x, all DECT-enabled; possible interface cards: – 4 x U_{pn} – 8 x U_{pn} – 2 x S_0 and 6 x U_{pn}
– Supply voltage	40 V ± 10%
– Supply power	3 VA per U_{pn} bus
– Range	1,000 m
a/b port	
for analogue terminals with pulse or DTMF dialling, flash duration of 60 to 310 ms	Basic module: 4 x Add-on module: up to 24 x; possible interface cards: – 4 x a/b – 8 x a/b – 2 x S_0 and 6 x a/b
– Supply voltage	40 V ± 10%
– Supply power	1.2 VA
– Feed current	25 mA
– Range	1,000 m
V.24 port (COM)	
for connection of a PC	Basic module: 1 x (optional)
– Range	3 m

OpenCom 130 (Cont.)

Doorstation equipment interface card	
for connection of doorstation equipment	Basic module: 1 x (optional)
Contact load of actor	1.5 A / 125 V
– Voltage range	$U_{\approx}$ = 5 V… 30 V
Sensor	Switched by low AC voltage
– Voltage range	$U_{\sim}$ = 6 V… 24 V

OpenCom 131

System data	
Mains power supply	230 V ~ 50 Hz
Rated power	Main module: 25 VA
Safety class	2
Permissible temperatures, stationary, weatherproofed	+5 °C to +40 °C
Dimensions (W x H x D)	396 x 390 x 100 mm
Weight (system only)	1.9 kg
S_0 ports	
Euro ISDN external (S_0 external) for basic access, DSS1 protocol	Main module: 1 x
Euro ISDN switchable (S_0 external / S_0 internal) for basic access, DSS1 protocol, or for ISDN terminals, DSS1 protocol	Main module: 1 x
– Supply voltage	40 V ± 10%
– Supply power	2 VA for internal
– Range	150 m internal
U_{pn} ports	
for system telephones	Main module: 3 x to connect system terminals, *not* DECT-enabled
– Supply voltage	40 V ± 10%
– Supply power	3 VA per U_{pn} bus
– Range	1,000 m

OpenCom 131 (Cont.)

a/b ports	
for analogue terminals with pulse or DTMF dialling, flash duration of 60 to 310 ms	Main module: 8 x
– Supply voltage	40 V ± 10%
– Supply power	1.2 VA
– Feed current	25 mA
– Range	1,000 m
V.24 port (COM)	
for connection of a PC	1 x (optional)
– Range	3 m
Doorstation equipment interface card	
for connection of doorstation equipment	1 x (optional)
Contact load of actor	1.5 A / 125 V
– Voltage range	$U_{\approx}$ = 5 V…30 V
Sensor	Switched by low AC voltage
– Voltage range	$U_{\sim}$ = 6 V…24 V

OpenCom 150

System data	
Mains power supply	230 V ~ 50 Hz
Rated power	160 VA
Safety class	2
Permissible temperatures, stationary, weatherproofed	+5 °C to +40 °C
Dimensions	Wall version: 396 x 390 x 100 mm (W x H x D) Rack version: – Width: 19-inch panel with flange for mounting in installation cabinet – Height: 3 U – Depth: approx. 340 mm
Weight (system only)	Wall version: 3.2 kg Rack version: 7.8 kg
S_0 ports	
Euro ISDN switchable (S_0 external / S_0 internal) for basic access, DSS1 protocol, or for ISDN terminals, DSS1 protocol	Possible interface cards: – 4 x S_0 – 2 x S_0 and 6 x U_{pn} – 2 x S_0 and 6 x a/b
– Supply voltage	40 V ± 10%
– Supply power	2 VA for internal
– Range	150 m internal
U_{pn} ports	
for system terminals and RFP 22 / 24 DECT base stations	Possible interface cards: – 4 x U_{pn} – 8 x U_{pn} – 2 x S_0 and 6 x U_{pn}
– Supply voltage	40 V ± 10%
– Supply power	3 VA per U_{pn} bus
– Range	1,000 m

OpenCom 150 (Cont.)

a/b ports	
for analogue terminals with pulse or DTMF dialling, flash duration of 60 to 310 ms	Possible interface cards: – 4 x a/b – 8 x a/b – 2 x S_0 and 6 x a/b
– Supply voltage	40 V ± 10%
– Supply power	1.2 VA
– Feed current	25 mA
– Range	1,000 m
V.24 port (COM)	
for connection of a PC	1 x (optional)
– Range	3 m
Doorstation equipment interface card	
for connection of doorstation equipment	1 x (optional)
Contact load of actor	1.5 A / 125 V
– Voltage range	$U_{\approx}$ = 5 V…30 V
Sensor	Switched by low AC voltage
– Voltage range	$U_{\sim}$ = 6 V…24 V

Note: The online help provides an overview of the limits that should be observed when configuring the OpenCom 100.

21. Notes on Disposal

In order to avoid any possible effects resulting from the disposal of electrical and electronic equipment containing substances damaging to the environment and human health, the European Parliament and Council directives

- 2002/96/EC on waste electrical and electronic equipment (WEEE) and
- 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS)

have been transferred into national law in all EU member states.

The primary aim of the legislation is the prevention of waste electrical and electronic equipment, and also the recycling, material recovery and any other form of recovery of such waste in order to reduce the quantities of waste to be disposed of and the amount of hazardous substances from electrical and electronic equipment in waste.

The product that you have purchased was developed in line with the current state of the art in an environmentally friendly manner and with a view to recycling. It therefore meets the specifications of the European directives.

The product is labelled with the symbol illustrated. If you wish to dispose of this product, this symbol obliges you to do so separately from unsorted domestic waste. Suitable facilities have been set up for the return of waste electrical and electronic equipment. Waste equipment can be handed in at these return centres free of charge. To find out where these return centres are located, please consult the information provided by the department of your local authority responsible for waste disposal.

Please note:
Electrical equipment does not belong in household waste.
Deposit it free of charge at a return centre.

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www.aastra.com

As of 19.07.2009
Subject to changes