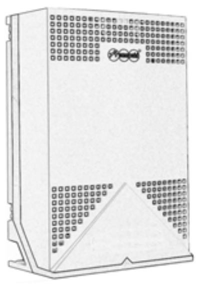
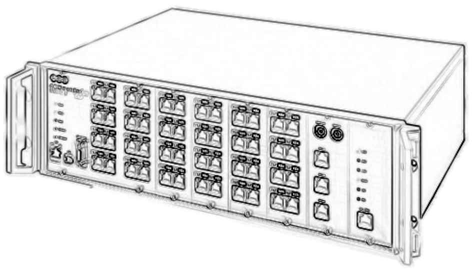
Configuration Manual

for the Administrator

Modular ISDN PBX

COMmander® Basic.2





Abbreviations used in this Manual

3PTY	Three party conference (**3** **P**ar**ty**)
AOCD	Charge information during and at the end of the connection (**A**dvice **O**f **C**harge **D**uring Call)
AOCE	Charge information at the end of the connection (**A**dvice **O**f **C**harge **E**nd of Call)
CF	Call forwarding
CCBS	Automatic call back on Busy (**C**ompletion of **C**alls to **B**usy **S**ubscriber)
CCNR	Automatic call back if nobody answers the call (**C**ompletion of **C**alls on **N**o **R**eply)
CD	Call deflection by the called partner (**C**all **D**eflection)
CD (PR)	Forward DDI of a Point-to-Point connection in the central office (**C**all **D**eflection (**P**artial **R**erouting))
CFB	**C**all **F**orwarding on **B**usy)
CFNR	**C**all **F**orwarding on **N**o **R**eply
CFU	**C**all **F**orwarding **U**nconditional
CLIP	**C**alling **L**ine **I**dentification **P**resentation
CLIP no screening	Presentation of customers specific telephone number information on the Point-to-Point connection (**C**alling **L**ine **I**dentification **P**resentation **no screening**)
CLIR	Call-by-Call suppression of the presented number (**C**alling **L**ine **I**dentification **R**estriction)
CNIP	Name presentation (**C**alling **N**ame **I**dentification **P**resentation)
CNIR	Call-by-Call suppression of the presented name (**C**alling **N**ame **I**dentification **R**estriction)
COLP	Presentation of the reached target number at the caller (**C**onnected **L**ine **I**dentification **P**resentation)
COLR	Suppression of the reached target number at the caller (**C**onnected **L**ine **I**dentification **R**estriction)
CW	Knocking (**C**all **W**aiting)
DDI	Direct Dialling In number in case of a PTP connection (**D**irect **D**ialling **I**n)
DSP	**D**igital **S**ignal **P**rocessor
ECT	**E**xplicit **C**all **T**ransfer
GSM	**G**lobal **S**ystem for **M**obile Communications
MSN	Multiple Subscriber Number in case of a PTMP connection (**M**ultiple **S**ubscriber **N**umber)
MWD	MWD number = Value added service number
NT	Network termination unit for the basic connection (**N**etwork **T**ermination)
UPS	**U**ninterruptable **P**ower **S**upply
VoIP	Internet telephony, voice transmission in IP networks (**V**oice **o**ver **I**nternet **P**rotocol)

Symbols and Signal Words used in this Manual

Warning	Warns of personal injury, for example, caused by hazardous electrical voltage.
Attention	Warns of damage to property.
Important	Indicates possible application errors and conditions that, for example, could cause function limitations or malfunctions during operation.
Note	Indicates supplementary information.

General limitation of legal responsibility and application

Auerswald products are not designed or intended for use in any life-saving and/or life support application and/or nuclear facility. When using our products for such applications a written agreement by Auerswald for each individual case is necessary in advance.

Accessory and Service components

Accessories and service parts can be bought at specialised stores or in the Internet shop distriCOM at http://www.districom.de. (Delivery is provided only in Germany and Austria.)

Dear Customer!

In order to install your new PBX, please commission the services of a trained professional.

This instruction manual thoroughly describes how to configure your PBX.

In order to get certain information quickly and carefully directed, the manual will offer you different helps and guide lines:

- The table of contents on page 3 gives you an idea of content and organization of the operating manual.
- The index on page 107 helps you to find certain text portions for a certain term.
- In the texts you will be referred to other chapters or pictures with the help of cross references.
- The headlines on each page remind you in which chapter you are at the present. On the left side of the pages the headlines of the actual chapter will be repeated. On the right side the headlines of the paragraph will be repeated.

Important: *Unintended use may cause e.g. functional restrictions or interferences, the destruction of the device or in a worst case scenario damage to persons.*

- *Read the **manual** carefully and store it for later reference.*
- *Also note the information about the guarantee, service, environment, CE symbol and declaration of conformity in the **insert** "Conditions of Guarantee, Information service".*
- *The device described in this manual is made for the indicated use only. If you are not sure about the **intended purpose** of the product, please contact your dealer.*

Security Advice

Warning

***Touching** the voltage carrying conductors or the telephone connections may cause an electric shock **dangerous to life**. Also individual modules may carry dangerous ringer voltages during operation.*

- *The case may only be opened by the **authorized dealer**[1].*
- ***Installation work** inside the open case as well as **maintenance services** involving the keys inside the case are only allowed to be executed by the **authorized dealer** [1].*
- *Remove the power plugs for both the PBX and all accessories from the power socket before you open the casing.*
- *The Schuko socket for connecting the PBX must be located near the PBX and be freely accessible at all times.*

Important: *In a PBX among other things personal data are processed subject to data privacy. These are numbers stored in the call data management or short messages (SMS) stored in the system telephones.*

In addition to this PBX systems may be attacked by dialer programs that enforce Internet connections via expensive dial-in numbers.

In general there is no one hundred percent protection against abuse of PBX functions. Please observe that a protection against abuse is only granted if ...

... unauthorized persons have no access to the PBX and its programming.

... the available authorizations (programming authorization via internal S_0 port, programming authorization, exchange line authorization, Call Restrictor etc.) are used reasonable.

... all options to assign passwords are consequently used. A responsible use of passwords is essential for the protection against abuse. Do not transfer passwords to unauthorized persons e.g. on a notepad.

... the access to data media e.g. backup discs by unauthorized persons is blocked. Destroy un-needed data media. Make sure that no paper remains in the public access area.

... make sure that only authorized persons have access to customer data. Make sure that no unauthorized person may process customer data (store, change, transfer, block, delete) or use it.

... the call data recording of your PBX and the LOGs of your NAT router are checked regularly for inconsistencies.

Additional advice against abuse may be found in the paper of the Bundesamt für Sicherheit in der Informationstechnik ("Sicherer Einsatz von digitalen Telekommunikationsanlagen") as well as on the service portal at the Auerswald web site (Internet address: www.auerswald.de).

1. Authorized dealer: These are persons that are trained for this purpose (e.g. certified electricians). They must have the necessary knowledge about the work in an area with potentially hazardous voltage. They must also have the knowledge about the latest electrical safety standards and requirements.

Translation Table of the Features

Sometimes features are known under different names. To find these in the manual if you do not know the used name used, a list of known features can be found here.

Feature	Corresponding Feature of the COMmander Basic.2
Return to operator	Fall-back to reception
Return on busy	Fall-back on Busy
Return to attendant	Fall-back to reception
Actuator	Relays
Call pick-up	Pick-up
Announcement before polling	Announcement before Answering
Automatic exchange line seizure	Direct exchange line telephone
Babyphone	Room monitoring
Class of service	Exchange line authorizations
Call Pick-up	Pick-up
Caller ID	Number presentation
Voice-announcement	InterCom
Direct call	Baby call/hotline
Station guarding	Do-not-Disturb
Remote maintenance	Remote programming
Relays	Relays
Charge limit	Call allowance account
Metering pulse	Charge meter pulse
Duplex operation	Handsfree mode (InterCom)
Tax indication	Charge display
Telephone directory	Phone book
Access to public exchange line	Transfer of an Exchange Line Access
Automatic line connection	Direct exchange line telephone
Music while waiting	Music on Hold
Hold-on tone	Music on Hold
Private call	Private exchange line access
Paging	Speaker announcement
Project code	Project numbers
Pre-selection of external lines	Exchange line reservation
Route selection	Exchange line access number
Knocking	Call waiting
Call number transmitting	Number presentation
Bell signal	Ringer rhythms
Call deviation	Call forwarding immediately
Call deviation on busy/on no reply	Call forwarding on Busy/on no Reply
Silence/station guarding	Do-not-Disturb
Sensor	Door bell button
Follow-me	Follow-me
Day/Night changeover	Configurations
Team function	Group creation
Switching	Transfer
Call switching	Transfer
Simplified call transfer	Blind Transfer
Switch to exchange line	Call transfer to externals
Adopt	Pick-up
Automatic call	Baby call/hotline
Charge indication	Charge display
On-hook dialling	Dial preparation
Waiting line	Waiting Field

Operation Advice

Help: You can open the online help with the information about the currently viewed page by clicking on the question mark symbol on each page. Inside the online help additional information or help files about other pages can be opened via “additional information” in an alphabetic ordered list.

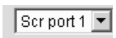
Log out: In the bottom left corner there is the button necessary to close the web interface. Via mouse click on “Logout” you can directly return to the registration page.

Accept entered Data: Before leaving a page it is necessary – with some exceptions – to confirm your changes via mouse click on the field “Execute” in the action line. This way the displayed data is stored in the PBX. The successful storage of data is shown by the flashing save icon right at the top of the page.

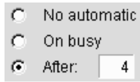
To store changes in individual table lines, you can also click the green save icon in the corresponding line.

Entry in a free field: A free field is available for the entry of a name or a number. Click with the left mouse button in the corresponding field and enter a number or a name with the keyboard. Before leaving the page it is necessary to confirm your entries via mouse click on the field “Execute” in the action line. The following signs must not be used: “ ” ' # $ & % < > / \

Change number or name: Click with the left mouse button twice on the entry to be overwritten. It will be marked in blue and can directly be overwritten or deleted with the Delete/Backspace key. Before leaving the page it is necessary to confirm your changes via mouse click on the field “Execute” in the action line.



Switch-over Functions: Functions that can do more that switching on/off and have a limited selection of only some options, a preselection has already been done in the field. To change this selection, you have to choose from the popup menu via mouse click. You open the popup menu with the left mouse button on the arrow.



For some functions there are little circles as switches additionally to the listed options. The selected setting is marked with a black point within the circle. The function is switched over via mouse click on an empty circle. The activation of a setting causes the parallel deactivation of all other choices.

Activate/deactivate Functions: A square represent a switch. An empty square means “out” or “no”; there against a little hook means “on” or “yes”. The selection is done with a simple click with the left mouse button.

Add entries in the list: To create new entries in the list, you fill in the empty entry fields in the bottom line of the table and confirm your entries with a mouse click on the field “Execute” in the action line. Then one or more lines are added to the list.

Delete entries in the list: To delete an entry, click the red recycle bin in the corresponding table line and, subsequently, click the green check mark below.

Alternatively, you select “Delete” in the action line first instead of “Accept”. Then you can mark one or more entries in the first table column via mouse click to delete them (little hook in the square).

If you like to select all entries you can activate the little box at the bottom left side of the action bar. Then you delete the marked entries with a mouse click on the field “Execute” in the action line.

Colour schemes: To adapt the interface to the personal taste the PBX is offering four colour schemes. The may be configured for the admin under *Administration* ▶ *Server configuration* and for the individual users under *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top.

Authorization Assignment

Authorization Levels for the Access to the Web Interface

To prevent that important settings are changed by mistake or by unauthorized persons, there are different authorization levels in the PBX. The access to the Web interface of the PBX is divided into three authorization levels: Administrator (admin), sub-administrator (sub-admin) and user.

Each of these authorization levels has a user name and a PIN (see table). These have to be entered for each registration to the Web interface (in case of an internal and an external access to the PBX).

Important: *All PINs in the PBX are unique that means that it is not possible to assign the same PIN twice in the PBX.*

Do not use dates of birth or dates as PINs. This makes it easy for an attacker to find out the correct PIN. PINs which are easy to guess, such as 111111 or 123456, should also be avoided.

As the PINs can also be entered via telephone, only digits are possible. A PIN is always 6 digits long.

After entering the wrong PIN three times to access the web interface, there is a timeout of 60 seconds. During this time no PIN entry is possible.

No PINs are pre-defined in the default factory settings.

Administrator (Admin)

The administrator is e.g. the authorized dealer or the administrator of the PBX.

The administrator has access to the configuration manager without restrictions. Via this access he can configure the PBX completely but he can also release PBX functions with the PBX dongle. This authorization level also allows the change of the other PINs without knowing them as well as the assignment of access authorizations.

Sub-Administrator (Sub-Admin)

A sub-administrator (up to four are possible) is an internal supervisor. This is the person that has the role of a local administrator at the location of the PBX. On the page *Administration* ▶ *User PINs* this functional level can be assigned to four individual internal subscribers of the PBX. Therefore a sub-administrator is also user at the same time and can also log-in as such at the Web interface.

The access authorizations to the Web interface are assigned to the sub-administrators by the administrator according to the local requirements. These assignments are made on the page *Administration* ▶ *Access Authorizations*. With the exception of some pages (e.g.

Administration ▶ *Access Authorizations*) the whole configuration manager or alternatively only some individual pages can be released to the sub-administrators.

Each of the four possible sub-administrators has the same access and modification authorizations.

Note: *If the sub-administrator enters his internal telephone number as user name, this registration is recognized as an user log-in and the Web interface is presented in user-mode.*

User

The user is any internal subscriber of the PBX that may receive additional authorizations by assigning a user PIN.

The access authorizations to the Web interface are assigned to the user by the administrator according to local requirements. This assignment/setting is done on the page *Administration* ▶ *Access Authorizations*. The possible range of page releases is limited to a very small number of own subscriber and group settings.

Each user has the same access authorizations. There modification rights may differ depending on the profiles (see *Profiles and Properties* on page 11).

Authorization Levels for the Operation via Telephone

On the system telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB*, the three authorization levels of the PBX with the corresponding PINs are also used. The authorizations are set individually for each telephone in the telephone menu or with the corresponding configuration manager or PC program *COMfortel Set* (see the manual for the system telephone). It is possible to release all the functions even for use without a password (authorization level “guests”).

Additionally the PINs of the PBX are needed for the operation of some functions with the standard telephone:

The user PIN mainly offers the access to a few personal functions such as private calls and the activation of Call Restrictor and Deblocker. The user PIN is used to remote-control the own telephone in some functions if the programming sequence is used on another internal telephone.

The admin PIN and the sub-admin PIN offer the access to functions such as the setting of call allowance accounts and the recording of announcements.

External Access to the PBX

The PBX is protected against an external access (Dial-up to the PBX via PPP/remote configuration) with the admin PIN or with two different PINs. First of all you need one of the three PINs that you also use for an internal access to log-in to the Web interface. For the preceding necessary dial-up you have to enter the user name “external” and the external PIN (or admin user name and admin PIN) into the connection dialog.

Besides this the external PIN is needed for the operation of some functions such as Follow-me, Remote control and Room Monitoring from an external telephone (Remote Programming). The dial-up happens via a special telephone number - the remote programming telephone number. The PBX accepts the call automatically and the entry of the PIN or the programming sequence is made via DTMF.

Note: The external PIN cannot be used for log in to the web interface.

	Corresponding PIN	**Corresponding User Name**
Authorization level Administrator	Admin PIN (is assigned during the first commissioning; may be changed on the page *Administration* ▶ *Server configuration*)	“admin” (in the default factory setting; may be changed on the page *Administration* ▶ *Server configuration*)[1]
Authorization level Sub-administrator	Sub-Admin PIN (corresponds to the user PIN of the corresponding sub-administrator)	“sub-admin” (not changeable; is valid for all four sub-administrators)
Authorization level User	User PIN (assigned to each user on the page *Administration* ▶ *User PINs*)	internal telephone number of the user
External Access	External PIN (assigned to the page *COMset* ▶ *Global settings* ▶ *Remote configuration*)	“external” (not changeable)

1. If the configuration manager of the PBX is to be reached from the Internet (via http or better via https), you should also change the user name of the administrator (admin) for reasons of security.

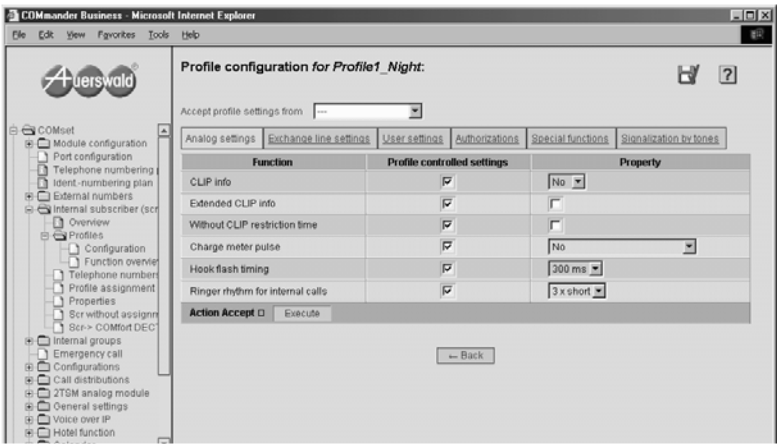
Profiles and Properties

Due to the assignment of a user PIN you can offer each user the option to influence the features of his own extension, his group or his voice mail/fax box. As only a little part of the features is reasonably configurable by the user, you can permanently set all other features (e. g. exchange line authorizations, other authorizations) by assigning of a profile (for subscriber, group, voice mail/fax box).

Under *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Profiles* ▶ *Configuration* you can configure the requested profiles (for groups under *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Profiles* ▶ *Configuration,* for voice mail/fax boxes under *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profiles* ▶ *Configuration*).

In doing so you only have to do make the settings in the column "***Property***" that you do not want to leave to the user himself. For these settings you have to make a hook in the column "***Profile controlled settings***".

COMset ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Profiles* ▶ *Configuration*



For the settings that do not have to be profile-controlled, the little hook has to be removed in the column "***Profile controlled settings***". In this case the setting in the column "***Property***" is without meaning.

If the profile was created and assigned to a subscriber, the features of the extension can be reviewed and changed under *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* (for groups *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties,* for voice mail/fax boxes *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties*). This is also possible for the user later.

In the column "***Mode***" the features of the extension are presented in an overview.

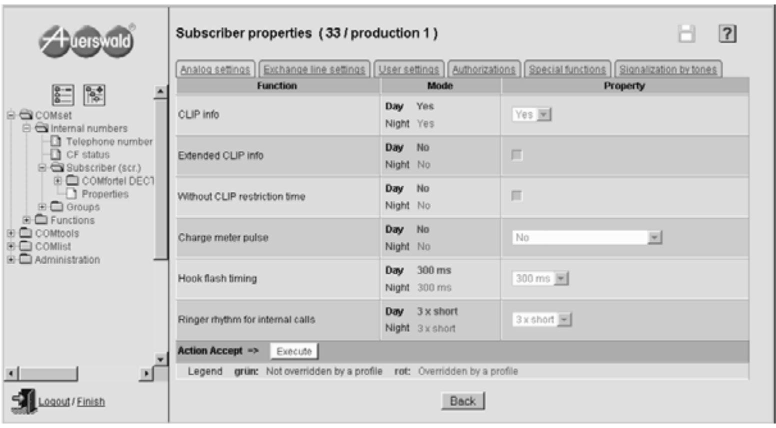
If the PBX uses several configurations (e.g. for day, night, holidays) and if different profiles are in use for the telephone/group/voice mail/fax box, you can learn this from the different entries listed on under the other.

To differ the changeable entries from the permanent ones, all entries are marked in colours.

Red character colour: The setting is not released by the administrator to be changed by the user (in other words controlled by the profile).

Green character colour: The setting can be changed by the user.

COMset ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties*

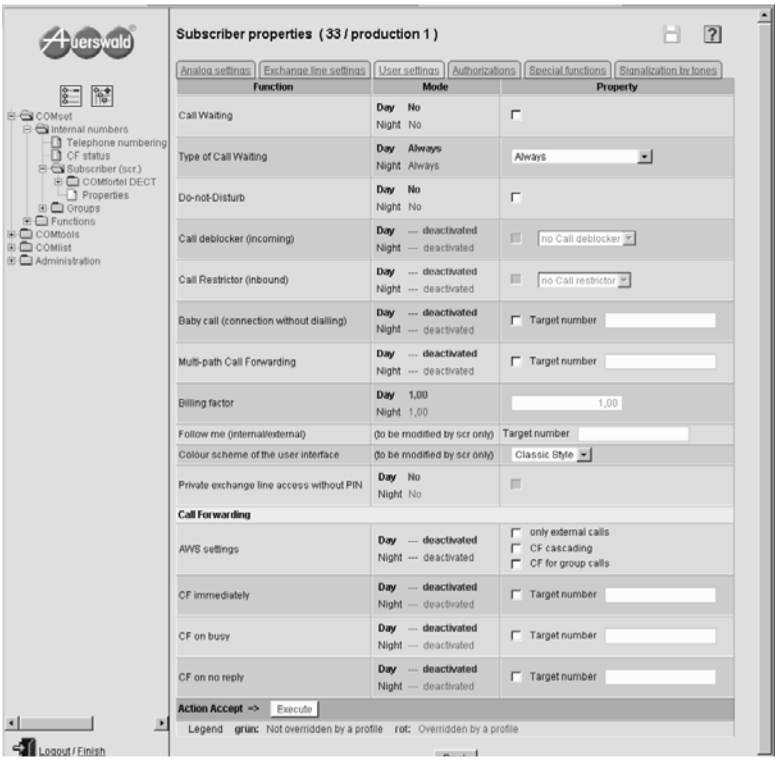


If you like to change a setting (requirement: the corresponding entry is marked in green), you will achieve it by working in the column “***Property***”.

Settings that you can do here are listed in green colour in the column “***Mode***”. If a setting is only controlled in some configurations (e.g. night) by the profile, you can change the setting in another configuration (e.g. day). If a setting is not controlled in any configuration by a profile, the setting will be changed in all configurations.

If you made requested settings on a page, you have to confirm these changes by clicking on the button “***Execute***” in the action line.

In the column “***Mode***” you can control these settings afterwards.



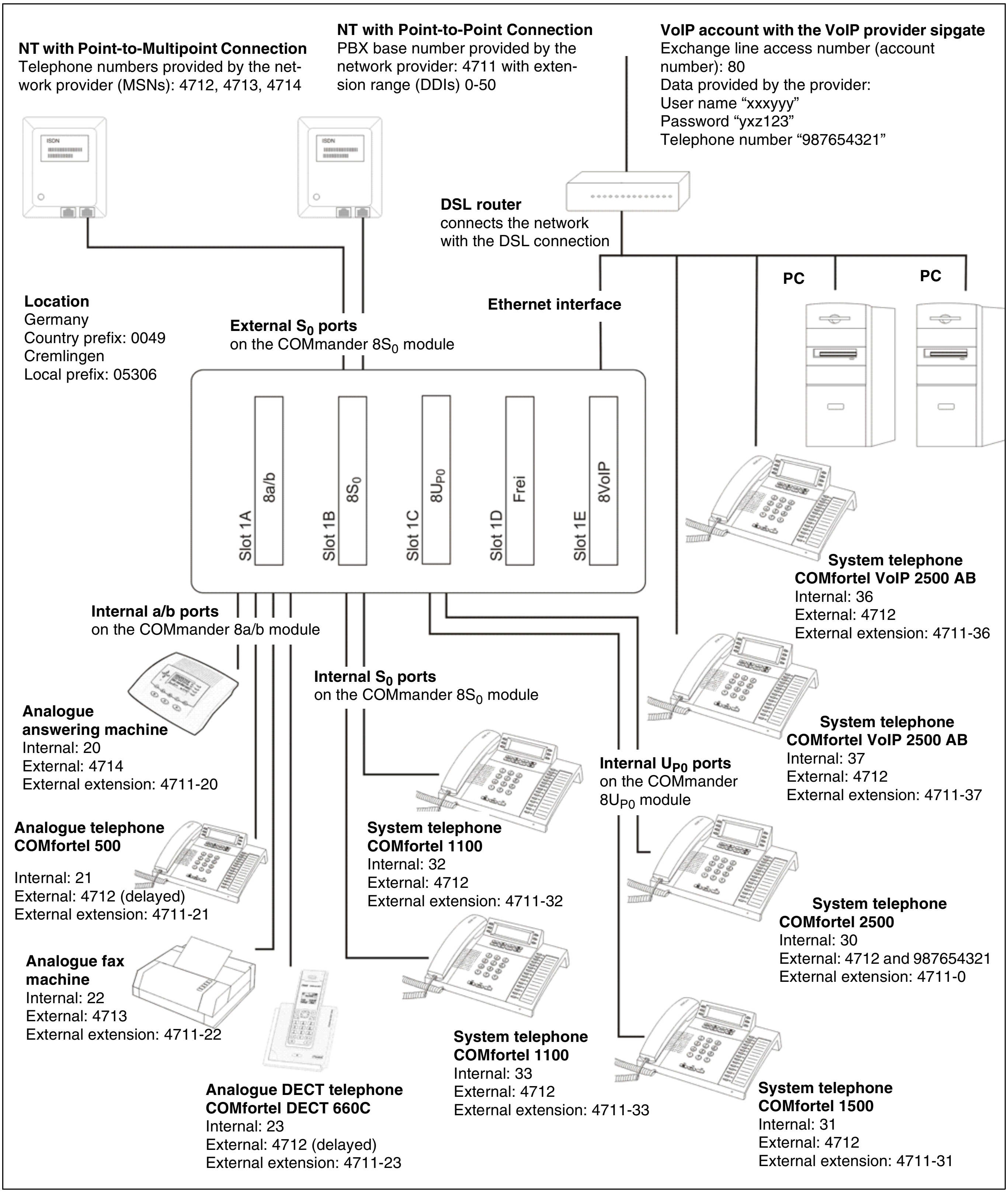
Example 1: Basic Configuration with Internal Subscribers and Call Distribution

At the end of this section, an example (see figure) is used to describe the procedure for configuring the *COMmander Basic.2* step by step.

Please refer to the installation and commissioning instructions for the basics on using the modules.

Note: *Note that this section thoroughly describes only the most important settings. You will be referred to the corresponding sections for additional settings.*

The combination of the modules and devices listed here is an example only and may vary greatly from your PBX.



Logging into the Configuration Manager

Important: *To be able to perform basic configuration, the PBX and your computer must be correctly connected with each other.*
The settings for this are described in the Installation and Commissioning Instructions and must be noted when the PBX is put into operation for the first time.

Start the configuration manager. Enter the user name "***admin***" and the 6-digit admin PIN and confirm by clicking "***Log-in***".

You must logged on as administrator in order to perform the configuration because initially only the administrator has the necessary rights. After you have logged in, you see a menu with a tree structure on the left side of the page. You now have access to the entire system configuration. The structure is similar to the directory structure on a hard drive where each folder can be described with a unique path.

For reasons of simplicity, this **path information** is shown above each figure for each configuration example. This shows the exact location of each menu option in the tree structure.

Hardware Configuration

Assign the current modules to the expansion slots on the mainboard.

To do this, open the page displayed in the figure to the right.

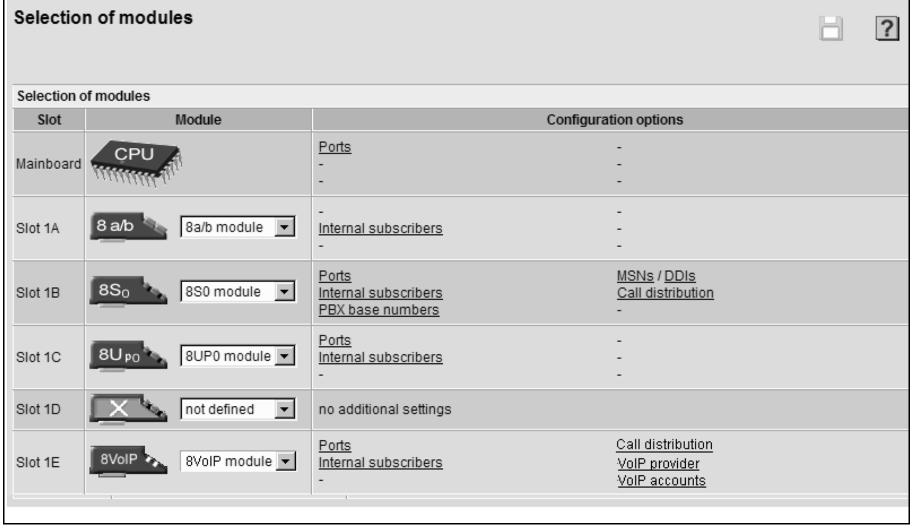
The "***Slot***" column lists each module slot on the mainboard (slot 1A to 1E).

Under "***Module***", select a module from the list field that has been inserted in the associated module slot. For available module slots, select "not defined" from the list field.

In the "***Configuration options***" column, you can use the hyperlinks to directly access the settings described at the end.

These settings complete the hardware configuration.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Hardware ▶ Selection of modules



Selection of modules

Selection of modules

Slot	Module	Configuration options	
Mainboard	CPU	Ports - -	- - -
Slot 1A	8 a/b — 8a/b module	- Internal subscribers -	- - -
Slot 1B	8S0 — 8S0 module	Ports Internal subscribers PBX base numbers	MSNs / DDIs Call distribution -
Slot 1C	8UP0 — 8UP0 module	Ports Internal subscribers -	- - -
Slot 1D	not defined	no additional settings	
Slot 1E	8VoIP — 8VoIP module	Ports Internal subscribers -	Call distribution VoIP provider VoIP accounts

Note: *Note that parameters set here correspond to the real hardware settings on your PBX (question: Which module is inserted in which slot?). If this is the case, the settings are marked in black. If this is not the case, the settings are marked in red.*

For more information on this, refer to the module descriptions in the Installation and Commissioning Instructions.

Port Configuration

Port configuration is the basic set-up for the inserted modules as well as for the CPU. The connection assignments for the ports must be set depending on the intended use and adapted to the actual connection options on site.

Note: *If a COMmander 8a/b module is inserted, no port configuration is necessary for this module.*

If using a COMmander 2TSM analog module, see Example 2: Adding a COMmander 2TSM analog Module on page 28.

If using a COMmander S_{2M} module, see Example 4: Installing a COMmander S_{2M} module for a primary multiplex interface on page 39.

Mainboard CPU

Here you can modify the settings for connecting to a computer or network that were configured during operation, if required.

In addition, you can specify here how many VoIP channels should be available for any internal or external VoIP calls.

To do this, open the page displayed in the figure to the right.

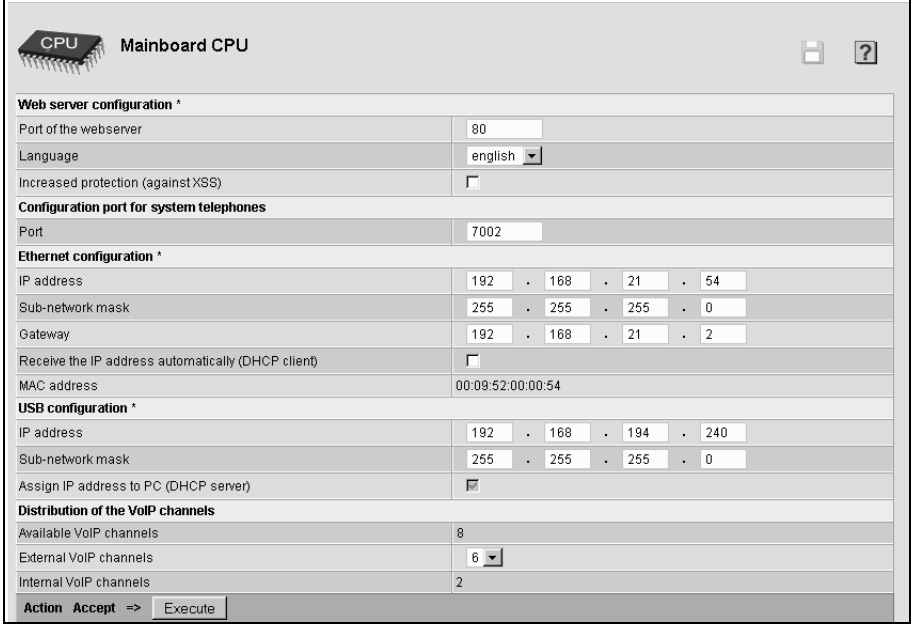
The PBX has two VoIP channels by default. When using a VoIP module, the two VoIP channels included by default are switched off! The number of VoIP channels available when using VoIP modules consists of the number of channels on the VoIP modules together with the additional channels enabled on the system dongle via the Upgrade Center.

Set the number of channels your want under "***External VoIP channels***". Take into account the capacity of your Internet connection.

The number of internal VoIP channels available for internal telephony is automatically calculated from the number of available VoIP channels minus the VoIP channels reserved for external calls.

Click "***Execute***" to accept the settings in the PBX.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Hardware ▶ Port configuration ▶ Mainboard CPU



Important: *In addition, the Ethernet settings are not applied until the system has been restarted (rebooted).*

It has to be noted that the last byte of the IP address for the USB configuration is permanently set to 240 (XXX.XXX.XXX.240) and that, as a result, the remote address is permanently set to .241.

VoIP Module

Configure the Ethernet configuration on the COMmander 8VoIP/16VoIP module.

To do this, open the page displayed in the figure to the right.

If you would like to allocate a permanent IP address to the module, clear the check box "***Automatically assign IP address (DHCP client)***".

Under "***IP address***", enter the desired IP address.

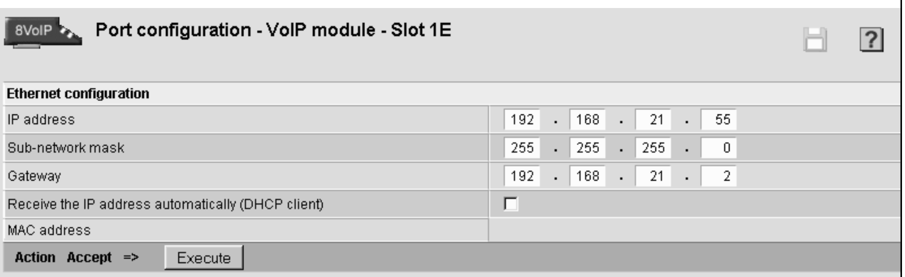
Under "***Sub-network mask***", change the subnet mask entered, if necessary.

Under "***Gateway***", change the gateway entered, if necessary.

Click "***Execute***" to accept the settings in the PBX.

This completes the configuration of the COMmander 8VoIP/16VoIP module.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Hardware ▶ Port configuration ▶ Slot ... VoIP



Important: *Please contact the responsible system administrator and configure the settings according to his instructions.*

Internal and external S_0 ports

Configure the internal and external S_0 ports on the COMmander $8S_0$ module and the COMmander $4S_0$ module.

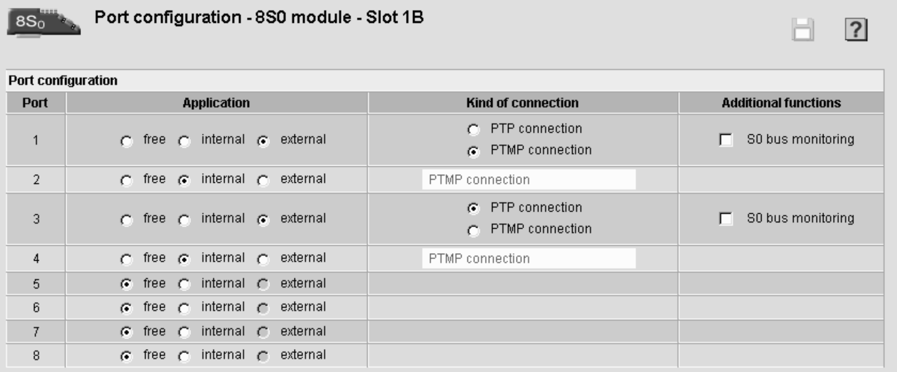
To do this, open the page displayed in the figure to the right.

Under "***Application***", configure the S_0 ports as an internal or external port as you wish. (Unused ports are set to available.)

On the external S_0 ports under "***Kind of connection***", configure the connection type of the NTs you have requested from your network provider. (When using internal ports, the connection type is automatically set to a PTMP connection.)

On the external S_0 ports under "***Additional functions***", activate, if necessary, S_0 bus monitoring in order to prevent waiting times on the external dial tone (recommended).

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Hardware ▶ Port configuration ▶ Slot ... 8S0



This completes the configuration of the COMmander $8S_0$ module.

Note: *Note that parameters set here correspond to the actual port configuration on your PBX (which port is set to internal or external). If this is the case, the settings are marked in black. If this is not the case, the settings are marked in red.*

The port configuration on the COMmander $4S_0$ module is the same as the configuration described here.

For more information on this, refer to the description of the COMmander $8S_0$ module and the COMmander $4S_0$ module in the installation and commissioning instructions.

Internal U_{P0} ports

Configure the internal U_{P0} port on the COMmander $8U_{P0}$ module.

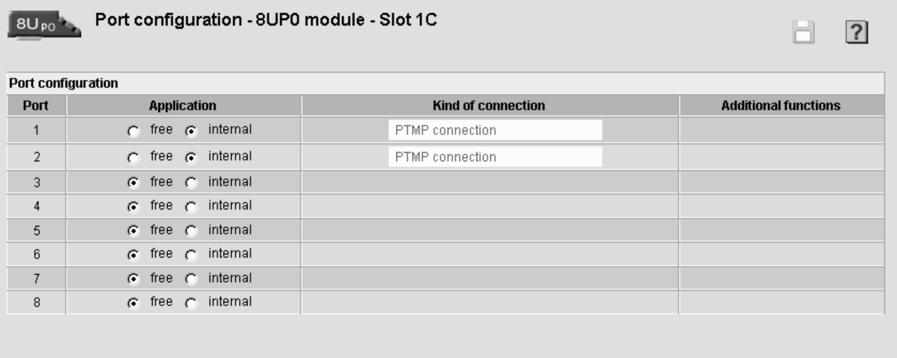
To do this, open the page displayed in the figure to the right.

Under "***Application***", configure the U_{P0} ports as an internal port as you wish. (Unused ports are set to available.)

(For U_{P0} ports, the connection type is automatically set to a Point-to-Multipoint connection.)

This completes the configuration of the COMmander $8U_{P0}$ module.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Hardware ▶ Port configuration ▶ Slot ... 8UP0



Note: *For more information on this, refer to the description of the COMmander $8U_{P0}$ module in the installation and commissioning instructions.*

Internal Telephone Numbers

In order for the connected terminal devices to be available, the internal ports must be configured with internal telephone numbers. This means that each terminal device connected receives a subscriber telephone number.

It is a good idea to create a telephone numbering plan for the devices (see also Example 1 on page 13) and then use the following steps to transmit this to the PBX.

Note: *On a Point-to-Point connection, the configuration of a linear call distribution is usual (e. g., a call to 4711-21 is distributed to the internal subscriber telephone number 21). This includes needing to assign internal telephone numbers that lie in the extension number range (DDIs) assigned by the network provider. Call distribution can then be done automatically (for more information on this, see Call Distribution on an ISDN Point-to-Point Connection on page 23 and Call Distribution on VoIP Point-to-Point Connections on page 24).*

If multiple internal devices need to be accessible via an external telephone number, a common internal group for the internal subscribers must first be configured.

The PBX lets you assign internal telephone numbers 10-9999. Of this number range 10-9999, the following telephone numbers are assigned:
- Account numbers
- Internal subscriber telephone numbers
- Internal group telephone numbers
- Internal CAPI dial-in numbers

- Internal telephone numbers for automatic switchboards
- Internal basis telephone numbers for open callbacks
- Internal telephone numbers for door terminals
- Internal telephone numbers for voice mail/fax boxes
- Internal telephone numbers for audio outputs
- Short-code numbers
- Emergency numbers

Double assigning a number is not possible!
On the page COMset ▶ Internal numbers ▶ Telephone numbering plan, you can get an overview of the internal telephone numbers already assigned at any time.

The telephone numbers can consist of two to four digits. Using telephone numbers with few digits (two or three-digit numbers) inevitably restricts the possible supply of telephone numbers with more digits. For example, if the telephone number 12 is assigned, the telephone numbers 120-129 and 1200-1299 are no longer available.

For some functions, only telephone numbers with a maximum of 3-digits can be assigned because the last digit is need for the function (e. g., door terminal numbers, telephone numbers for open callbacks).

For more than 48 subscriber telephone numbers, an additional activation is required. For more information on this, refer to the Installation and Commissioning Instructions.

Analog Subscribers

Assign internal telephone numbers from your telephone number plan to the internal a/b ports.

To do this, open the page displayed in the figure to the right.

In the "***Telephone Number***" column under "***from:***", enter a telephone number from your telephone number plan.

Under "***Name***", enter a suitable name with a maximum of 16 characters.

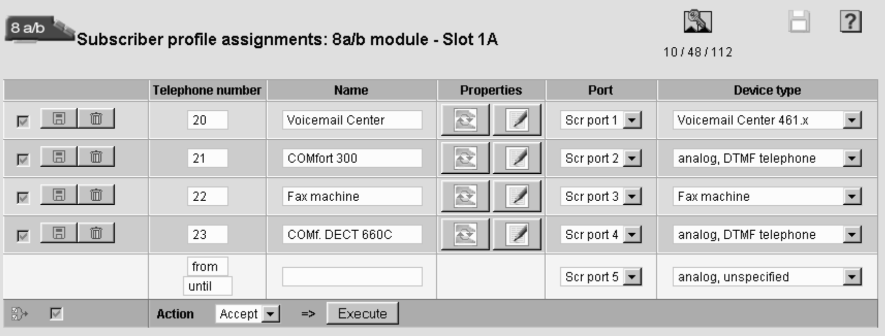
Under "***Port***", select the a/b port that the end device is connected to from the list field.

Under "***Device type***", select a suitable type for your end device from the list field.

Then click "***Execute***" to apply the settings to the PBX.

Configure the settings for all additional telephone numbers.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Internal numbers ▶ Subscriber (scr) ▶ Telephone numbers ▶ Slot ... 8a/b



This completes the configuration of the analog subscribers.

Note: *If you have contiguous telephone number ranges, you can enter the beginning and end values under "**from: until**". Only after clicking "**Execute**" will the name, port and device type be available for entries.*

As opposed to an S_0 port, only one telephone can be connected to an a/b port.

*You can delete analog subscribers, which are no longer needed, on this page. Click the red recycle bin in the corresponding table line and, subsequently, click the green check mark below. Or under "**Action**", configure Delete and select the subscribers to be deleted from the left column. Then click "**Execute**".*

ISDN subscribers on internal S_0 ports

Assign the internal telephone numbers from your telephone number plan to the internal S_0 ports.

To do this, open the page displayed in the figure to the right.

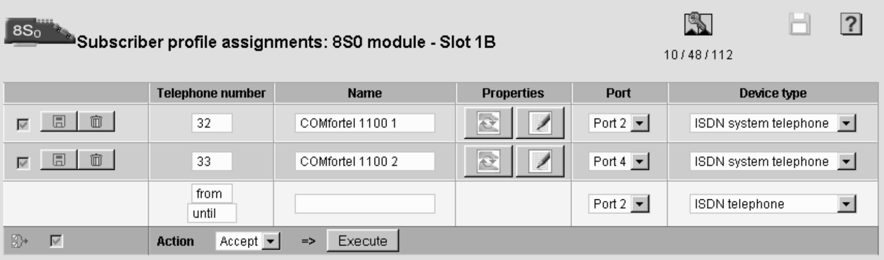
In the "***Telephone Number***" column under "***from:***", enter a telephone number from your telephone number plan.

Under "***Name***", enter a suitable name with a maximum of 16 characters.

Under "***Port***", select the S_0/U_{P0} ports that the end device is connected to from the list field.

Under "***Device type***", select a suitable type for your end device from the list field.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Internal numbers ▶ Subscriber (scr) ▶ Telephone numbers ▶ Slot ... 8S0



Then click "***Execute***" to apply the settings to the PBX.

Configure the settings for all additional telephone numbers.

This completes the configuration of the ISDN subscribers on internal S_0 ports.

Note: *If you have contiguous telephone number ranges, you can enter the beginning and end values under "**from: until**". Only after clicking "**Execute**" will the name, port and device type be available for entries.*

It is a good idea to not connect more than two devices on each S_0 port in order to allow separate calling on either of the two B channels.

The internal telephone number entered here needs to be entered in the ISDN unit or system telephone as the (first) MSN. Also for an ISDN unit, make sure that the telephone number is transmitted to the PBX.

*ISDN subscribers, which are no longer needed, on this page. Click the red recycle bin in the corresponding table line and, subsequently, click the green check mark below. Or under "**Action**", configure Delete and select the subscribers to be deleted from the left column. Then click "**Execute**".*

ISDN subscribers on internal U_{P0} ports

Assign the internal telephone numbers from your telephone number plan to the internal U_{P0} ports.

To do this, open the page displayed in the figure to the right.

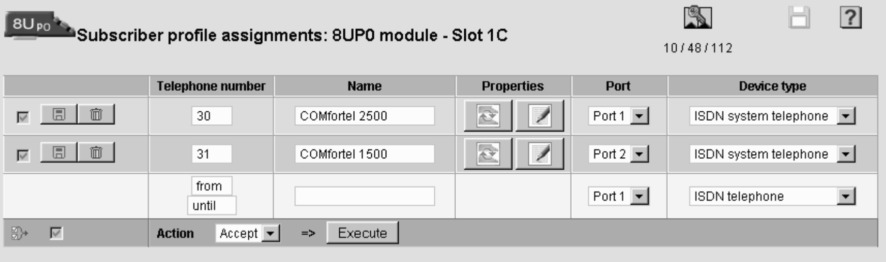
In the "***Telephone Number***" column under "***from:***", enter a telephone number from your telephone number plan.

Under "***Name***", enter a suitable name with a maximum of 16 characters.

Under "***Port***", select the S_0/U_{P0} ports that the end device is connected to from the list field.

Under "***Device type***", select a suitable type for your end device from the list field.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Internal numbers ▶ Subscriber (scr) ▶ Telephone numbers ▶ Slot ... 8UP0



Then click "***Execute***" to apply the settings to the PBX.

Configure the settings for all additional telephone numbers.

This completes the configuration of the ISDN subscribers on internal U_{P0} ports.

Note: *If you have contiguous telephone number ranges, you can enter the beginning and end values under "**from: until**". Only after clicking "**Execute**" will the name, port and device type be available for entries.*

One system telephone, COMfortel 1100/1500/2500/2500 AB, can be connected to each U_{P0} port. When using standard ISDN telephones, a U_{P0}/S_0 adapter is required. For more information on this, refer to the description in the Installation and Commissioning Instructions.

The internal telephone number entered here needs to be entered in the ISDN unit or system telephone as the (first) MSN. Also for an ISDN unit, make sure that the telephone number is transmitted to the PBX.

*ISDN subscribers, which are no longer needed, on this page. Click the red recycle bin in the corresponding table line and, subsequently, click the green check mark below. Or under "**Action**", configure Delete and select the subscribers to be deleted from the left column. Then click "**Execute**".*

VoIP Subscribers

Assign the VoIP telephones internal telephone numbers from your telephone number plan.

To do this, open the page displayed in the figure to the right.

In the "***Telephone number***" column under "***from:***", enter a telephone number from your telephone number plan.

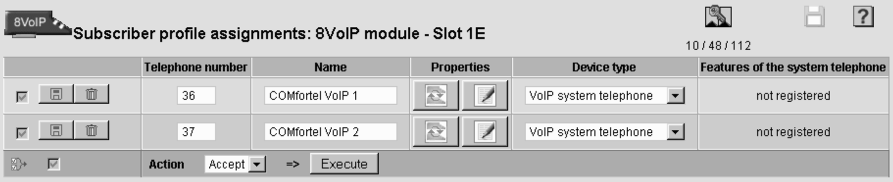
Under "***Name***", enter a suitable name with a maximum of 16 characters.

Under "***Device type***", select a suitable type for your end device from the list field.

Click "***Execute***" to accept the settings in the PBX.

Configure the settings for any additional telephone numbers.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Internal numbers ▶ Subscriber (scr) ▶ Telephone numbers ▶ Slot ... VoIP



This completes the configuration of the VoIP subscribers.

Note: *If you have contiguous telephone number ranges, you can enter the beginning and end values under "**from: until:**" . Only after clicking "**Execute**" will the name and device type be available for entries.*

The maximum number of possible VoIP subscribers is specified by the maximum number of possible internal VoIP channels: In order to configure VoIP subscribers, at least one internal VoIP channel must be available.

On the page Administration ▶ User PINs, you can configure a user PIN for each VoIP subscriber (recommended).

Configure the following settings on a VoIP system telephone: Enter the Internal number as an MSN and – if configured – the user PIN as the registration PIN.

Configure the following settings on a standard VoIP telephone or soft phone: Create a provider named "PBX" and then enter the PBX IP address as the registrar and domain. Create an account for the provider "PBX" and enter the internal phone number for the user name and the associated user PIN as the password – if configured. For some standard VoIP telephones or soft phones, it might also be necessary to enter the IP address on the PBX as a proxy.

*VoIP subscribers that are no longer needed can be deleted on this page. Click the red recycle bin in the corresponding table line and, subsequently, click the green check mark below. Or under "**Action**", configure Delete and select the subscribers to be deleted from the left column. Click "**Execute**".*

Creating Internal Groups

Create the internal groups required.

To do this, open the page displayed in the figure to the right.

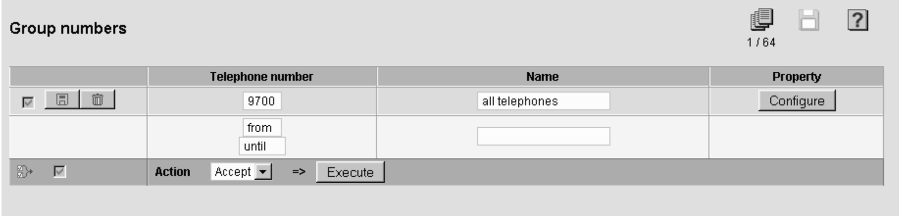
In the "***Telephone Number***" column under "***from:***", enter a telephone number for the group.

Under "***Name***" enter a suitable name with a maximum of 16 characters.

Then click "***Execute***" to apply the settings to the PBX.

Configure the settings for all additional groups required.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Internal numbers ▶ Groups ▶ Telephone numbers



Note: *If you have contiguous telephone number ranges, you can enter the beginning and end values under "**from: until**". Only after clicking "**Execute**" will the name be available for entries.*

*You can delete groups, which are no longer needed, on this page. Click the red recycle bin in the corresponding table line and, subsequently, click the green check mark below. Or under "**Action**", configure Delete and select the groups to be deleted from the left column. Then click "**Execute**".*

Assigning Group Members (with Ringer Delay)

Assign members to the available groups.

To do this, open the page displayed in the figure to the right.

Under "***Please select a group:***", select the group to be configured first from the list field.

In the column "***Telephone number / name***" under "***from: until:***", select all the subscribers and groups from the list fields that should be called when the group is called.

Then click "***Execute***" to apply the settings to the PBX.

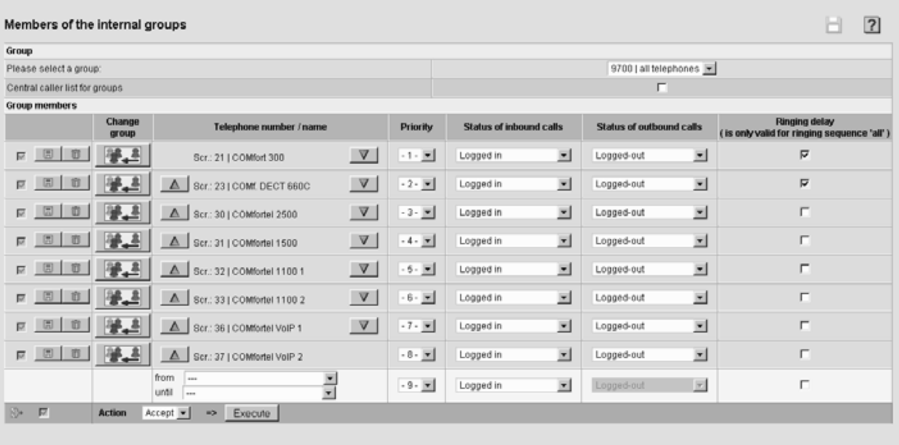
If some of the group members should get a delayed call when the group is called, activate this under "***Ringing delay (is only valid for ringing sequence "all")***".

Then click "***Execute***" to apply the settings to the PBX.

Configure the settings for all additional groups.

This completes group creation.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Internal numbers ▶ Groups ▶ Group members



Note: *The ringer delay time set to 5 seconds can be changed on page COMset ▶ Internal numbers ▶ Groups ▶ Properties ▶ Configure ▶ Reachability. You also have the option of selecting other ringing sequences (for more on this, refer to Ringing Sequence on page 69).*

*When a group is created, the existing settings under "**Status of inbound/outbound calls**" do not need to be changed for the call distribution to be correct (see also Assign, Log in, Log out on page 68).*

*On this page, you can delete subscribers and groups that should no longer be called when the group is called. Click the red recycle bin in the corresponding table line and, subsequently, click the green check mark below. Or under "**Action**", configure Delete and select the group members to be deleted from the group in the left column. Then click "**Execute**".*

External Telephone Numbers

The telephone numbers on the ISDN connection provided by your network provider must be entered in the PBX. In addition, the data for the VoIP accounts used and the associated VoIP provider must be entered.

For each external connection, a separate telephone number must be assigned (in Example 1 on page 13, this is an ISDN Point-to-Point connection, an ISDN Point-to-Multipoint connection as well as a VoIP account with the provider ...).

Collect the telephone numbers/data provided by your network provider/VoIP provider for the following settings.

Location

In the PBX, configure the location where it should be operated.

To do this, open the page displayed in the figure to the right.

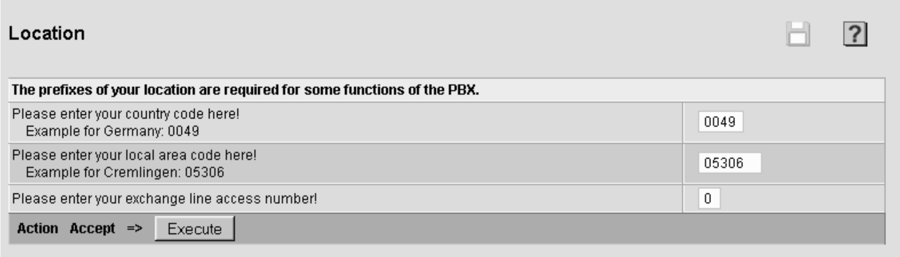
Under “***Country code***”, enter the country prefix for the location of the installation.

Under “***local area code***”, enter the local prefix for the location of the installation.

Then click “***Execute***” to apply the settings to the PBX.

This sets the location in the PBX.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ *External numbers* ▶ *Location*



ISDN Point-to-Multipoint Connection

In the PBX, configure each existing ISDN Point-to-Multipoint connection with the telephone numbers provided by the network provider.

To do this, open the page displayed in the figure to the right.

Under “***Multiple subscribers numbers for port***”, select the external S_0 port that you would like for your Point-to-Multipoint connection from the list field.

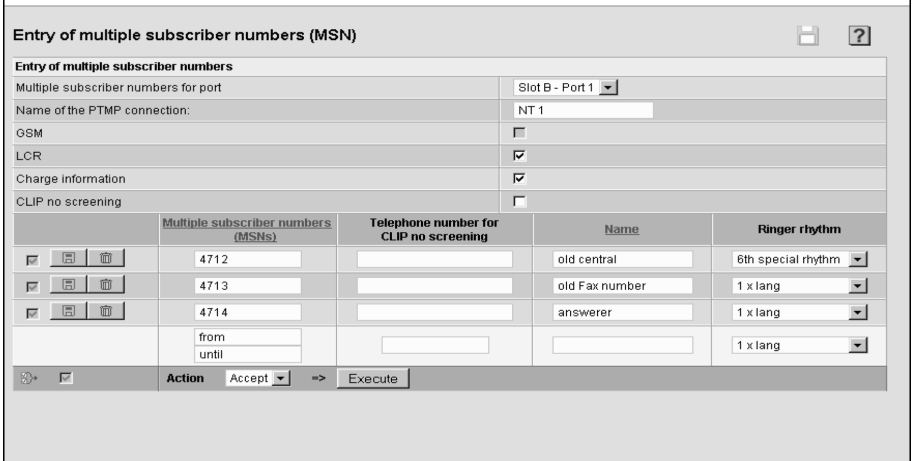
Under “***Name of the PTMP connection***”, enter any name with a maximum of 16 characters.

In the “***Multiple subscriber number (MSN)***” column under “***from:***”, enter your first available MSN.

Under “***Name***”, enter a suitable name with a maximum of 16 characters.

Under “***Ringer rhythm***”, select the desired ringer rhythm for external calls over this telephone number from the list field.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection*



Then click “***Execute***” to apply the settings to the PBX.

Configure the settings for all additional MSNs.

The Point-to-Multipoint connection is now configured.

Note: *If you have contiguous telephone number ranges, you can enter the beginning and end values under “**from: until**”. Only after clicking “**Execute**” will the name and ringer rhythm be available for entries.*

*A change under “**GSM**” and “**LCR**” is only necessary in exceptional cases (for example, when operating a GSM gateway) (see VoIP and GSM Routing (Exception Telephone Numbers) on page 89 and Least Cost Routing with Soft-LCR easy on page 84 and Least Cost Routing with Soft-LCR 4.0 on page 85).*

*Under “**Charge information**”, a change is only necessary if problems are caused by call charges (see Recording Call Data on page 62).*

*Under “**CLIP no screening**”, a change is only necessary if you would like to transfer special telephone numbers (see Customer-defined Telephone Number Information Presentation for “CLIP no Screening” on page 54).*

ISDN Point-to-Point Connection

In the PBX, configure each existing ISDN Point-to-Point connection with the telephone numbers provided by the network provider.

To do this, open the page displayed in the figure to the right.

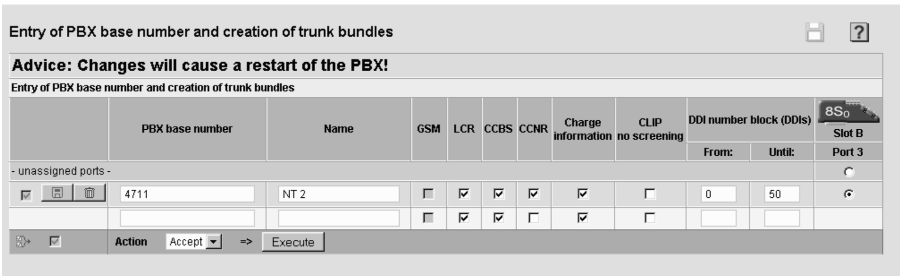
Under “***PBX base number***”, enter the PBX base number (without extension).

Under “***Name***”, enter any name with a maximum of 16 characters.

Under “***DDI number block (DDIs)***”, enter the extension range with the lowest and the highest DDI.

Then click “***Execute***” to apply the settings to the PBX.

The figure shows the following page with the settings for Example 1 on page 13:
COMset ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles*



The PTP connection is now configured.

In the right column, select the applicable port for the telephone number entered.

Note: *Make sure to follow the instructions provided by your network provider for entering the DDI number block exactly. If you want to enter a one or two-digit DDI range, your network provider must provide you with one and two-digit DDIs. The same applies to a two and three-digit DDI range.*

*A change under "**GSM**" and "**LCR**" is only necessary in exceptional cases (for example, when operating a GSM gateway) (see VoIP and GSM Routing (Exception Telephone Numbers) on page 89 and Least Cost Routing with Soft-LCR easy on page 84 and Least Cost Routing with Soft-LCR 4.0 on page 85).*

*By enabling **CCBS** and/or **CCNR** a callback on busy (CCBS) or on no reply (CCNR) can be offered to external callers if the service feature has been released by the network operator (see Automatic Callback on Busy (CCBS) on page 51 and Automatic Callback on no Reply (CCNR) on page 52).*

*Under "**Charge information**", a change is only necessary if problems are caused by call charges (see Recording Call Data on page 62).*

*Under "**CLIP no screening**", a change is only necessary if you would like to transfer special telephone numbers (see Customer-defined Telephone Number Information Presentation for "CLIP no Screening" on page 54).*

Voice-over-IP (VoIP)

Up to 100 accounts can be configured in the PBX. The PBX supports two different types of VoIP account:

- VoIP accounts with one or more VoIP phone numbers (similar the Point-to-Multipoint connection on ISDN)
- VoIP accounts with a DDI number block (similar to the -PBX connection on ISDN) based on the SIP-DDI feature (also known as SIP trunking)

To receive the necessary access data, accounts must first be set up with one or more VoIP providers. For this purpose, you need to register your name and address with a provider via their web site. Then a telephone number accessible from the land line network and the Internet as well as an account with a user name (is also known as the user name, authorization user and SIP ID) and password are assigned. Most of the time, the registered connection is set up within just a few minutes and can be used almost immediately.

Note: *In order to resolve the names of Internet addresses, the PBX needs (just like with a PC connected to the Internet) the address for a DNS server. This means that the address provided by your system administrator or by your Internet service provider needs to be entered. You may also enter a second address just in case the main DNS server is unavailable. If you have not configured this setting during the initial system set-up, you can now do it using the information found on the page Administration ▶ Server configuration.*

VoIP Provider

In the PBX, you can configure the VoIP providers with which you have accounts.

To do this, open the page displayed in the figure to the right.

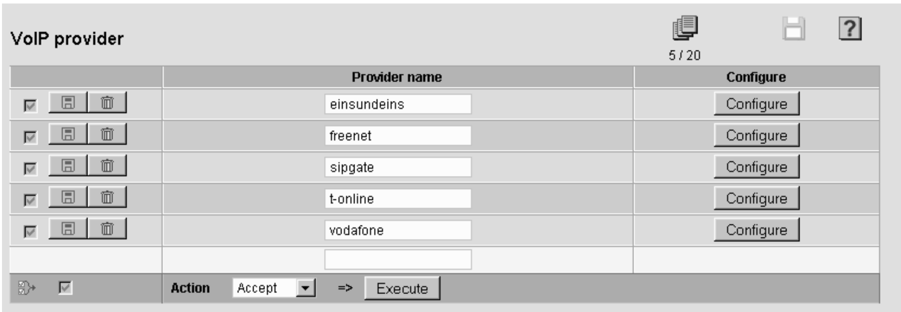
Frequently-used VoIP providers and their configurations are already provided as default providers in the system. If you would like to use them, activate them as follows.

Under "***Action***", activate the default providers and then click "***Execute***".

If your provider is not one of the default providers, configure your provider as follows.

Under "***Provider name***", enter any name with a maximum of 16 characters. Then click "***Execute***".

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ External numbers ▶ Voice over IP (VoIP) ▶ Provider



Click "***Configure***" *next to the names entered.*

Configure the settings for the provider. You will receive the data directly from the provider or over the corresponding lists on the Internet. For more information, also refer to the online Help *COMset ▶ External numbers ▶ Voice over IP (VoIP) ▶ Provider*.

Then click "***Execute***" to apply the settings to the PBX.

Then click "***Back***".

The provider is now configured.

If additional providers are required, enter additional names under "***Provider name***" and configure the settings for them.

Click the ***"Online configurations"*** button to download provider configurations supported by Auerswald from the Internet.

VoIP Accounts

In the PBX, configure the VoIP accounts you have available.

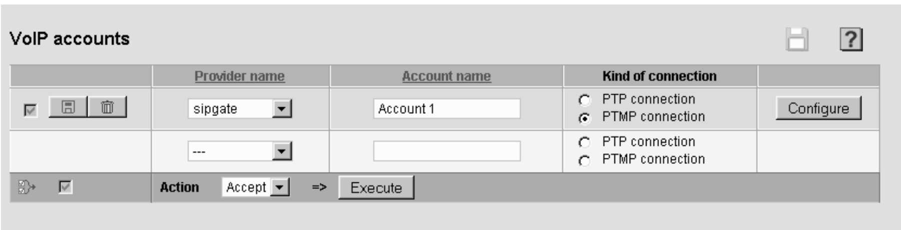
To do this, open the page displayed in the figure to the right.

Under "***Provider name***", select the desired account provider from the list field.

Under "***Account name***", enter any name with a maximum of 16 characters.

Under "***Kind of connection***" and depending on the available telephone numbers, select the kind of VoIP account from the list field.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ External numbers ▶ Voice over IP (VoIP) ▶ Accounts



For VoIP accounts with one or more VoIP phone numbers, select "***PTMP connection***".

For VoIP accounts with a DDI number block based on the SIP-DDI feature (also known as SIP trunking), select "***PTP connection***".

Click "***Execute***" to accept the settings in the PBX.

Click "***Configure***" to configure the VoIP account.

VoIP Point-to-Multipoint Connection

Under "***Exchange line access number (account number)***", enter an internal telephone number from your telephone number plan.

Under "***User name***" and "***Password***", enter the access data provided during registration.

In the column "***Multiple subscriber number (MSNs)***" under "***From:***", enter your first available MSN.

Under "***Display name***", enter any name with a maximum of 16 characters.

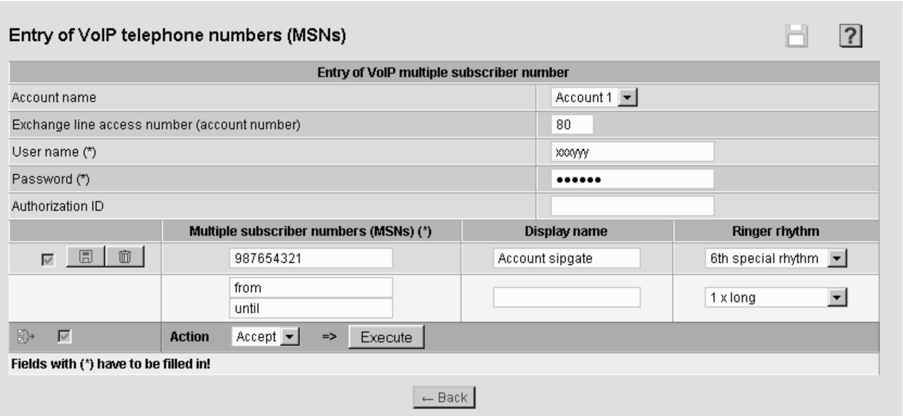
Under "***Ringer rhythm***", select the desired ringer rhythm for external calls via this telephone number from the list field.

Then click the "***Execute***" button to accept the settings to the PBX.

Configure the settings for any additional MSNs.

The account is now configured.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ External numbers ▶ Voice over IP (VoIP) ▶ Accounts ▶ Configure



If there are other accounts present, configure the settings for them as well.

Note: *If you have contiguous telephone number ranges, you can enter the beginning and end values under "**from: until:**". Only after clicking "**Execute**" will the name and ringer rhythm be available for entries.*

*When using the provider T-Online, the Internet phone numbers received from the provider must be entered under "**Display name**", "**User name**" and "**Multiple subscriber number (MSNs)**".*

VoIP Point-to-Point Connection

Under "***Exchange line access number (account number)***", enter an internal telephone number from your telephone number plan.

Under "***User name***" and "***Password***", enter the access data provided during registration.

Under "***PBX base number***", enter the PBX base number (without extension).

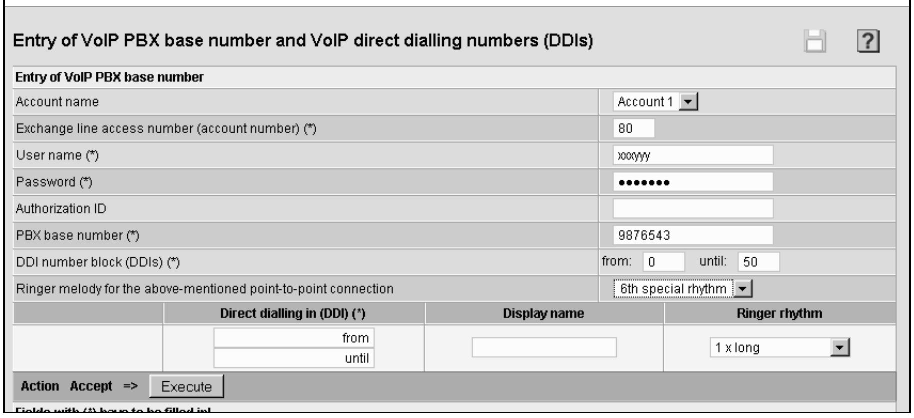
Under "***DDI number block (DDIs)***", enter the extension range with the lowest and the highest DDI.

Under "***Ringer melody for the above-mentioned point-to-point connection***", select the desired ringer rhythm from the list field for external calls via this telephone number.

Then click the "***Execute***" button to accept the settings to the PBX.

The account is now configured.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ External numbers ▶ Voice over IP (VoIP) ▶ Accounts ▶ Configure



If there are other accounts present, configure the settings for them as well.

Note: *Make sure to follow the instructions provided by your provider for entering the DDI number block exactly. If you want to enter a one or two-digit DDI range, your provider must provide you with one and two-digit DDIs. The same applies to a two and three-digit DDI range.*

Note: *DDIs will only have to be entered here if you want to create, for example, a display name, a ringer rhythm or a call forwarding for these DDIs.*

Call Distribution

Call distribution is required to establish a connection between the external telephone numbers already configured and the internal subscribers. (Question: Which internal subscriber rings when a call comes in on an external telephone number?).

It is a good idea to create a telephone numbering plan for the devices (see also Example 1 on page 13) and then use the following steps to transmit this to the PBX.

On a Point-to-Point connection (ISDN and VoIP), the configuration of a linear call distribution is usual (e. g., a call to 4711-21 is distributed to the internal subscriber telephone number 21). This includes having assigned internal telephone numbers that lie in the extension number range (DDIs) assigned by the network provider. Call distribution can then occur automatically.

Organizational, system-dependent or company internal guidelines can require diverging from linear call distribution in particular situations. To do this, the PBX gives you the option of configuring divergent call distribution. This configuration is done in two steps. In the first step, the PBX is notified which telephone numbers are to be taken out of the linear call distribution. In the second step, the telephone numbers are directly assigned internal subscribers.

When using a Point-to-Multipoint connection (ISDN and VoIP), the internal telephone numbers are individually assigned to the MSNs. Automatic call distribution is not possible here.

If multiple internal devices should be accessible over an external telephone number, an internal group, to which the subscribers in question are assigned, is selected as the destination for the call distribution.

Distributing Calls on the ISDN Point-to-Multipoint Connection

In the PBX, configure the associated internal telephone number for each registered MSN.

To do this, open the page displayed in the figure to the right.

Under "***Call distribution for ...***", select from the list field "***ISDN Point-to-multipoint connections***".

Under "***ISDN Point-to-multipoint connection***", select from the list field one of the already configured Point-to-Multipoint connections.

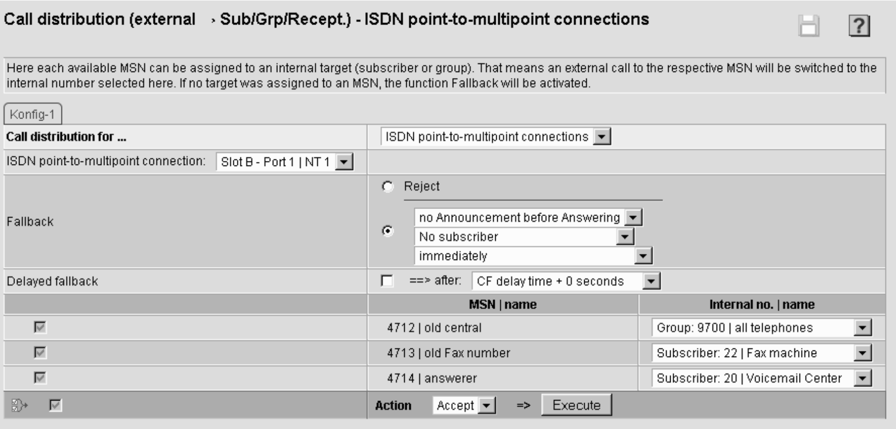
The MSNs for this connection are listed under "***MSN / name***".

Under "***Internal no. / name***", select an internal telephone number for each MSN from the list fields.

Then click "***Execute***" to apply the settings to the PBX.

This completes call distribution configuration on the ISDN Point-to-Multipoint connection.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ External numbers ▶ ISDN connections ▶ Call distribution



If additional ISDN Point-to-Multipoint connections are present, under "***ISDN Point-to-multipoint connection***", select the next Point-to-Multipoint connection from the list field and configure the settings for it.

Note: *For more information on this subject, refer to* *External Call Distribution to Subscribers, Groups and Voice Mail/Fax Boxes on page 43**.*

Call Distribution on an ISDN Point-to-Point Connection

In the PBX, configure the linear call distribution as well as the call distribution that deviates from it for the extension range entered.

To do this, open the page displayed in the figure to the right.

Under "***Call distribution for ...***", select from the list field "***ISDN Point-to-point connections***".

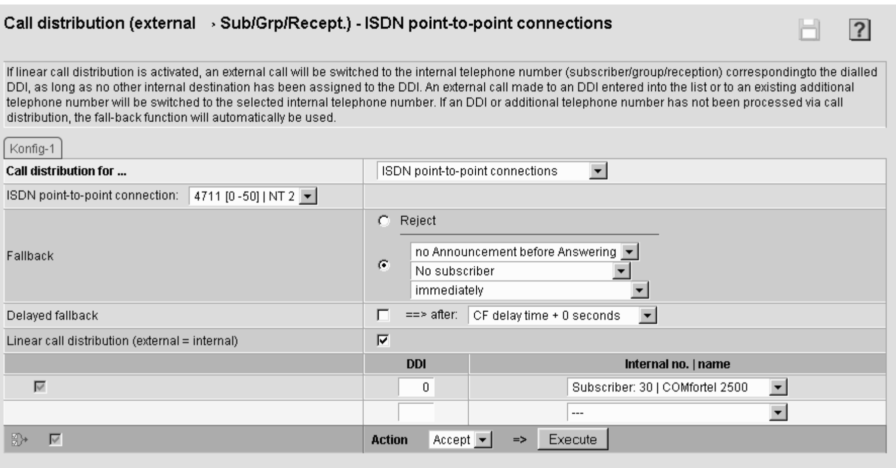
Under "***ISDN Point-to-point connection***", select one of the Point-to-Point connections already configured from the list field.

Under "***Linear call distribution (external = internal)***", activate linear call distribution (it is active by default).

Under "***DDI***", enter – if present – one of the DDIs which deviates from the linear call distribution.

Under "***Internal no. / name***", select an internal telephone number for the registered DDI from the list field.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ External numbers ▶ ISDN connections ▶ Call distribution



Then click “***Execute***” to apply the settings to the PBX.

Enter additional DDIs that deviate from the linear call distribution and select the associated internal telephone numbers.

Note: *Telephone numbers that diverge from the linear call distribution must be found in the DDI number block entered.*

For more information on this subject, refer to External Call Distribution to Subscribers, Groups and Voice Mail/Fax Boxes on page 43.

Call Distribution on VoIP Point-to-Multipoint Connections

On the PBX, configure the associated internal telephone number for each MSN entered.

To do this, open the page displayed in the figure to the right.

Under “***Call distribution for ...***”, select “***VoIP point-to-multipoint connections***” from the list field.

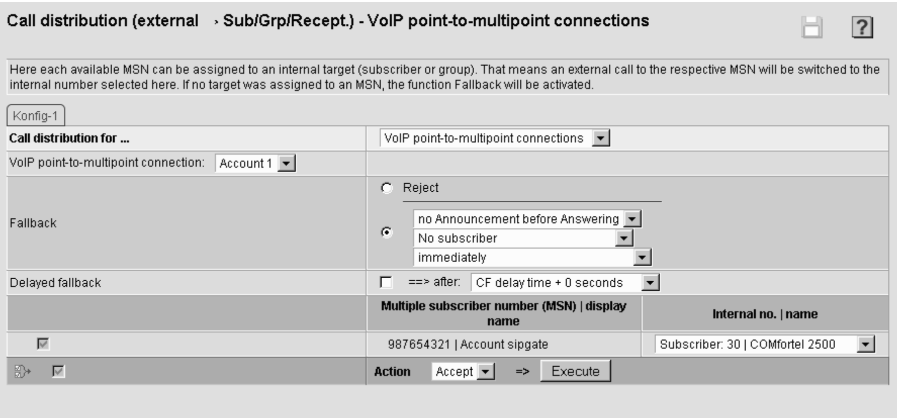
Under “***VoIP point-to-multipoint connection***”, select one of the VoIP accounts with Point-to-Multipoint connection from the list field that has already been configured.

Under “***Multiple Subscriber Number (MSN) / Display Name***”, you will find a list of the MSNs available for this VoIP account.

Under “***Internal no. / name***”, select an internal telephone number for each MSN from the list fields.

Click “***Execute***” to accept the settings in the PBX.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ External numbers ▶ Voice over IP (VoIP) ▶ Call distribution



This completes call distribution configuration on the VoIP Point-to-Multipoint connection.

If other VoIP accounts with a Point-to-Multipoint connection are available, under “***VoIP point-to-multipoint connection***”, select the next VoIP account from the list field and configure the settings for it.

Note: *For more information on this subject, refer to External Call Distribution to Subscribers, Groups and Voice Mail/Fax Boxes on page 43.*

Call Distribution on VoIP Point-to-Point Connections

In the PBX, configure the linear call distribution as well as the call distribution that deviates from it for the extension range entered.

To do this, open the page displayed in the figure to the right.

Under “***Call distribution for ...***”, select “***VoIP point-to-point connections***” from the list field.

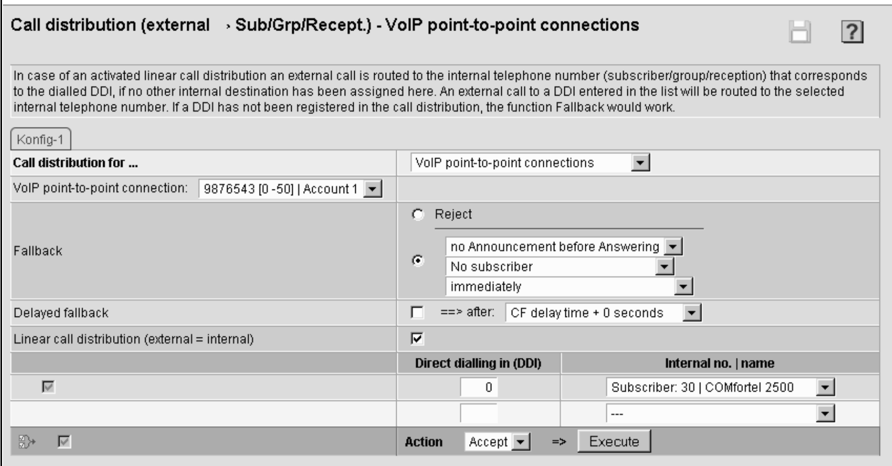
Under “***VoIP point-to-point connection***”, select one of the VoIP accounts with a Point-to-Point connection from the list field that has already been configured.

Under “***Linear call distribution (external = internal)***”, enable linear call distribution (it is enabled by default).

Under “***DDI***”, enter – if available – one of the DDIs which deviates from the linear call distribution.

Under “***Internal no. / name***”, select an internal telephone number for the registered DDI from the list field.

The figure shows you the following page:
COMset ▶ External numbers ▶ Voice over IP (VoIP) ▶ Call distribution



Click “***Execute***” to accept the settings in the PBX.

Enter additional DDIs that deviate from the linear call distribution and select the associated internal telephone numbers.

Note: *Telephone numbers that diverge from the linear call distribution must be found in the DDI number block entered.*

For more information on this subject, refer to External Call Distribution to Subscribers, Groups and Voice Mail/Fax Boxes on page 43.

End of Basic Configuration Based on Example 1

The basic configuration of Example 1 on page 13 is now completed.

In order to get an overview of the telephone numbers set and their distribution in the PBX, there are several helpful views described at the end.

For more information on how to best check the function of the configured settings, refer to Commissioning and testing your PBX on page 27.

Internal Subscriber Overview

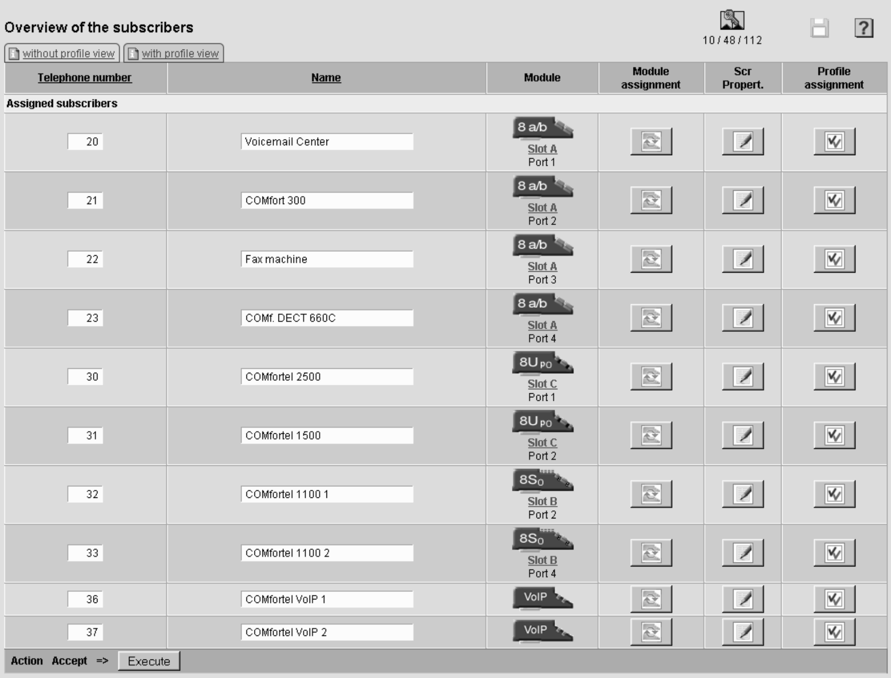
Check the internal subscribers configured in your PBX.

To do this, open the page displayed in the figure to the right.

Note: *You can use the printing function in your browser to print out this overview.*

It is advisable to print the page in landscape format.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Internal numbers ▶ Subscriber (scr) ▶ Overview



Internal Group Overview

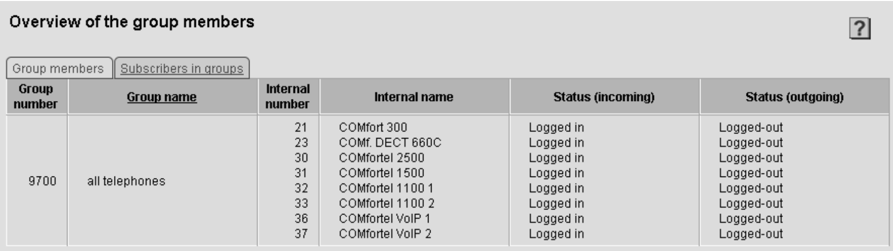
Check the internal groups configured in your PBX.

To do this, open the page displayed in the figure to the right.

Note: *You can use the printing function in your browser to print out this overview.*

It is advisable to print the page in landscape format.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ Internal numbers ▶ Groups ▶ Overview



Call Distribution Overview

Check the assignment of external to internal telephone numbers (call distribution).

To do this, open the page displayed in the figure to the right.

Note: *You can use the printing function in your browser to print out this overview.*

It is advisable to print the page in landscape format.

The figure shows you the following page with the settings for Example 1 on page 13:
COMset ▶ *External numbers* ▶ *Overview*

Overview of the call distribution

External number	DDI			Konfig-1		
	from	until		Telephone number	Name	Type
PTP connections						
4711 - NT 2	0	50	0 -	30	COMfortel 2500	Subscriber
			20 -	20	Voicemail Center	Subscriber
			21 -	21	COMfort 300	Subscriber
			22 -	22	Fax machine	Subscriber
			23 -	23	COMf. DECT 660C	Subscriber
			30 -	30	COMfortel 2500	Subscriber
			31 -	31	COMfortel 1500	Subscriber
			32 -	32	COMfortel 1100 1	Subscriber
			33 -	33	COMfortel 1100 2	Subscriber
			36 -	36	COMfortel VoIP 1	Subscriber
			37 -	37	COMfortel VoIP 2	Subscriber
			50 -	---		---
			all other DDIs	---		---
PTMP connections						
4712 - old central				9700	all telephones	Group
4713 - old Fax number				22	Fax machine	Subscriber
4714 - answerer				20	Voicemail Center	Subscriber
VoIP point-to-point connections						
VoIP point-to-multipoint connections						
987654321 - Account sipgate				30	COMfortel 2500	Subscriber

Commissioning and testing your PBX

After basic configuration is completed, the PBX can be put into operation and the connected devices can be tested for function. The tests described in the following apply to all the telephone numbers and devices connected to your PBX.

Important: *The prerequisite is that your PBX, as described in the Installation and Commissioning Instructions, has been installed and all the devices properly connected under the supervision of a professional and in compliance with the safety instructions.*

Note: *The tests described are only a few of the options you have for performing general testing on your PBX. These instructions cannot replace a complete initial start-up procedure.*

Note that for standard VoIP telephones, a dial tone is generated by the telephone itself and therefore can not indicate whether calling is possible.

Checking Telephone Connections and Internal Telephone Numbers

1. Pick up the receiver on an internal telephone.
2. You hear the internal dial tone. This signal tells you that you can now start dialling a number.
3. Dial the internal telephone number for a neighboring telephone.
4. The telephone called rings.
5. Hang up the receiver.

If you do not hear an internal dial tone on individual telephones, ...

... Check the line between the telephone and the wall socket or the telephone itself.

... Check whether internal telephone numbers have been defined for the telephones in question (see Internal Subscriber Overview on page 25).

... Check whether the internal telephone number defined for the telephone has been entered as the first MSN in the telephone (only ISDN and system telephones). Note that for ISDN telephones, this MSN must also be transmitted from the telephone to the -PBX (see the manual for the telephone).

... Disconnect the telephone in question for approx. 5 seconds from the PBX- and, if necessary, the 230-V- power supply (pull the power plug).

If you do not hear an internal dial tone on any of the telephones, ...

... Check whether the PBX- power plug is properly inserted into the power socket.

... Disconnect the PBX- for approx. 5 seconds from the 230-V- power supply (pull the power plug).

If the telephone called does not ring, ...

... Check whether the internal telephone number called has been assigned to the telephone in question (see Internal Subscriber Overview on page 25).

... Check whether the telephone bell is switched off (see the manual for the telephone).

... Check the registration status and, if necessary, the access data on VoIP telephones.

If the connection is immediately disconnected after picking up the VoIP telephone being called or no calling is possible, ...

... check the codec configured in the telephone and change it, if possible. The PBX supports the following codecs: G.711 and iLBC.

If no incoming call on a VoIP telephone is possible, although it has been registered, ...

... check the codec configured in the telephone and change it, if possible. The PBX supports the following codecs: G.711 and iLBC.

Checking External Connections

1. Pick up the receiver on an internal telephone.
2. You hear the internal dial tone. This signal tells you that you can now start dialling a number.
3. Dial the exchange line access number.
4. You hear the external dial tone. This signal tells you that an external line is available for dialling an external telephone number.
5. Dial the external telephone number for a neighboring internal telephone.
6. The telephone called rings.
7. Hang up the receiver.
8. If the telephone should be accessible over additional external telephone numbers, repeat the procedure for all of the additional external telephone numbers.

If the telephone called does not ring, ...

... Check whether the external telephone number called has been assigned to the telephone in question (see Call Distribution Overview on page 26).

If you do not hear an external dial tone, ...

... Check the function of the NT by connecting a single ISDN telephone to the NT (for an NT with connection type -Point-to-Point connection, the ISDN telephone must be compatible for operating on the -Point-to-Point connection). If it is still not possible to make a call, disconnect the NT for approx. 5 seconds from the 230-V- power supply (pull the power plug) ***and*** *the exchange line connection (pull the TAE connector). If you can now make outbound calls again, this means that the NT was disrupted. If the malfunction remains, notify your line fault service.*

If none of the telephones that are accessible over an external connection (NT) ring, ...

... Disconnect the NT for approx. 5 seconds from the 230-V- power supply (pull the power plug) ***and*** *the exchange line connection (pull the TAE connector). If you can now make outbound calls again, this means that the NT was disrupted. If the malfunction remains, notify your line fault service.*

Example 2: Adding a COMmander 2TSM analog Module

In the following example (see figure), Example 1 on page 13, a *COMmander 2TSM analog module* is added to the PX.

When this is done, the *COMmander 2TSM analog module* should carry out several functions that are frequently used.

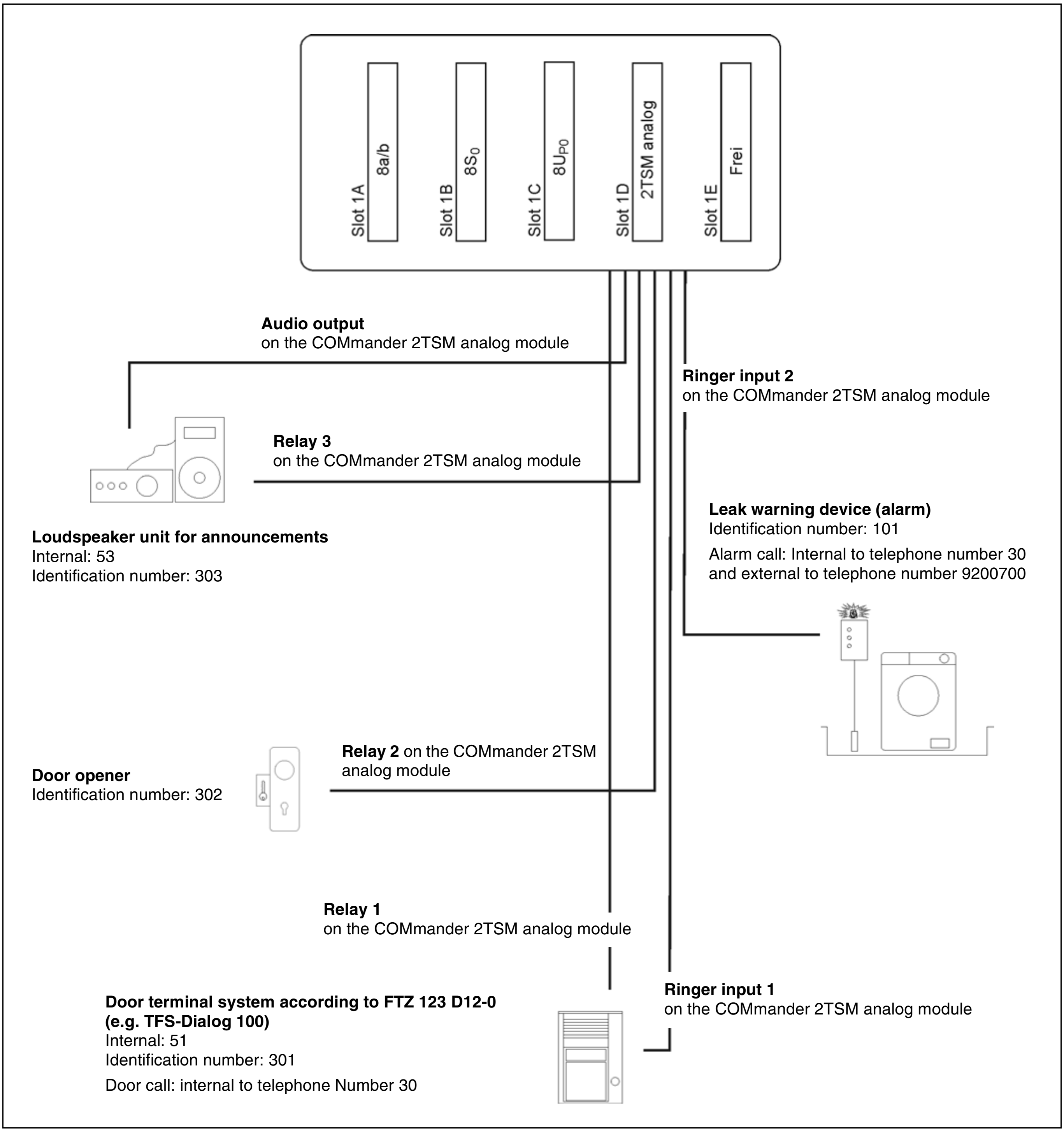
All the necessary configuration settings are explained step-by-step.

***Important:** When installing a COMmander 2TSM analog module, note the description for converting or upgrading the system found in the Installation and Commissioning Instructions.*

***Note:** Note that this section thoroughly describes only the most important settings. You will be referred to the corresponding sections for additional settings.*

The combination of the devices listed here is an example only and may vary greatly from your PBX.

Analog door terminal systems (e.g. TFS-Dialog 200, TFS-Dialog 300 and TFS-Universal a/b) are not connected via a COMmander 2TSM analog module but - like an analog telephone - to an internal analog a/b port of the PBX. The configuration is done on the door terminal system itself, including door terminal call distribution to internal subscriber and group telephone numbers. For more information on this, please refer to the instructions of the door terminal system.



Logging into the Configuration Manager

Important: *To be able to perform basic configuration, the PBX and your computer must be correctly connected with each other.*
The settings for this are described in the Installation and Commissioning Instructions and must be noted when initially operating the PBX.

Start the configuration manager. Enter the user name “***admin***” and the 6-digit admin PIN and confirm by clicking “***Log-in***”.

You must logged on as administrator in order to perform the configuration because initially only the administrator has the necessary rights. After you have logged in, you see a menu with a tree structure on the left side of the page. You now have access to the entire system configuration. The structure is similar to the directory structure on a hard drive where each folder can be described with a unique path.

For reasons of simplicity, this **path information** is shown above each figure for each configuration example. This shows the exact location of each menu option in the tree structure.

Hardware Configuration

Assign the current modules to the expansion slots on the mainboard.

To do this, open the page displayed in the figure to the right.

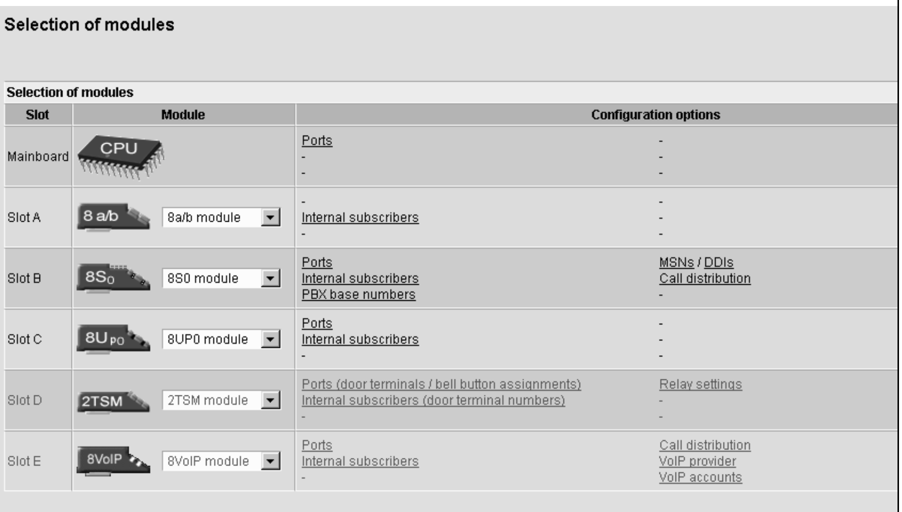
The “***Slot***” column lists each module slot on the mainboard (slot 1A to 1E).

Under “***Module***”, select a module from the list field that has been inserted in the associated module slot. For available module slots, select “not defined” from the list field.

In the “***Configuration options***” column, you can use the hyperlinks to directly access the settings described at the end.

These settings complete the hardware configuration.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Hardware ▶ Selection of modules



Selection of modules

Selection of modules

Slot	Module	Configuration options	
Mainboard	CPU	Ports - -	- - -
Slot A	8 a/b — 8a/b module	- Internal subscribers -	- - -
Slot B	8S0 — 8S0 module	Ports Internal subscribers PBX base numbers	MSNs / DDIs Call distribution -
Slot C	8UP0 — 8UP0 module	Ports Internal subscribers -	- - -
Slot D	2TSM — 2TSM module	Ports (door terminals / bell button assignments) Internal subscribers (door terminal numbers) -	Relay settings - -
Slot E	8VoIP — 8VoIP module	Ports Internal subscribers -	Call distribution VoIP provider VoIP accounts

Note: *Note that parameters set here correspond to the real hardware settings on your PBX (question: Which module is inserted in which slot?). If this is the case, the settings are marked in black. If this is not the case, the settings are marked in red.*

For more information on this, refer to the module descriptions in the Installation and Commissioning Instructions.

Configuring Ringer/Alarm Inputs

Configure the ringer inputs on the COMmander 2TSM analog module.

To do this, open the page displayed in the figure to the right.

Under "***Name***", enter a suitable name with a maximum of 16 characters.

Under "***Input***", select between the two functions ringer input for the door terminal and alarm input from the list field.

Under "***Type of contact***", select the level according to your connections from the list field. The setting "*High level*" switches the input connection to +12 Volts for signalling. The setting "*Low level*" switches GND on for signalling.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Port configuration

2TSM **Port configuration - 2TSM**

Module	Door bell button	Name	Input	Type of contact
2TSM Slot 1D	Bell 1	Door bell	Door terminal 1	High level
	Bell 2	Alarm contact	Alarm	High level
	Bell 3		Door terminal 1	High level
	Bell 4		Door terminal 1	High level
Action Accept => Execute				

Then click "***Execute***" to apply the settings to the PBX.

This completes the configuration for the ringer input for the door terminal and for the alarm input.

Important: *Depending on the type of door bell contact assignment and door terminal system, you need to configure the correct level setting under Type of contact. Various connections are described in the Installation and Commissioning Instructions.*

Internal Telephone Numbers

In order for the door terminal and the audio output to be reachable, both internal telephone numbers must be configured.

It is a good idea to create a telephone numbering plan for the devices (see also Example 2 on page 28) and then use the following steps to transmit this to the PBX.

Note: *The PBX lets you assign internal telephone numbers 10-9999. Of this number range 10-9999, the following telephone numbers are assigned:*
- Exchange line access numbers (account numbers) in case of VoIP
- Internal subscriber telephone numbers
- Internal group telephone numbers
- Internal CAPI dial-in numbers
- Internal telephone numbers for automatic switchboards
- Internal basis telephone numbers for open callbacks
- Internal telephone numbers for door terminals
- Internal telephone numbers for voice mail/fax boxes
- Internal telephone numbers for audio outputs
- Short-code numbers
- Emergency numbers

Double assigning a number is not possible!
On the page COMset ▶ Internal numbers ▶ Telephone numbering plan, you can get an overview of the internal telephone numbers already assigned at any time.

The telephone numbers can consist of two to four digits. Using telephone numbers with few digits (two or three-digit numbers) inevitably restricts the possible supply of telephone numbers with more digits. For example, if the telephone number 12 is assigned, the telephone numbers 120-129 and 1200-1299 are no longer available.

For some functions, only telephone numbers with a maximum of 3-digits can be assigned because the last digit is need for the function (e. g., door terminal numbers, telephone numbers for open callbacks).

Telephone Number for the Audio Output

Configure an internal telephone number for calling the audio output to make an announcement.

To do this, open the page displayed in the figure to the right.

Under "***Telephone Number for the audio output (Audio Out)***", enter a 2 to 4-digit telephone number from your internal telephone number plan.

Under "***Name***", enter a suitable name with a maximum of 16 characters.

Then click "***Execute***" to apply the settings to the PBX.

This completes the telephone number assignment for the audio output.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Telephone numbers

2TSM **- Subscriber - 2TSM analog**

	Door terminal	Telephone number for door terminal	Name
2TSM Slot 1D	1	51	Door 1
	2		
		Telephone number for the audio output (Audio Out)	**Name**
	Telephone number	53	Loudspeaker
	Second ringer (configuration dependent by corresponding call distribution)		**Property**
	activated for subscriber/group		---
	active when pressing ringer 1 ("Door bell")		☐
	Alarm input		
	active when pressing ringer 3		☐
	active when pressing ringer 4		☐
Action Accept => Execute			

Important: *Note that for loudspeaker announcements, authorization must be assigned.*

Telephone Number for the Door Terminal

To do this, open the page displayed in the figure to the right.

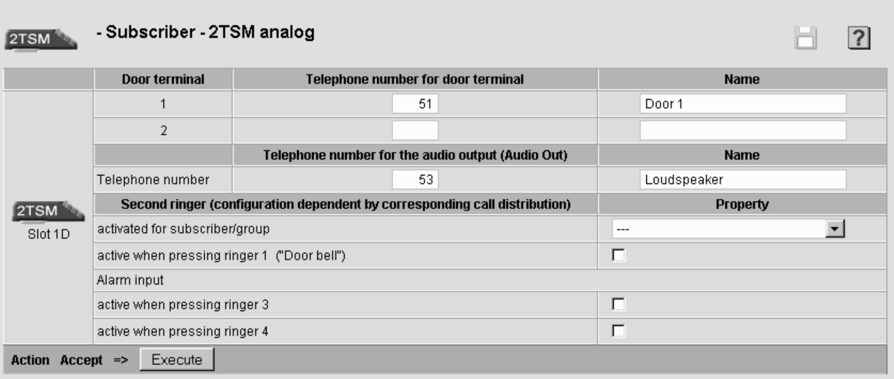
Under "***Telephone number for door terminal***", enter a 2 or 3-digit telephone number from your internal telephone number plan.

Under "***Name***", enter a suitable name with a maximum of 16 characters.

Then click "***Execute***" to apply the settings to the PBX.

This completes the telephone number assignment for the door terminal.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Telephone numbers



Door Terminal Call Distribution and Door Bells

When the door bell rings, both internal and external subscribers as well as groups can be signalled. You can assign a separate target number for each door bell button for this.

In order for the person being called to recognize that a call is coming over a door bell button, telephone number detection can be configured for external destination numbers. For internal destination numbers, the internal telephone number is transmitted to the door + door bell number.

In addition for internal door terminal call distribution, an individual ringer rhythm can be set for each door bell button.

Furthermore, it is possible to configure the signalling to 5-30 seconds to ensure the signalling lasts a specific amount of time on the telephone.

Call Distribution

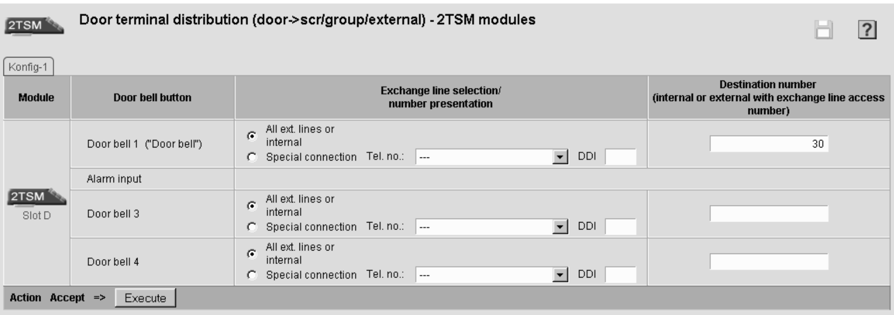
Configure which telephone number should be called when the door bell button is pressed.

To do this, open the page displayed in the figure to the right.

Under "***Destination number (internal or external with exchange line access number)***", enter the internal and external telephone number that is to be called when the door bell button is pressed.

If you have entered an external telephone number and would like to have a specific telephone number presented to the person being called, under "***Exchange line selection/ number presentation***", select a specific connection based on the telephone number of the person being called.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Call distribution



Then click "***Execute***" to apply the settings to the PBX.

This completes the configuration of door terminal call distribution.

Note: *For more information about the settings, refer to the chapter* *Door Call Distribution on page 44.*

Ringer Rhythms and Cadence

Configure the ringer rhythm and ringing time on the door bell button for internal door terminal call distribution.

To do this, open the page displayed in the figure to the right.

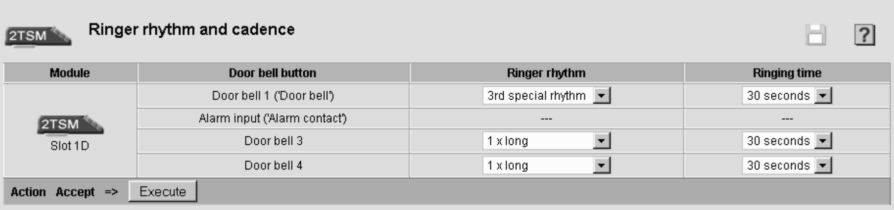
Under "***Ringer rhythm***", select the desired ringer rhythm from the list field.

Under "***Ringing time***", select the desired ringing time from the list field.

Then click "***Execute***" to apply the settings to the PBX.

This sets the ringer rhythms and durations.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Rhythm and cadence



Note: *For ISDN and VoIP subscribers, the function of this setting is dependent on the standard telephone connected. Standard telephones frequently support only their own ringer rhythms and not those programmed in the system.*

Configuring Relays

In order to perform various switching and control procedures, the + COMmander 2TSM analog module has a total of three relays that must be configured according to each intended use. Two of these relays come with factory default settings for controlling the door station. If no door station is being operated, these relays can be reconfigured for other uses.

Note: *The PBX requires 2 to 4-digit ID numbers for uniquely assigning the configured relays, alarms and configurations. These numbers can be freely assigned in the 10-9999 range.*
Double assigning a number is not possible!
On the page COMset ▶ Internal numbers ▶ Ident. numbering plan, you can get an overview of the ID numbers already assigned at any time.

Door Terminal Function

On Relay 1, configure the door terminal function.

To do this, open the page displayed in the figure to the right.

Under "***Relay mode***", select the desired operational mode (in this case, the door terminal function) from the list field.

Under "***Identification number for this relay***", select a 2 to 4-digit number.

Under "***Name for the identification number***", enter a suitable name with a maximum of 16 characters.

Then click "***Execute***" to apply the settings to the PBX.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Relay settings ▶ Slot ... TSM ▶ Relay 1

Note: *When doing this, note how the connections are described in the Installation and Commissioning Instructions.*

Relay Settings for the Door Opener

On Relay 2, configure the door opener function.

To do this, open the page displayed in the figure to the right.

Under "***Relay mode***", select the desired operational mode (in this case, the door opener function) from the list field.

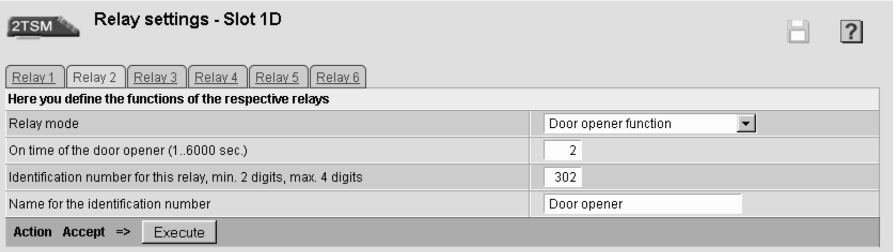
Under "***On time of the door opener (1...6000 sec.)***", enter the time that is required to open your door safely.

Under "***Identification number for this relay***", select a 2 to 4-digit number.

Under "***Name for the identification number***", enter a suitable name with a maximum of 16 characters.

Then click "***Execute***" to apply the settings to the PBX.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Relay settings ▶ Slot ... TSM ▶ Relay 2



Note: *When doing this, note how the connections are described in the Installation and Commissioning Instructions.*

Relay Settings for the Audio Output

On Relay 3, configure the announcement.

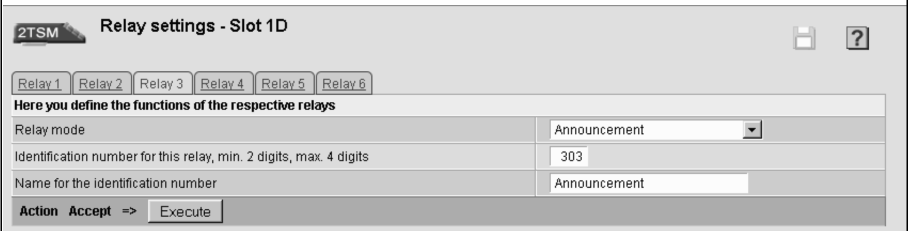
To do this, open the page displayed in the figure to the right.

Under "***Relay mode***", select the desired operational mode (in this case, the announcement) from the list field.

Under "***Identification number for this relay***", select a 2 to 4-digit number.

Under "***Name for the identification number***", enter a suitable name with a maximum of 16 characters.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Relay settings ▶ Slot ... TSM ▶ Relay 3



Then click "***Execute***" to apply the settings to the PBX.

This completes the relay setting configuration.

Note: *When doing this, note how the connections are described in the Installation and Commissioning Instructions.*

Configuring the Alarm Input

The PBX lets you process status, alarm or monitor signals and to activate them according to the current configuration. You can use any door bell button inputs still available for alarm inputs (see Configuring Ringer/Alarm Inputs on page 30). If an alarm is activated, both internal and external subscribers can be called by the PBX as well as sirens and flashing beacons turned on.

Important: *Note that the alarm functions of the PBX do not satisfy alarm system safety requirements for protection of life and limb. However, the PBX is well equipped for simple monitoring procedures that do not require strict security.*

Note: *The PBX requires 2 to 4-digit ID numbers for uniquely assigning the configured relays, alarms and configurations. These numbers can be freely assigned in the 10-9999 range.*
Double assigning a number is not possible!
On the page COMset ▶ Internal numbers ▶ Ident. numbering plan, you can get an overview of the ID numbers already assigned at any time.

Alarm Settings

Configure the alarm.

To do this, open the page displayed in the figure to the right.

Under "***Activate alarm***", select from various activation options from the list field.

Under "***Number of alarm loops***", select the desired number of alarm runs from the list field.

Under "***Alarm announcement***", select the desired alarm announcement from the list field.

Assign to each "***Alarm subscriber (internal or external with exchange line access number)***" the internal and/or external telephone numbers that are to be sequentially called when an alarm has been triggered.

If you have entered an external telephone number and would like to have a specific telephone number presented to the person being called, under "***Exchange line selection/ number presentation***", select a specific connection based on the telephone number of the person being called.

Under "***Alarm delay time in seconds (0..99)***", enter the time between alarm is triggered and the start of the first alarm run. Within this time, you can deactivate the alarm (for example, for a false alarm) using an internal telephone.

Under "***Alarm waiting time in seconds (0..99)***", enter the time between two successive alarm runs.

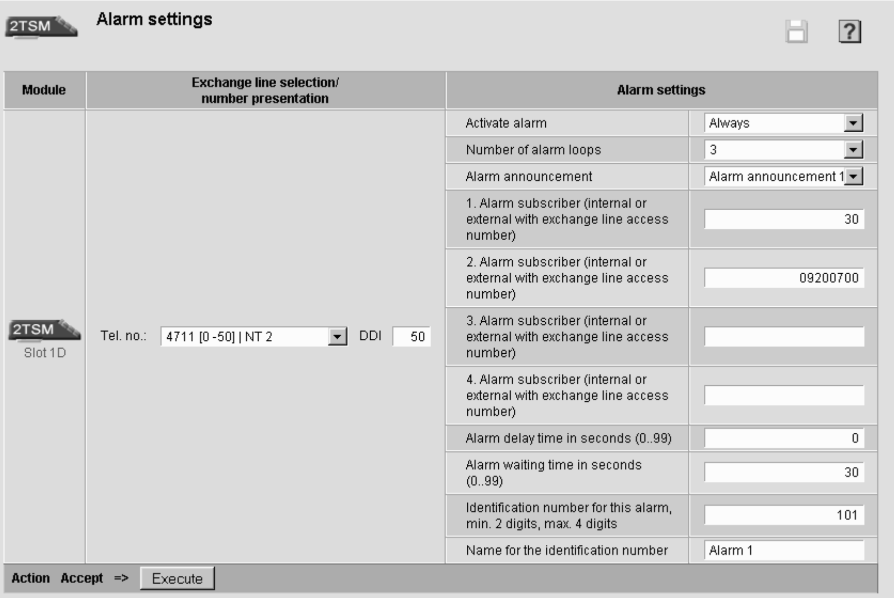
Under "***Identification number for this alarm***", select a 2 to 4-digit number.

Under "***Name for the identification number***", enter a suitable name with a maximum of 16 characters.

Then click "***Execute***" to apply the settings to the PBX.

This completes the configuration of the alarm settings.

The figure shows you the following page with the settings for Example 2 on page 28:
COMset ▶ Door/switching/music module ▶ Alarm settings



Example 3: Time-dependent Configurations (Day/Night Switching)

In the following example (see figure), two configurations are added to the PBX in Example 2 on page 28.

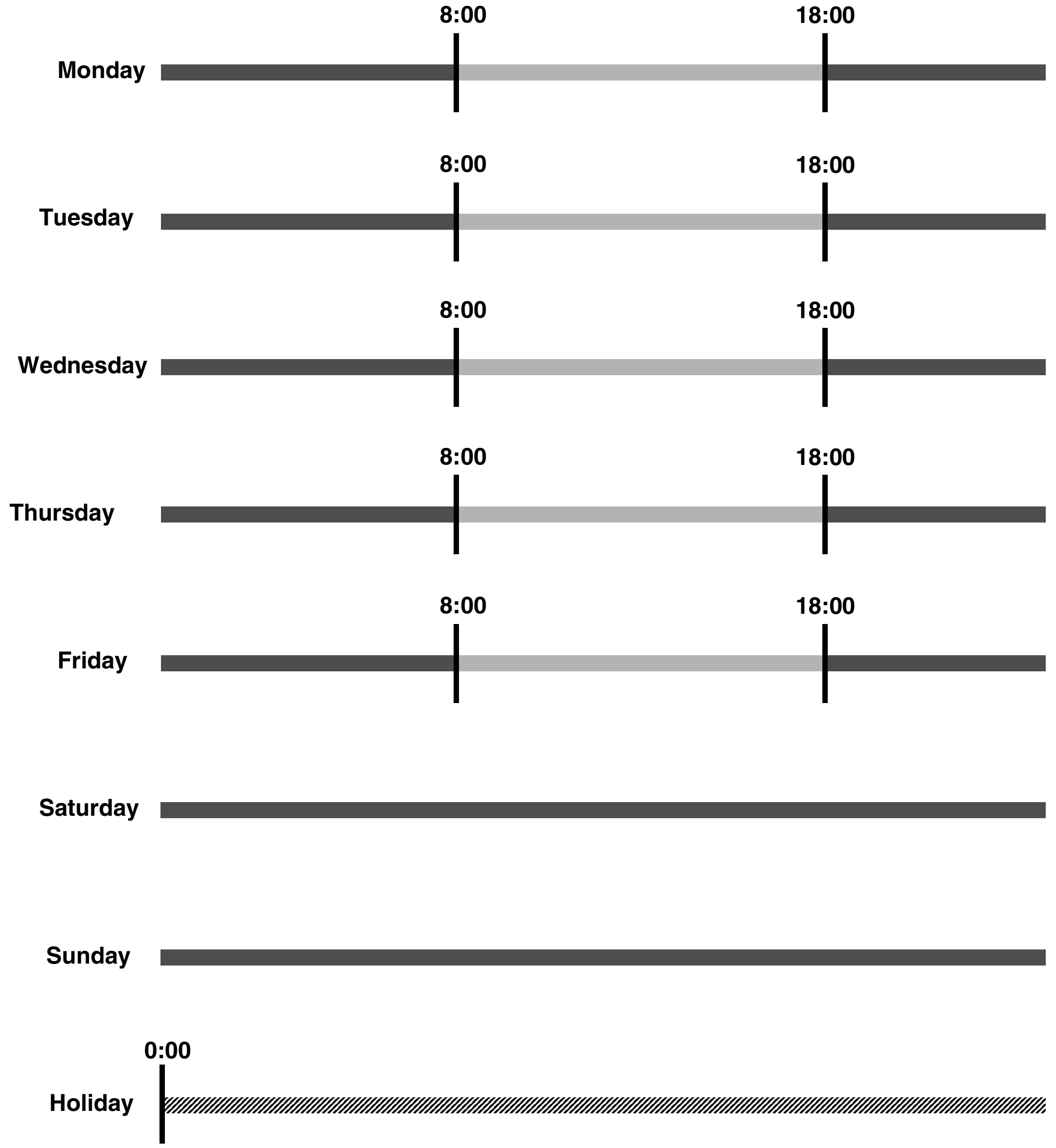
All the necessary configuration settings are explained step-by-step.

Note: *Note that this section thoroughly describes only the most important settings. You will be referred to the corresponding sections for additional settings.*

Configuration 1
Name: Business hours
Identification number: 201
Call distribution: as in Example 1 on page 13
Door call: as in Example 2 on page 28

Configuration 2
Name: Night-weekend
Identification number: 202
Call distribution: as Configuration 1 except 4711-0 and 4712 to 20 (Voicemail Center 461.2)
Door call: external to 0160 99999 (mobile phone, facilities manager)

Configuration 3
Name: Holiday
Identification number: 203
Call distribution: as Configuration 1 except 4711-0 and 4712 to 20 (Voicemail Center 461.2)



Creating Configurations and Time-dependent Switching

The PBX gives you the option of customizing various function configurations, for example, day, night, weekend, vacation and holidays settings. There are eight configurations available.

The switch from one configuration to another can be made according to a schedule based on PBX system time or manually from internal or external subscribers (telephones with the corresponding authorization).

Note: *The PBX requires 2 to 4-digit ID numbers for uniquely assigning the configured relays, alarms and configurations. These numbers can be freely assigned in the 10-9999 range.*
Double assigning a number is not possible!
On the page COMset ▶ Internal numbers ▶ Ident. numbering plan, you can get an overview of the ID numbers already assigned at any time.

Creating Configurations

Create the configurations and activate automatic configuration switching.

To do this, open the page displayed in the figure to the right.

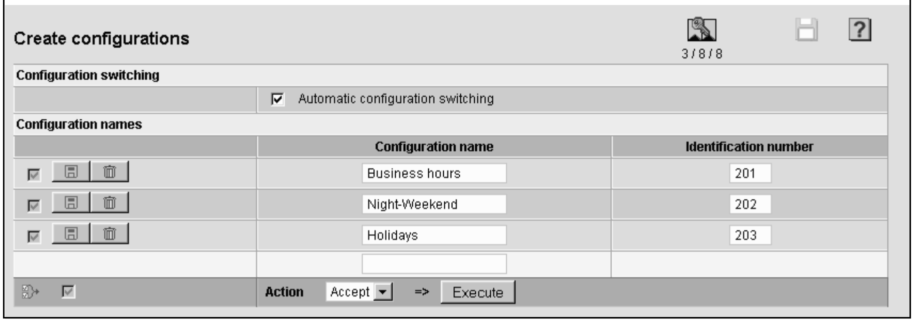
Under "***Configuration names***", enter a suitable name with a maximum of 16 characters in both lines.

Then click "***Execute***" to apply the settings to the PBX.

If you would like to create more than two configurations, enter additional names.

If necessary under "***Identification number***", enter a new 2 to 4-digit number for each configuration (a default number is automatically assigned during configuration).

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ Configurations ▶ Create configurations



Under "***Configuration switching***", activate automatic configuration switching.

Then click "***Execute***" to apply the settings to the PBX.

This completes the configurations.

Copying Configurations

In order for the new configurations to reach a defined state, first copy all of the settings for the first, already completed configuration to the newly created configurations.

To do this, open the page displayed in the figure to the right.

Under "***Source***", select the already completed configuration from the list field.

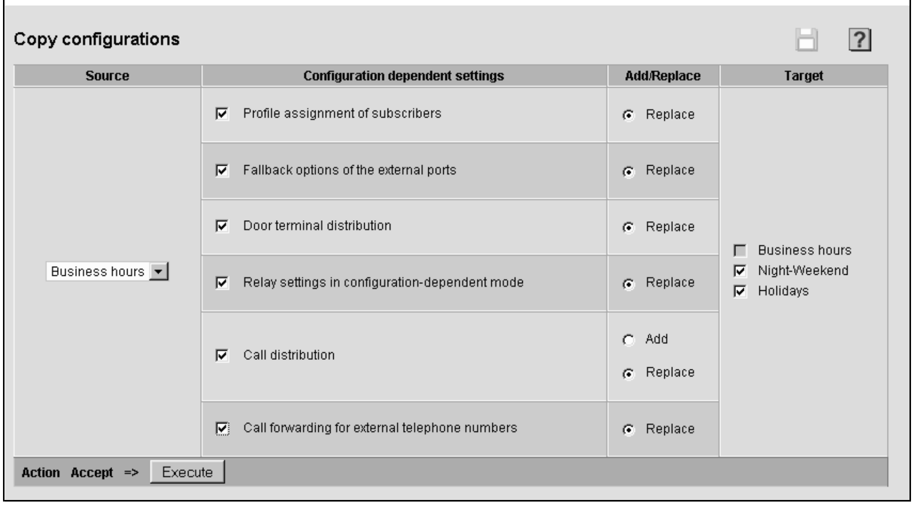
Under "***Configuration dependent settings***", activate the settings to be copied.

Under "***Target***", activate the newly created configurations.

Then click "***Execute***" to apply the settings to the PBX.

The new configurations now have a defined state.

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ Configurations ▶ Copy configurations



Configuring Switching Times

On the individual weekday cards, set the switching times using the configurations to be activated.

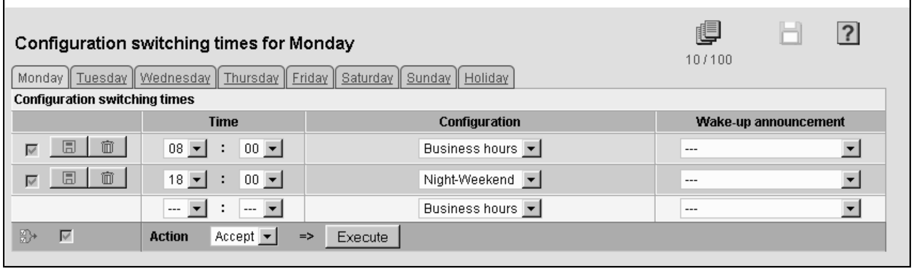
To do this, open the page displayed in the figure to the right.

Under "***Time***", select the first switching time from the list fields.

Under "***Configuration***", select the configuration that should be activated for the switching time from the list field.

Then click "***Execute***" to apply the settings to the PBX.

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ Configurations ▶ Switching times ▶ Monday



Configure the settings for all additional switching times on this weekday.

This completes the configuration for the configuration switching times for Monday.

Then configure the settings for the switching times for the other weekdays.

Note: *If the same switching times are required for more than one weekday, it is sufficient to set them up for only one day. The switching times can be copied to other weekdays.*

Only a few changes need to be entered. The configuration that was last valid on the previous day is always transmitted to the next day. Therefore, it is not necessary to activate a configuration at midnight if the configuration had already been activated on the previous day or also on other days in the past.

Exception: If the "normal" procedure is interrupted by special switching times/configurations on a holiday, the PBX resumes the "normal" procedure on the day after the holiday.
Example: Outside of the holidays, each weekday starts and ends with Configuration 2. If Configuration 3 is activated due to a holiday, it only remains activated until midnight (unless, the next day is also a holiday). The next "normal" weekday starts again with Configuration 2.

Configuring the holidays is done separately in the holiday table on the page COMset ▶ Functions ▶ Calendar ▶ Holidays. In addition to or instead of public holidays, you can also use this to enter company holidays, for example. All entered and activated holidays are handled the same way regarding the switching times.

Copying Switching Times

Copy the switching times for the first weekday set to the week days with the same switching times.

To do this, open the page displayed in the figure to the right.

Under "***Source***", select the configured weekday from the list field.

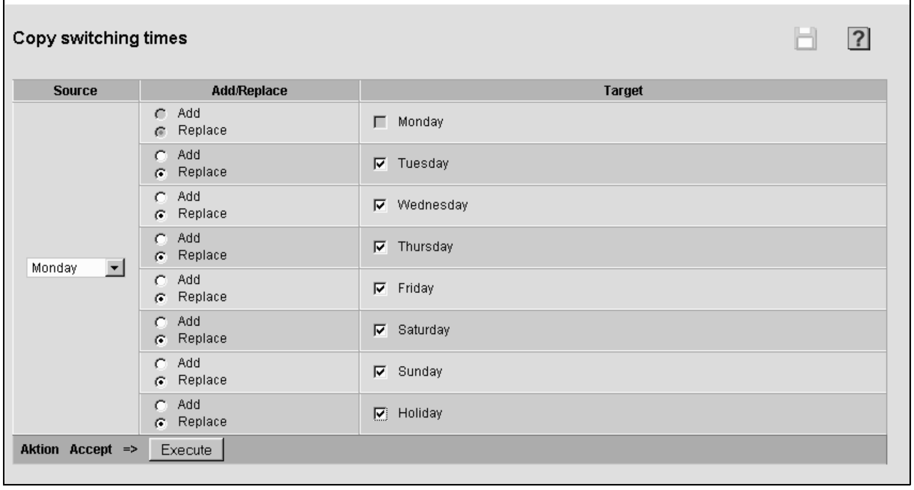
Under "***Add/Replace***", activate action to be performed.

Under "***Target***", activate the weekdays on which the same switching times are required.

Then click "***Execute***" to apply the settings to the PBX.

This completes the configuration for the configuration switching times.

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ Configurations ▶ Copy switching times



Checking Switching Times

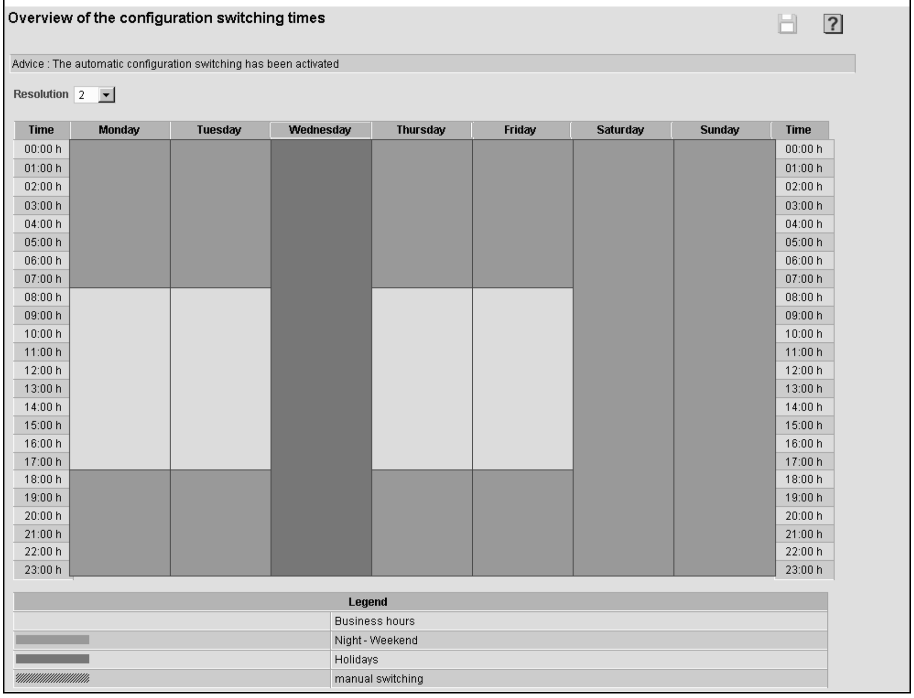
Check the switching times that are configured for the current week in your PBX.

To do this, open the page displayed in the figure to the right.

If a holiday has been configured for the current week, the weekday in question is highlighted in colour. In addition, the changes resulting from the holiday switching times configured are displayed.

If the automatic configuration switching is changed by a manual switchover, it is also displayed.

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ Configurations ▶ Overview switching times



Making Configuration-dependent Settings

The following settings are configuration-dependent:

- Profile assignment of the subscribers (and for example, therefore exchange line authorization, telephone number display and call forwarding for subscribers)
- Profile assignment of the groups (and for example, therefore text before ringing, exchange line authorization, telephone number display and call forwarding for groups)
- Fallback options for the external ports (fallback telephone number and fallback according to time)
- Call distribution for subscribers, groups and door calls
- Relays with the operation mode "Configuration-dependent"
- Call distribution to internal destination numbers (subscribers, groups, voice mail/fax boxes (only with COMmander VMF module), automatic receptions)
- Door call distribution (only with COMmander 2TSM analog module)
- VoIP/GSM routing
- Call forwarding for own external numbers
- Readiness of the voice mail/fax boxes; reject anonymous fax calls; call acceptance of the voice mailboxes, e. g. replacement function (only with COMmander VMF module)

Profile Assignment of Subscribers

Configure the profile assignment for the subscribers as configuration-dependent.

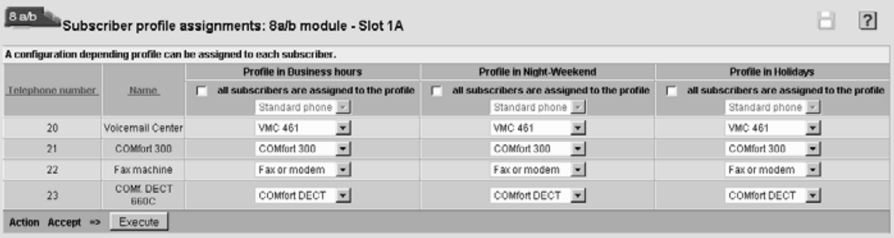
To do this, open the page displayed in the figure to the right.

Select the desired profile from the list fields.

Then click "***Execute***" to apply the settings to the PBX.

Note: *The profiles to be assigned here can be configured under COMset ▶ Internal numbers ▶ Subscriber (scr) ▶ Profiles ▶ Configuration.*

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ Internal numbers ▶ Subscriber (scr) ▶ Profiles ▶ Profile assignment ▶ Slot ...



Profile Assignment of Groups

Configure the profile assignment for the groups as configuration-dependent.

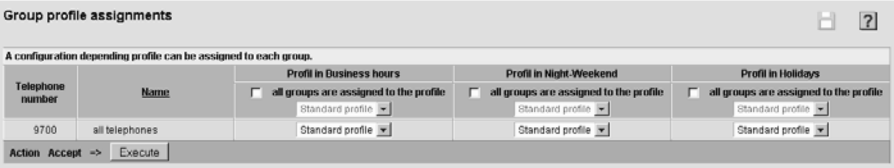
To do this, open the page displayed in the figure to the right.

Select the desired profile from the list fields.

Then click "***Execute***" to apply the settings to the PBX.

Note: *The profiles to be assigned here can be configured under COMset ▶ Internal numbers ▶ Groups ▶ Profiles ▶ Configuration.*

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ Internal numbers ▶ Groups ▶ Profiles ▶ Profile assignment



Call Distribution on ISDN Point-to-Multipoint Connections

Configure the call distribution as configuration-dependent.

To do this, open the page displayed in the figure to the right.

Click the appropriate tab to select the configuration to be changed.

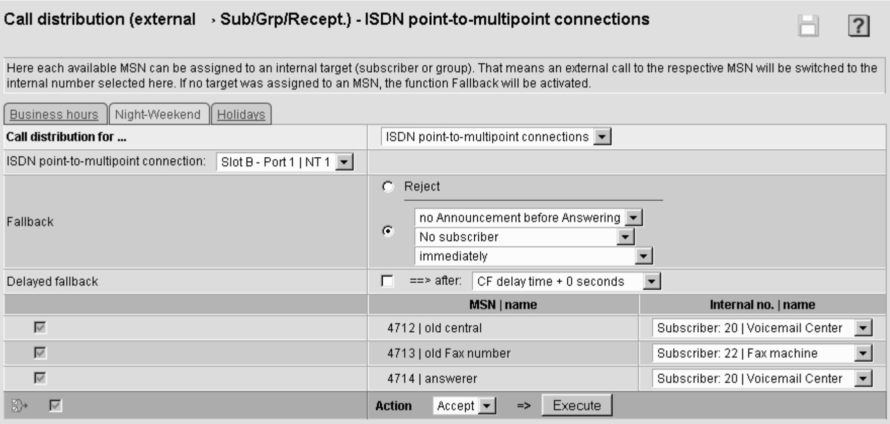
Under "***Call distribution for ...***", select from the list field "***ISDN Point-to-multipoint connections***".

Under "***ISDN Point-to-multipoint connection***", select from the list field one of the already configured Point-to-Multipoint connections.

Under "***Internal no. / name***", select an internal telephone number for each MSN from the list fields.

Then click "***Execute***" to apply the settings to the PBX.

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ External numbers ▶ ISDN connections ▶ Call distribution



Call Distribution on ISDN Point-to-Point Connections

Configure the call distribution as configuration-dependent.

To do this, open the page displayed in the figure to the right.

Click the appropriate tab to select the configuration to be changed.

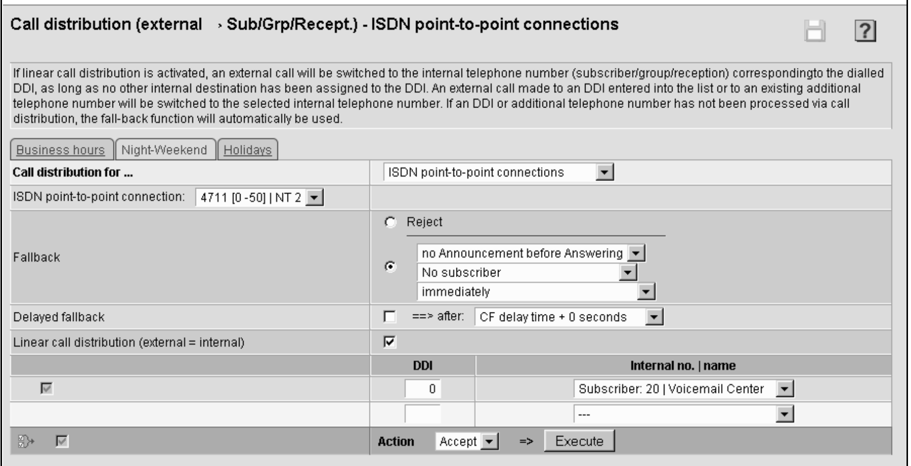
Under “***Call distribution for ...***”, select from the list field “***ISDN Point-to-point connections***”.

Under “***ISDN Point-to-point connection***”, select one of the Point-to-Point connections already configured from the list field.

Under “***Internal no. / name***”, select an internal telephone number for the DDI from the list field.

Then click “***Execute***” to apply the settings to the PBX.

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distribution*



Door Call Distribution

Configure door terminal call distribution as configuration-dependent.

To do this, open the page displayed in the figure to the right.

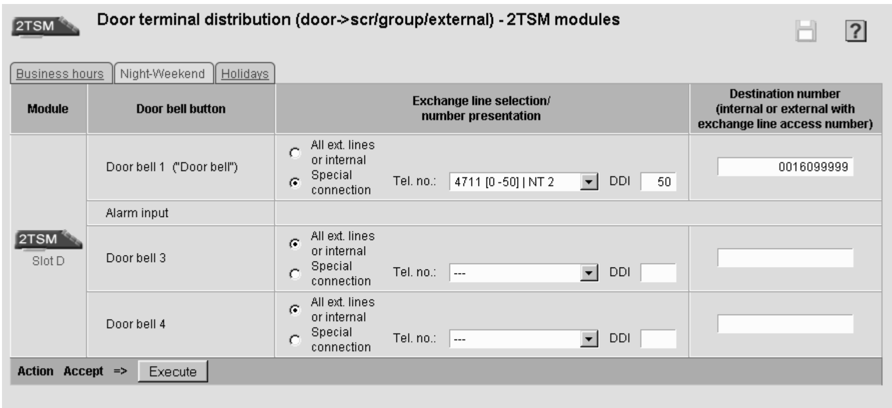
Click the appropriate tab to select the configuration to be changed.

Under “***Destination number (internal or external with exchange line access number)***”, enter the internal and external telephone number that is to be called when the door bell button is pressed.

If you have entered an external telephone number and would like to have a specific telephone number presented to the person being called, under “***Exchange line selection/ number presentation***”, select a specific connection based on the telephone number of the person being called.

Then click “***Execute***” to apply the settings to the PBX.

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ *Door/switching/music module* ▶ *Call distribution*



Relay Settings

Configure Relay 3 as configuration-dependent.

To do this, open the page displayed in the figure to the right.

Under “***Relay mode***”, select the desired operational mode (in this case configuration-dependent) from the list field.

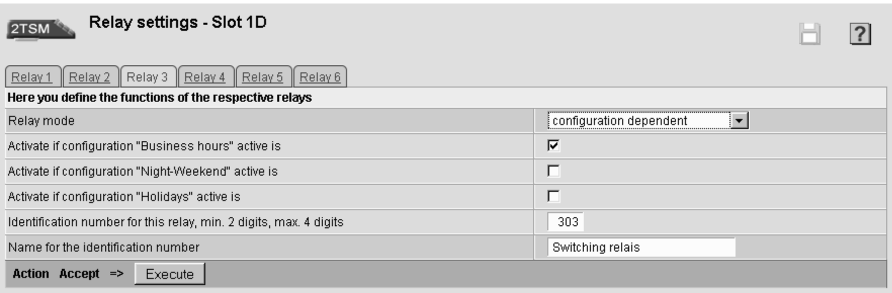
Activate the configurations in which the relays should be activated.

Under “***Identification number for this relay***”, select a 2 to 4-digit number.

Under “***Name for the identification number***”, enter a suitable name with a maximum of 16 characters.

Then click “***Execute***” to apply the settings to the PBX.

The figure shows you the following page with the settings for Example 3 on page 34:
COMset ▶ *Door/switching/music module* ▶ *Relay settings* ▶ *Slot ... TSM* ▶ *Relay 3*



Example 4: Installing a COMmander S_{2M} module for a primary multiplex interface

In the following example (see figure), a COMmander S_{2M} module is added to the PBX from Example 2 on page 28. The NT with a Point-to-Point connection is simultaneously replaced by an NTPM (Primary Multiplex Interface) with the same telephone number range. Instead of the two previous B-channels, this provides 30 B channels for external calls.

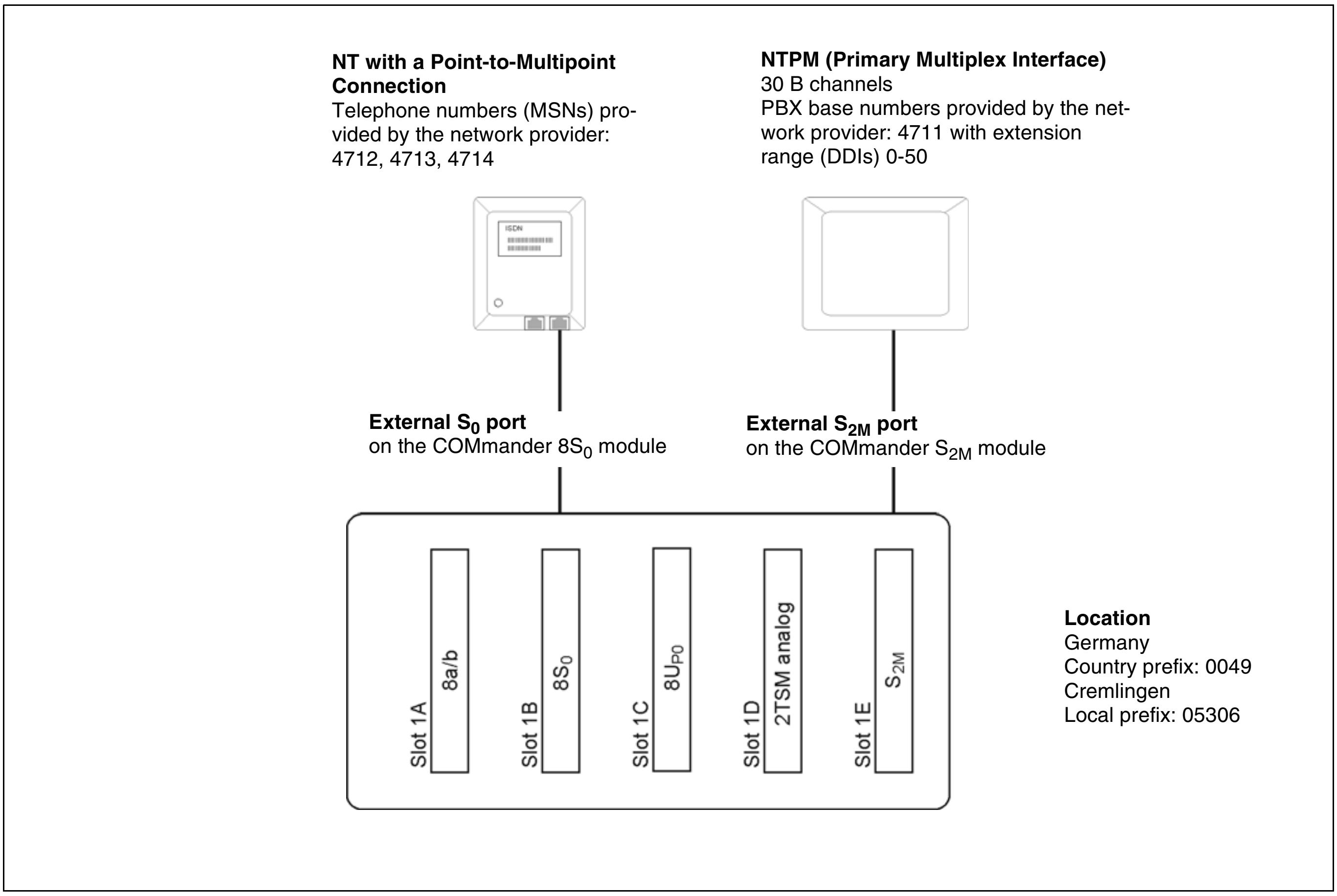
All the necessary configuration settings will be explained step-by-step.

Important: *If you would like to continue to use the telephone numbers and therefore also the settings from a previous Point-to-Point connection as shown in the example, proceed as follows: Do not yet change the existing settings on the external S_0 port! Carry out the hardware configuration and the port configuration for the primary multiplex interface as described in the following. On the page COMset ▶ External numbers ▶ ISDN connections ▶ Telephone numbers ▶ PBX base number/bundles, switch the port for the PBX base numbers entered from the external S_0 port to the S_{2M} port. Now set the available S_0 port to "free" or "internal".*

When installing a COMmander S_{2M} module, refer to the description for retrofitting or upgrading the PBX in the installation and commissioning instructions.

Note: *Note that this section thoroughly describes only the most important settings. You will be referred to the corresponding sections for additional settings.*

The combination of the devices listed here is an example only and may vary greatly from your PBX.



Logging into the Configuration Manager

Important: *To be able to perform basic configuration, the PBX and your computer must be correctly connected with each other.*
The settings for this are described in the Installation and Commissioning Instructions and must be noted when initially operating the PBX.

Start the configuration manager. Enter the user name "***admin***" and the 6-digit admin PIN and confirm by clicking "***Log-in***".

You must logged on as administrator in order to perform the configuration because initially only the administrator has the necessary rights. After you have logged in, you see a menu with a tree structure on the left side of the page. You now have access to the entire system configuration. The structure is similar to the directory structure on a hard drive where each folder can be described with a unique path.

For reasons of simplicity, this **path information** is shown above each figure for each configuration example. This shows the exact location of each menu option in the tree structure.

Hardware Configuration

Assign the current modules to the expansion slots on the mainboard.

To do this, open the page displayed in the figure to the right.

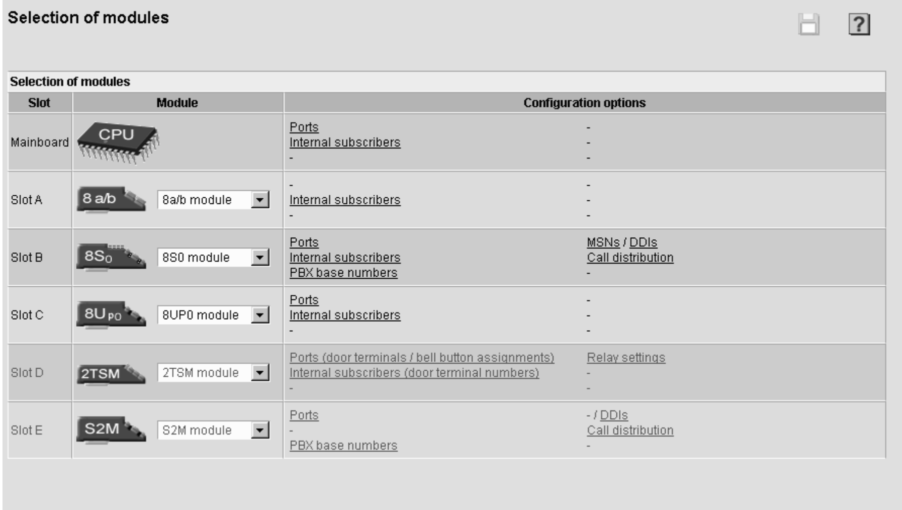
The "***Slot***" column lists each module slot on the mainboard (slot 1A to 1E).

Under "***Module***", select a module from the list field that has been inserted in the associated module slot. For available module slots, select "not defined" from the list field.

In the "***Configuration options***" column, you can use the hyperlinks to directly access the settings described at the end.

These settings complete the hardware configuration.

The figure shows you the following page with the settings for Example 4 on page 39:
COMset ▶ Hardware ▶ Selection of modules



Note: *Note that parameters set here correspond to the real hardware settings on your PBX (question: Which module is inserted in which slot?). If this is the case, the settings are marked in black. If this is not the case, the settings are marked in red.*

For more information on this, refer to the module descriptions in the Installation and Commissioning Instructions.

Port Configuration

Port configuration is the basic set-up for the inserted modules as well as for the CPU. The connection assignments for the ports must be set depending on the intended use and adapted to the actual connection options on site.

S_{2M} port, external

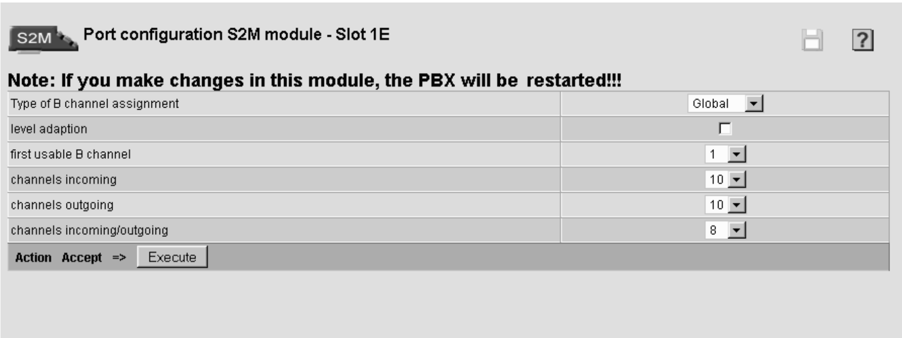
Configure the external S_{2M} port on the COMmander S_{2M} module.

To do this, open the page displayed in the figure to the right.

Under "***Type of B channel assignment***", select from the list field "Global" (B channels that are not permanently assigned) or "Selected" (B channels that are individually and permanently assigned). The application for the type of B channel assignment is made to the network provider and must be entered here together with the remaining values according to the order confirmation from the network provider.

Under "***first usable B channel***", select the first B channel from the list field that has been provided by your network provider or which he has provided as first for the PBX due to restricted conditions.

The figure shows you the following page with the settings for Example 4 on page 39:
COMset ▶ Hardware ▶ Port configuration ▶ Slot ... S2M



Under "***channels incoming***", select the number of B channels configured for incoming calls from the list field.

Under "***channels outgoing***", select the number of B channels configured for outgoing calls from the list field.

Under "***channels incoming/outgoing***", select the number of B channels configured for both directions from the list field.

Click "***Execute***" to accept the settings in the PBX.

This completes the configuration for the COMmander S_{2M} module.

Note: *Under "**level adaption**", changes are only necessary in exceptional cases (when a very long connection between the NTPM and the PBX results in problems with, for example, call interruption). This function is normally disabled.*

External Telephone Numbers

The telephone numbers provided by your network provider for the network termination unit (NT 2) must be entered in the PBX.

Collect the telephone numbers/data provided by your network provider/VoIP provider for the following settings.

Primary Multiplex Interface

In the PBX, configure the existing primary multiplex interface with the telephone numbers provided by the network provider.

To do this, open the page displayed in the figure to the right.

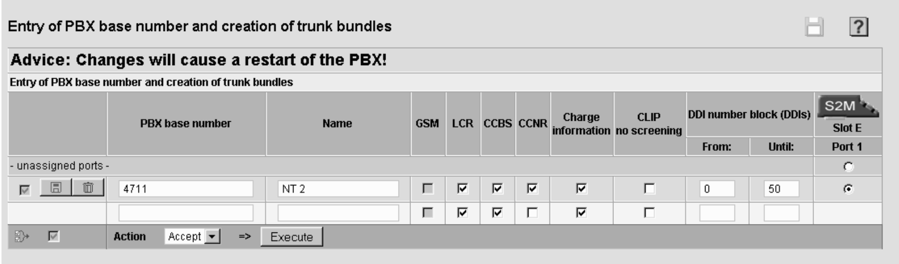
Under "***PBX base number***", enter the PBX base number (without extension).

Under "***Name***", enter any name with a maximum of 16 characters.

Under "***DDI number block (DDIs)***", enter the extension range with the lowest and the highest DDI.

Then click "***Execute***" to apply the settings to the PBX.

The figure shows the following page with the settings for Example 4 on page 39:
COMset ▶ External numbers ▶ ISDN connections ▶ Telephone numbers ▶ PBX base numbers/trunk bundles



The primary multiplex interface is now configured.

Select the S_{2M} port in the line in question.

Note: *For addition settings (call distribution), refer to the descriptions of Point-to-Point connection (see Call Distribution on page 23).*

Make sure to follow the instructions provided by your network provider for entering the DDI number block exactly. If you want to enter a one or two-digit DDI range, your network provider must provide you with one and two-digit DDIs. The same applies to a two and three-digit DDI range.

*A change under "**GSM**" and "**LCR**" is only necessary in exceptional cases (for example, when operating a GSM gateway) (see VoIP and GSM Routing (Exception Telephone Numbers) on page 89 and Least Cost Routing with Soft-LCR easy on page 84 and Least Cost Routing with Soft-LCR 4.0 on page 85).*

*By enabling **CCBS** and/or **CCNR** a callback on busy (CCBS) or on no reply (CCNR) can be offered to external callers if the service feature has been released by the network operator (see Automatic Callback on Busy (CCBS) on page 51 and Automatic Callback on no Reply (CCNR) on page 52).*

*Under "**Charge information**", a change is only necessary if problems are caused by call charges (see Recording Call Data on page 62).*

*Under "**CLIP no screening**", a change is only necessary if you would like to transfer special telephone numbers (see Customer-defined Telephone Number Information Presentation for "CLIP no Screening" on page 54).*

Features – Function and Configuration

This chapter describes individual features on the PBX. It includes information about hardware and software requirements, configuration and how to use the features. The table near every feature contains the following information:

HW requirements	If a special module (e.g., COMmander 2TSM analog module), a system telephone or special accessory (e.g., a speaker) is necessary, it is listed here. Telephones and external connections as well as the necessary a/b, S_0 and U_{P0} modules are not listed here.
SW requirements	The required PBX firmware version for the function described is indicated here. If a system telephone is necessary for the operation of the function, the firmware version of the system telephone as well as the version of the required PC software are listed here. The version listed may be necessary for using the function or a partial aspect of it.
Dongle release	If the release of a special function is necessary, it will be listed here. The general subscriber release necessary for all functions is not listed here.
Configuration via / Setting via	The following lists the configuration manager pages which contain settings for those features that can be or need to be configured, one after the other and from the top to the bottom. In addition, if a system telephone is required for the configuration, it is also listed here. Configuration on a system telephone can be carried out via the telephone menu (not on COMfortel 1100), via the PC program COMfortel Set or – on the COMfortel VoIP 2500 AB – via the configuration manager on the telephone.

Direct changes to subscriber and group properties in subscriber and group configuration settings are always described here. However, the same settings can also be configured via the subscriber and group profile (see also Profiles and Properties on page 11). If the settings should be dependent on the configuration or not changeable by the user, it is mandatory that the settings be changed via the profile. The pages indicated in the following:

... ▶ Properties ... + selection in the list field at the top
can always be substituted by the following pages:

... ▶ Profiles ▶ Configuration ▶ Configure ▶ ...

Additional information about the individual settings can be found in the online help. This can be accessed from every page of the web interface by clicking the question mark symbol. ?.

Call Distribution and Reachability

External Call Distribution to Subscribers, Groups and Voice Mail/Fax Boxes

To reach internal subscribers, groups and voice mail/fax boxes from external telephones, you need to assign external numbers to internal subscribers and groups.

Separate call distribution is set for each external connection. A single connection as is defined here as:
- A PTMP connection (fixed network or GSM)
- A single PTP connection (fixed network or GSM)
- A bundled PTP connection made up of several S_0 connections with the same PBX base numbers and the same DDI number block
- A S_{2M} connection
- A VoIP account

HW requirements	COMmander VMF module (for voice mail and fax boxes)
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Telephone numbers* *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Telephone numbers* *COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Voice mailboxes* ▶ *Telephone numbers* *COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Fax boxes* ▶ *Telephone numbers* *COMset* ▶ *External numbers* ▶ *Analogue connections* ▶ *Call distributions* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *Direct dialing numbers for PTP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *Additional PBX Base numbers* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distributions* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ *Provider* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ *Accounts* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ Call distributions

For explanations concerning the table, see page 42

Configuration of the Feature

First of all the external numbers available on the external ISDN connections need to be entered.

A PBX of this size may have multiple DDIs on a PTP connection, so these DDIs are not entered individually. Instead, the DDIs made available by the network provider are entered as a DDI number block (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles*). Only in additional configuration options are the DDIs that differ from the linear call distribution or that are necessary for other functions entered (e.g., for remote configuration). Each DDI entered is listed under *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *Direct dialing numbers for PTP connection*.

In comparison, all the available MSNs must be entered for a Point-to-Multipoint connection (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection*).

To be able to configure call distribution for connected GSM gateways, enter the dialable telephone numbers – that you entered in the GSM gateway configuration – as MSNs or DDIs (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *Direct dialing numbers for PTP connection* or *...* ▶ *PTMP connection*). However, before you can enter GSM gateway DDIs in the PTP connection, you need to create the requested connection by entering a PBX base number of your choice with a DDI number block, e.g., PBX base number 4711 with a DDI number block 1 to 2 (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles*).

If accounts have been created with one or more VoIP providers, these account data must be entered (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *provider* and ... ▶ *Accounts*; see also Voice over IP (VoIP) on page 87).

In some countries, the Point-to-Point connection may provide additional telephone numbers besides the DDIs that are not appended to the PBX base number. These numbers must be entered as additional numbers into the Point-to-Point connection (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *Additional PBX Base numbers*).

After this, internal numbers for subscribers (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Telephone numbers*), groups (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Telephone numbers*), voice mailboxes (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Voice mailboxes* ▶ *Telephone numbers)* and fax boxes (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Fax boxes* ▶ *Telephone numbers)* must be created. To achieve linear call distribution, the internal numbers must match the DDIs assigned by the network provider. For this purpose, the PBX has a free numbering plan for assigning 2 to 4-digit numbers in the 10-9999 range.

For the DDIs in a Point-to-Point connection, a linear call distribution is used. Every available DDI in the DDI number block is assigned to the corresponding internal number, e.g., 4711-99 to the internal number 99. Linear call distribution may be deactivated separately per connection. The DDIs must then be entered individually and assigned to the internal numbers directly (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distributions*).

An individual subscriber, group or a voice mail/fax box may be selected for each DDI and for each MSN (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call distributions*).

If no internal number has been assigned to the number called by the external telephone, you can configure a fallback number that takes the call immediately or after a delay (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call distributions*).

Use/Check of Features

1. Configure call distribution for an external number.
2. Call the external number.
3. Check if the correct subscriber is ringing.
4. Check settings under *"COMset* ▶ *External numbers* ▶ *Overview"*.

Dependency/Limitations

An external number may be assigned to a single subscriber, group or voicemail/fax box. If several subscribers ring simultaneously, they are grouped.

External numbers, e.g., those used as remote switching/programming numbers or call through numbers, may not be used in call distribution.

Different ringer rhythms can be configured to differentiate external calls coming in over different external numbers (e.g., business numbers, private numbers) according to the ringer rhythm (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *Direct dialing numbers for PTP connection, ...* ▶ *PTMP connection* and *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Accounts*). In the case of a PTP connection, it is necessary to enter the DDIs separately.

If two or three-digit direct dial-in numbers are requested for the PTP connection, they must be supported by the network provider and entered into the DDI number block. A DDI number block of 10-999 lets you use, e.g., two and three-digit direct dial-in numbers; however, a DDI number block of 100-999 only allows three-digit dial-in numbers.

If 1-digit DDIs are needed, they must be entered manually and assigned to the internal numbers in the call distribution. This is only possible if these 1-digit DDIs are supported by the network provider and are entered in the DDI number block (e.g., 0-199). Note that if you use 0 as the central telephone number, the dial-in numbers 00-09 and 000-099 will no longer be available.

Door Call Distribution

Door bell ringing may be signalled on internal and external telephones.

HW requirements	A *COMmander 2TSM analog Module* and a door terminal system according to FTZ 123 D12-0 (e.g. TFS-Dialog 100)
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Door/switching/music module* ▶ *Port configuration* *COMset* ▶ *Door/switching/music module* ▶ *Telephone numbers* *COMset* ▶ *Door/switching/music module* ▶ *Call distributions* *COMset* ▶ *Door/switching/music module* ▶ *Rhythm and cadence*

For explanations concerning the table, see page 42

Configuration of the Feature

At least one of the available door bell button inputs on the *COMmander 2TSM analog Module* must be assigned to a door terminal as a door bell button (*COMset* ▶ *Door/switching/music module* ▶ *Port configuration*).

After this, an internal or external target number must be entered for each door bell button (*COMset* ▶ *Door/switching/music module* ▶ *Call distributions*).

In the case of an external target number, the exchange line used by the PBX and telephone number to be presented, if applicable, must also be selected (*COMset* ▶ *Door/switching/music module* ▶ *Call distributions*). The following settings are possible:

- “All external connections or internal”: The external target number can be dialled over any available fixed network connection.
- “Special connection”: The external target number can be dialled via a special fixed network connection selected by the telephone number to be presented.
- “VoIP account”: The external target number can be dialled over a specified account.

For internal door calls, a ringer rhythm and the ringing duration must be configured for each door bell button (*COMset* ▶ *Door/switching/music module* ▶ *Rhythm and cadence*).

To call the door and to present the internal number, a door number must be configured (*COMset* ▶ *Door/switching/music module* ▶ *Telephone numbers*). Select this from the telephone number plan that permits the allocation of 2 to 4-digit telephone numbers in the 10-9999 range.

Use/Check of Features

1. Configure a door and the corresponding door call distribution.
2. Press a door bell button.
3. Check if the number entered is called with the correct ringer rhythm and the correct duration.

Dependency/Limitations

If “All external connections or internal” is set, VoIP and GSM routing is used if the target number was entered in the exception telephone number table and was activated for the current configuration (see page 89).

Analog door terminal systems (e.g. TFS-Dialog 200, TFS-Dialog 300 and TFS-Universal a/b) are not connected via a COMmander 2TSM analog module but - like an analog telephone - to an internal analog a/b port of the PBX. The configuration is done on the door terminal system itself, including door terminal call distribution to internal subscriber and group telephone numbers. For more information on this, please refer to the instructions of the door terminal system.

Second Ringer Bell

If the customer likes to notice the ringing at places where he cannot hear his telephone, you can connect a second ringer bell to the *COMmander 2TSM analog Module*.

HW requirements	A *COMmander 2TSM analog Module*
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Door/switching/music module* ▶ *Telephone numbers* *COMset* ▶ *Door/switching/music module* ▶ *Relay settings*

for explanations concerning the table see page 42

Configuration of the Feature

You can configure the second ringer bell to ring for calls to a subscriber or group number and/or when pressing a door bell button (*COMset* ▶ *Door/switching/music module* ▶ *Telephone numbers*).

If e.g. a lamp should be switched on simultaneously to the second ringer bell, it is possible to configure as switching relay on the *COMmander 2TSM analog module* (*COMset* ▶ *Door/switching/music module* ▶ *Relay settings*).

Use/Check of Features

1. Configure a second ringer bell.
2. Call the corresponding subscriber or press the corresponding door bell button.
3. Check if the second ringer bell is ringing.

Call Order of the Group Members

See Ringing Sequence on page 69.

Call Forwarding for Subscribers

Thanks to Call Forwarding (CF) for subscribers, you can forward internal and external calls targeted to your telephone to other internal telephones or external connections.

As there may be different reasons for not being able to take a call, there are three different kinds of call forwarding: “CF unconditional”, “CF on busy” and “CF on no reply”.

If “CF on busy” and “CF on no reply” are activated at the same time, both variants are used. Depending on the situation, if the telephone is busy or if nobody takes the call, the call is forwarded to one or more destination numbers.

If “CF unconditional” is activated in addition to “CF on busy” and/or “CF on no reply”, only “CF unconditional” is used; this means that all the calls are forwarded to the “CF unconditional” destination. In this case, the other forwarding settings are ignored but are still active in the background. As soon as “CF unconditional” is switched off, the other active forwarding settings are used again.

HW requirements	– – –
SW requirements	Version 3.8A (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Signalization by tones* + subscriber selection in the list field at the top *COMset* ▶ *General settings* ▶ *General*

For explanations concerning the table, see page 42

Configuration of the Feature

Individual call forwarding can be activated and an access number set up for each subscriber (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

To forward only external calls directed to your telephone to other internal telephones or external connections, activate “only external calls” (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

To activate subscriber CF for calls already forwarded to your telephone (CF cascading), activate “CF cascading” (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

To activate subscriber CF for group calls to your telephone, activate “CF on group call” (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top). To forward the complete group call to a specific destination, use the group call forwarding function instead.

Individual users can be restricted from being able to arbitrarily configure call forwarding to an external target (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

The “Delay time for CF on no reply” can be changed. This applies to subscriber and group call forwarding (*COMset* ▶ *General settings* ▶ *General*).

“Fallback on busy” can be turned on for the “CF on busy”. If the busy subscriber hangs up, the already forwarded call falls back to him again – in as much as the call has not yet been picked up (*COMset* ▶ *General settings* ▶ *General*).

A special dial tone can be activated to remind the user about, for example, configured call forwarding or Do-not-Disturb (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Signalization by tones* + subscriber selection in the list field at the top).

Use/Check of Features

1. Configure call forwarding for subscribers.
2. Call the forwarded telephone.
3. Depending on the type of call forwarding activated, the call arrives at the target configured.
4. A forwarded system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus* shows “CF unconditional” on the display.
5. A forwarded subscriber is indicated by a yellow glowing LED on the busy indicator light area on system telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus*.

Dependency/Limitations

“Call forwarding for subscribers” may be configured on each internal telephone using a programming sequence.

On the system telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus,* “Call forwarding for subscribers” may be configured using a function key/menu (see the manual for the telephone).

On an analog T-Net telephone, “Call forwarding for subscribers” may be configured using a T-Net menu/function key – if present – (see the manual for the telephone).

If you configure an internal telephone as the forwarding target, it requires at least partial exchange line authorization for forwarded external calls.

To avoid having to use a second call channel in the case of call forwarding to an external destination, the PBX tries to reroute the call directly at the network provider using call deflection. The following requirements must be fulfilled for this purpose: 1. The incoming call was made via an ISDN network provider supporting call deflection. 2. The rerouted telephone number is neither called as a group member, nor is it required to consider parallel calling or fallback on busy.

Call Forwarding for Groups

With call forwarding for groups, you are able to forward internal and external calls targeted to a group to other internal telephones or external connections. This allows a person who is not a member of this group and cannot log in to take these calls.

Call forwarding for groups makes sure that somebody is reached under the group number even if the calls can temporarily not be accepted by members of the group.

As there may be different reasons for not taking a call, e.g., nobody is there during a short or long period of time or all the logged-in telephones are busy, there are four different call forwarding settings: “CF unconditional”, “CF on busy” and “CF on no reply” and “CF if all subscribers are logged out”. A different forwarding target may be configured for each of the four call forwarding types.

If “CF on busy”, “CF on no reply” and “CF if all subscribers are logged out” are active at the same time, all three variants work. Depending on which case occurs, whether the line is busy or nobody takes the call or all subscribers are logged out, the call is forwarded to one or more destination numbers.

If “CF unconditional” is also activated for “CF on busy” and/or “CF on no reply”, only “CF unconditional” works; this means that all the calls are forwarded to the forwarding destination for “CF unconditional”. The other variants are ignored but still activated. As soon as “CF unconditional” is switched off, the other variants that are still switched on are active again.

HW requirements	– – –
SW requirements	Version 3.8AE (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top *COMset* ▶ *General settings* ▶ *General*

For explanations concerning the table, see page 42

Configuration of the Feature

Individual call forwarding can be activated and an access number set up for each group (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top).

To forward only external calls directed to your group to other internal telephones or external connections, activate “only external calls” (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top).

To activate group CF for calls already forwarded to your group (CF cascading), activate “CF cascading” (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top).

To activate group CF for calls directed to a superordinate group, activate “CF on group call” (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top).

The “Delay time for CF on no reply” can be changed. This applies to subscriber and group call forwarding (*COMset* ▶ *General settings* ▶ *General*).

“Fallback on busy” can be turned on for the “CF on busy”. If a subscriber in the busy group hangs up, the already forwarded call falls back to him again – in as much as the call has not yet been picked up (*COMset* ▶ *General settings* ▶ *General*).

Use/Check of Features

1. Configure call forwarding for groups.
2. Call the forwarded group.
3. Depending on the type of call forwarding activated, the call arrives at the target configured.

Dependency/Limitations

If you configure an internal telephone as the forwarding target, it requires at least partial exchange line authorization for forwarded external calls.

“Call forwarding for groups” may be configured on each internal telephone using a programming sequence. The telephone used must be a member of the group to be forwarded and have authorization for CF/Follow me for groups (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

LCR will be used for call forwarding for groups to an external target (e.g., mobile telephone) if at least one member of the group has LCR activated. It does not matter if the corresponding subscriber is logged in or not.

The group property “Call Restrictor settings of the subscribers” functions like “No Call Restrictor” for call forwarding for groups.

The group property “Call Deblocker settings of the subscribers” functions like “No Call Deblocker” for call forwarding for groups.

The group property “Exchange line authorization of the subscribers” functions like “International” for call forwarding for groups.

The group property “Short-code dialling authorization for the subscribers” functions like “Short-code dialling authorization for all group members” like for call forwarding for groups.

Preferred exchange line/Number Presentation: The number of the group is always presented.

“Exchange line authorization for the subscribers” functions like “Standard/Standard” for call forwarding for groups.

To avoid having to use a second call channel in the case of call forwarding to an external destination, the PBX tries to reroute the call directly at the network provider using call deflection. The following requirements must be fulfilled for this purpose: 1. The incoming call was made via an ISDN network provider supporting call deflection. 2. The rerouted telephone number is neither called as a group member, nor is it required to consider parallel calling or fallback on busy.

Call Forwarding for External Numbers (ISDN and VoIP)

With call forwarding, external calls targeted to exchange line numbers can be forwarded to other external connections.

This call forwardings may also be configured in the central office. In addition to the monthly fee for this service, the calls forwarded from your connection to the target are charged to you.

As there may be different reasons for not taking a call, there are three different call forwarding settings: “CF unconditional”, “CF on busy” and “CF on no reply”. A different forwarding target may be configured for each of the three call forwarding types.

If “CF on busy” and “CF on no reply” are active at the same time, both variants work. Depending on which case occurs - if it is busy or nobody takes the call - the call is forwarded to one or more destination numbers.

If “CF unconditional” is also activated in addition to “CF on busy” and/or “CF on no reply”, only “CF unconditional” works; that means that all calls are forwarded to the forwarding destination of “CF unconditional”. The other variants are ignored, but they are still activated. As soon as “CF unconditional” is switched off, the other variants that are still switched on are active again.

HW requirements	– – –
SW requirements	Version 3.8AC (PBX) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *PTP connection* *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Call Forwarding* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Call Forwarding* ▶ *PTP connection* *COMset* ▶ *External numbers* ▶ *VoIP / GSM Routing* ▶ *Exception number* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

Call forwarding for ISDN DDIs may be configured under *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *PTP connection*.

Call forwarding for ISDN MSNs may be configured under *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *PTMP connection*.

Call forwarding for VoIP DDIs may be configured under *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Call Forwarding* ▶ *PTP connection*.

Call forwarding for VoIP MSNs may be configured under *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Call Forwarding* ▶ *PTMP connection*.

You can set each port so that call forwarding may be configured by the network provider (not in case of VoIP) instead of using the PBX (2nd call channel, *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *...*). Configuration via the user's telephone stays the same. It is possible that the configuration may take more time (time to acknowledgement).

The delay time configured for the call forwarding "on no reply" via the PBX (2nd call channel) may be changed per port (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call Forwarding* ▶ *...*). The delay time for call forwarding configured in the central office is fixed to 20 seconds.

The default configuration for call forwarding for MSNs and DDIs via the PBX (2nd call channel) is the call channel on the same port (e.g., because the charges for different S_0 ports are assigned to different persons). Since the probability is higher that call forwarding may not be executed due to a busy call channel, you can change this setting dependent on the port and authorize any call channel (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call Forwarding* ▶ *...*).

The routing can also be done depending on the destination number. Routing must be activated for this (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call Forwarding* ▶ *...*) and the destination number recorded in in the Call Through list (*COMset* ▶ *External numbers* ▶ *VoIP / GSM Routing* ▶ *Exception number*).

Individual subscribers can be restricted from configuring call forwarding for external numbers using a telephone (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

On a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB*, a function key can be configured for call forwarding for external numbers (see the manual for the telephone).

Use/Check of Features

1. Configure call forwarding.
2. Call the forwarded number.
3. Depending on the type of call forwarding activated, the call should reach the target configured.

Dependency/Limitations

After a firmware update with firmware version 3.8, it is also possible to set call forwarding for external telephone numbers as configuration-dependent. For situations where your settings are supposed to apply even after configuration switchovers, you can use the so-called permanent configuration. If permanent configuration is enabled, the configuration-dependent settings will be overridden by the permanent configuration settings.

On a system telephone COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB, you can switch on/off/over the call forwarding of external numbers either for the current configuration or for permanent configuration using the function key (see the manual for the telephone). With a standard telephone you can change call forwarding in the permanent configuration and switch on the permanent configuration or the current configuration using a digit sequence. As soon as every call forwarding has been switched off in the permanent configuration, the permanent configuration for the corresponding telephone number is switched off and the configuration-dependent settings apply. Tip: To override the configuration-dependent settings with no call forwarding, enable call forwarding in the permanent configuration without specifying a destination number.

If you would like to forward calls to a T-Net Box, network provider must release call forwarding.

Multi-path Call Forwarding

With multi-path call forwarding, internal and external calls targeted to a subscriber or group may also ring on an additional number (internal or external).

This is useful, for example, if a single user needs to be reachable on two internal telephones or internally and on his mobile telephone at the same time, without configuring call forwarding.

Multi-path call forwarding is always executed even if the "main telephone number" is, for example, rerouted or busy (exception: enabled MSN/DDI call forwarding via the network provider).

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

Multi-path call forwarding can be activated and a destination number configured for each subscriber separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

Multi-path call forwarding can be activated and a destination number configured for each group separately (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top).

Use/Check of Features

1. Configure multi-path call forwarding for a subscriber.
2. Call the corresponding subscriber.
3. The subscriber and the configured target of the multi-path call forwarding both ring.

Dependency/Limitations

Multi-path call forwarding for a group is executed during the entire calling period independent of the ringer rhythm for the group.

Multi-path call forwarding for a group is not executed if the group itself is the target of a forwarded call.

Busy-on-Busy

With help of the Busy-on-Busy function, it is possible to define the maximum active call connections for one group (in single increments until the maximum number of possible external B channels minus two). If this number has been reached, all additional callers hear the busy tone. However, members of the group may make additional calls (outbound calls) if there are still external B channels available.

The “busy after one connection” setting makes sense if the group is used to call one user on several telephones.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

A Busy-on-Busy setting can be activated for each group separately (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top).

Use/Check of Features

1. Configure Busy-on-Busy with “Busy after 1 connection” for a group.
2. Call the group.
3. Accept the call on one of the ringing telephones.
4. Call the group again. You should now hear the busy tone.

Dependency/Limitations

The Busy-on-Busy and Announcement before Answering functions (see Announcement before Answering (Greeting Message) on page 61) can work hand-in-hand. If the function Announcement before Answering “On busy” or “Always” is activated, it may be necessary – especially with a S2M connection – to limit the number of callers that are automatically put on hold in the waiting loop. To do this, the Busy-on-Busy function can be activated for a certain number of connections. If, for example, two telephones are logged into the corresponding group and both are in a call, a maximum of 4 callers will be put into the waiting loop if the limit is six. Callers not put on hold in the waiting loop due to the Busy-on-Busy restriction will directly hear the busy tone instead of the Announcement before Answering. This prevents more callers being on hold than can be accepted within a reasonable time.

Stay Reachable via B Channel Reservation

See B Channel Reservation on page 77.

Knocking/Call Waiting

Knocking is the name for what occurs when an additional call is made during an existing call.

Knocking is indicated by an attention tone and/or a display text (depending on the telephone).

The current call may now be interrupted, disconnected or the caller may be rejected in order to continue the current call undisturbed.

The caller hears a normal ring tone and does not know that there is already an ongoing phone call on the connection called.

The PBX supports knocking for external calls, internal calls, door and alarm calls.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

For each subscriber, knocking must be activate/deactivated and the knocking type selected separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

Use/Check of Features

1. Make a call (e.g., between two internal telephones).
2. Call one of the telephones externally.
3. Audible knocking should be heard on the telephone called (depending on the telephone, the caller knocking is shown on the display).
4. Take the caller with R1 (first call is disconnected).

   or: Take the caller with R2 (first call is put on hold)

   or: Reject the caller with R0 (first call continues).

   or: Select the requested function from the system telephone menu (see the manual for the telephone).

Dependency/Limitations

On some ISDN telephones, knocking needs to be activated separately for the telephone itself (see the manual for the telephone).

On the system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus*, the user can activate/deactivate knocking via the function key/menu (see the manual for the telephone).

Alarm call knocking cannot be deactivated.

If the call knocking is an alarm call, the current call is disconnected when accepting the alarm call.

If a query call is being made when another call knocks, at least one of the two calls must be disconnected before accepting the caller who is knocking.

On an analog T-Net telephone, the “Accept/Reject a knocking caller” via the T-Net function key/menu – if present – is also possible (see the manual for the telephone).

On ISDN telephones and system telephones, the “Accept/Reject a knocking caller” – if possible – can be done using an existing function key or the menu (e.g., with “call accept/reject”; see the manual for the telephone).

Alternation, Conference, Connect, Callback

Alternation “HOLD”

Alternation is the switching between two different calls (internal or external). In contrast to a conference call, both call partners must alternate talking. During the conversation with the first call partner, the other call is put on hold on the PBX. During this time, the call partner hears Music on Hold or an announcement.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	– – –

For explanations concerning the table, see page 42

Configuration of the Feature

This feature does not require configuration via the web interface.

Use/Check of Features

1. Make a call (e.g., between two internal telephones).
2. Make a second call via Query (R + number).
3. Switch between both call partners with the feature Alternation (R2).
4. The active call is disconnected with R1.

   Or: Disconnect the call on hold using R0.

Dependency/Limitations

On ISDN telephones and system telephones, “Alternation” – if possible – is done using a function key or menu (e.g., with “Alternation”, “R-key” or “Call1/Call2”; see the manual for the telephone).

On an analog T-Net telephone, “Alternation” is also possible using the T-Net function key or menu – if present – (see the manual for the telephone).

You do not need the support of the public exchange for the Alternation of two external call partners (telephone service “Alternation” (HOLD)“) because the PBX supports the Alternation via the PBX (2nd call channel). That means, the PBX takes over the job from the public exchange. For this function you need at least two external voice channels.

Three Party Conference “PITY”

A three party conference call is when you talk with two subscribers at the same time (externally or internally).

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	– – –

For explanations concerning the table, see page 42

Configuration of the Feature

The feature does not require configuration over the web interface.

Use/Check of Features

1. Make a call (e.g., between two internal telephones).
2. Make a second call by starting a query (R + number).
3. Start a conference with R0 or via the system telephone menu (see the manual for the telephone).
4. Return to alternation with R2 (status as before the conference).

   Or: Return to alternation with R1 (with change of the active call).

   Or: Select the required function from the menu of the system telephone (see the manual for the telephone).

Dependency/Limitations

On an analog T-Net telephone, “Finishing a conference and return to Alternation” is also possible using the R0 function key/menu – if present – (see the manual for the telephone).

To use this function on ISDN telephones and system telephones, the “Finishing a conference and return to a specific target” function must be supported using a function key or menu (e.g., with “Alternation”, “single connection” or “Call1/Call2”; See the manual for the telephone).

You do not need the support of the public exchange for a Conference of two external call partners (telephone service “Three-party Conference (3PTY)“) because the PBX supports Conferences via the PBX (2nd call channel). That means, the PBX takes over the job from the public exchange. For this function you need at least two external voice channels.

Connect/Call Transfer "ECT"

An internal subscriber can be transferred to another subscriber (internally or externally) using the ECT function. It does not matter if the call partner to be transferred has called you or you have called him. This procedure, connecting two call partners during a query, is called a transfer.

When transferring an external call to another internal telephone, it is not necessary to wait for the second connection before hanging up the receiver. The internal telephone will continue to ring (Blind Transfer).

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *General settings* ▶ *General* *COMlist* ▶ *Call data/charges* ▶ *General settings*

For explanations concerning the table, see page 42

Configuration of the Feature

Authorization can be configured for each individual subscriber for transferring external calls to another external call partner (*COMset* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

A call transferred between two external call partners can be limited to a certain time frame (*COMset* ▶ *General settings* ▶ *General*). This has been implemented for safety reasons since an accidental transfer, for example, to the talking clock or the weather forecast would result in an endless connection.

During a blind transfer, the waiting caller hears the ring tone instead of the Music on Hold. If this behaviour is unwanted, Music on Hold may be played during the ringing period (*behaviour* ▶ *General settings* ▶ *General*).

The setting "Call charge recording changes with subscriber" (*COMlist* ▶ *Call data/charges* ▶ *General settings*) assigns the charges incurred from the time the external calls are transferred to the connected internal telephone. On the exchange line, the telephone service "Advice of Charge during the connection (OCD)" is required; otherwise, all charges are assigned to the last subscriber.

Dependency/Limitations

On some ISDN telephones, the "Transfer" may be made using an available function key or menu instead of going on-hook; see the manual for the telephone; it may also be called "ECT" "Transfer").

With some ISDN telephones, it may happen that your telephone rings after hanging up and that you are reconnected to your first calling partner if you pick up the receiver. Refer to the operating instructions for the telephone to check whether the "PBX Transfer" is activated. Resolve the problem, if necessary.

An external call can only be transferred to an internal subscriber with the necessary exchange line authorization (at least "inbound" for the corresponding exchange line).

For transferring two call partners, you do not need public exchange support (telephone service ECT) because the PBX supports the transfer via the PBX (2nd call channel). That means, the PBX takes over the job from the public exchange. For this function you need at least two external voice channels.

Automatic Callback on Busy (CCBS)

With this function, you are called back automatically when a busy subscriber is available again.

When using this feature, the caller's telephone receives a signal when the partner called is available again.

If the caller hears the busy tone after dialling the number, he can activate the CCBS service by dialling a digit sequence or using the callback key. The callback request remains active at the central office for 45 minutes. If the partner called hangs up the receiver within this time frame, the callback starts and the caller's telephone rings. Picking up the receiver establishes the connection to the target and deletes the callback request in the central office.

If the callback is already unnecessary, it may be actively deleted in the central office.

It is also possible to start a callback on Busy for calls to internal telephones. The callback request is not stored in the central office but in the PBX.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMset* ▶ *General settings* ▶ *CLIP texts*

For explanations concerning the table, see page 42

Configuration of the Feature

If you use an ISDN PTMP connection, it is not necessary to configure the feature via the Web interface. If you use an ISDN PTP connection, you can activate the CCBS function separately for each PBX base number – provided CCBS has been enabled by the network operator.

On system telephones, the callback is shown in the display while ringing. The text displayed may be changed under *"COMset* ▶ *General settings* ▶ *CLIP texts"*.

Use/Check of Features

1. Call a subscriber that is in a call – busy tone.
2. Activate callback (e.g., by dialling the digit sequence R *37#) – acknowledgement tone.
3. Hang up the receiver.
4. As soon as the subscriber has finished the call, the telephone that has activated the callback rings.

Dependency/Limitations

The technical requirements for a callback must be provided, e.g., callback to a PBX is not possible with some network providers on a Point-to-Point connection. If a VoIP account was used for the external call, no callback is possible.

For this function, the network provider must release the telephone service "Completion of Calls to Busy Subscriber (CCBS)" feature.

On ISDN telephones and system telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus*, the "callback" – if possible – is made using a function key or menu (see the manual for the telephone).

If you do not hear an acknowledgement tone, the callback request was refused by the PBX/public exchange.

It is possible that the public exchange will start the callback before the subscriber called has hung up because one of the B channels on his ISDN connection is available.

The PBX/public exchange attempts callback for up to 45 minutes. If the person called has not finished his call by then, the callback is deleted automatically.

In case of a callback, if you do not pick up the receiver in time (after you have been called for 20 seconds), the callback is also cancelled.

If you use an analog T-Net telephone as an internal subscriber telephone and you can start a “Callback on Busy” via the T-Net function key/menu, you can also use this operation (see the manual for the telephone).

Automatic Callback on no Reply (CCNR)

With this function, you are called back automatically as soon as the subscriber called hangs up after his next call.

If the caller hears the ringing tone, but the called person does not take the call, the caller will be able to activate the CCNR service by dialling a digit sequence or pressing the callback button. The callback request remains active in the central office for 45 minutes. If the partner called makes a call and hangs up the receiver within this time frame, the callback starts and the caller’s telephone rings. Picking up the receiver establishes the connection to the target, which rings and deletes the callback request in the central office.

It is also possible to start a Callback on no Reply for calls to internal telephones. The callback request is not stored in the central office but in the PBX.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMset* ▶ *General settings* ▶ *CLIP texts*

For explanations concerning the table, see page 42

Configuration of the Feature

If you use an ISDN PTMP connection, it is not necessary to configure the feature via the Web interface. If you use an ISDN PTP connection, you can activate the CCNR function separately for each PBX base number – provided CCNR has been enabled by the network operator.

On system telephones, the callback is shown on the display while ringing. The text displayed may be changed under *“COMset* ▶ *General settings* ▶ *CLIP texts”*.

Use/Check of Features

1. Call a subscriber.
2. Do not pick up the call on the telephone called.
3. Activate callback with the function key – acknowledgement tone.
4. Replace the receiver.
5. Pick up the receiver of the telephone called and hang it up again.
6. The telephone that has activated the callback rings.

Dependency/Limitations

The technical requirements for a callback must be met, e.g., with some of the network providers, callback to a PBX is not possible on a Point-to-Point connection. If a VoIP account was used for the external call, no callback is possible.

For this function, the network provider must release the telephone service “Completion of Calls on No Reply (CCNR)” feature.

If you do not hear an acknowledgement tone, the callback request was refused by the public exchange.

In case of a callback, if you do not pick up the receiver in time (after you have been called for 20 seconds), the callback is also cancelled.

A callback is attempted by the PBX/public exchange for up to 45 minutes. If the person called has not made a call by then, the callback is deleted automatically.

Number display and suppression

Number Display “CLIP” on Your Own Telephone

Using this feature, the caller’s number is presented on the display of a system telephone, an ISDN telephone or a CLIP-capable analog telephone. A prerequisite is that the caller present his number.

Some CLIP-capable analog telephones support the recording of the date and time in the caller list. Sometimes the display of the name instead of the number is possible (e.g., *COMfortel 500*).

HW requirements	System telephone, ISDN telephone with display or CLIP-capable analog telephone
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Analog settings* + subscriber selection in the list field at the top *COMtools* ▶ *Telephone book* ▶ *Telephone numbers*

For explanations concerning the table, see page 42

Configuration of the Feature

For a system telephone or ISDN telephone, it is not required that this feature be configured over the web interface.

Analog CLIP telephones need a specific ringer signal to detect this information. To do this, the display of the CLIP and the extended CLIP information (date, time, name) must be activated on the analog telephones in question (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Analog settings* + subscriber selection in the list field at the top).

If the call is not accepted, the telephone is blocked for further calls for 10 seconds. This CLIP timeout is necessary for some telephones but can be disabled (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Analog settings* + subscriber selection in the list field at the top).

If the telephone has no internal phone book, the name must be entered into the PBX phone book to have a name display (*COMtools* ▶ *Telephone book* ▶ *Telephone numbers*). In additional, internal PBX names are presented.

Use/Check of Features

1. Call the CLIP-capable internal telephone from an external telephone.
2. Check the number presented.

Number Display “CLIP” for the Partner Called (Number Presentation (Outgoing))

With this feature, it is possible to select the number presented to the partner called. A different number may be presented for a business or a private call or for a call as a member of a group.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

Number presentation (outgoing) can be set for each subscriber or group separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ ... or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + subscriber/group selection in the list field at the top). For a single subscriber, the settings for work and private calls can be configured separately.

Three settings are available each time:

- “None”: The telephone number is suppressed.
- “Standard”: Both the DDI parallel to the internal telephone number and the MSN on the PTMP connection are displayed on the PTP connection over which the subscriber is called (if the subscriber is not a member of the call distribution for this connection, the first MSN is presented).
- “Exchange line dependent”: The telephone number to be presented must be selected under “Expert” for each external connection separately. A single connection in this definition is:
  - A PTMP connection (only fixed network)
  - A single PTP connection (only fixed network)
  - A bundled PTP connection made up of several S_0 connections with the same PBX base numbers and the same DDI number block
  - A S_{2M} connection
  - A VoIP account

Use/Check of Features

1. Configure the number presentation (outgoing).
2. Call the CLIP-capable external telephone.
3. Check the number presented.

Dependency/Limitations

For the setting “None”, the telephone service “Calling Line Identification Restriction (CLIR)” must be released by the network provider.

On a connection with a GSM gateway, the telephone number presentation configured in the PBX is not used because the actual telephone number presentation is directly configured in the individual GSM gateway.

On a connection with CLIP no screening activated, the number presentation defined here is not considered.

Customer-defined Telephone Number Information Presentation for “CLIP no Screening”

CLIP no Screening is a telephone service for outbound calls. This function allows the party called to see a customer-defined telephone number instead of the telephone number assigned by the network provider.

In this context, “No screening” means that the customer-defined number is not checked by the central office against the assigned numbers. It may contain any telephone number you define (e.g., a service telephone number).

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *Direct dialing numbers for PTP connection*

For explanations concerning the table, see page 42

Configuration of the Feature

Enter all the MSNs/DDIs, for which you would like to present another telephone number for the party called, in the MSN/DDI table. Enter the desired telephone number for CLIP no screening into the column next to each MSN/DDI (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *Direct dialing numbers for PTP connection* or ... ▶ *PTMP connection*).

To present the telephone number for CLIP no Screening and to suppress the telephone number received by the network provider, you must activate CLIP no Screening for each connection separately (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* or ... ▶ *PTMP connection*). A single connection in this definition is:
- A single PTMP connection
- A single PTP connection)
- A bundled PTP connection consisting of several S_0 connections with the same PBX base number and the same DDI number block.

Use/Check of Features

1. Configure CLIP no Screening.
2. Call the CLIP-capable external telephone.
3. Check the telephone number presented.

Dependency/Limitations

For this function, the telephone service “CLIP no screening” must be activated with the network provider (usually for a fee).

The assigned telephone number is also presented to the public exchange - but cannot be seen by the party called.

Calling Line Identification Restriction “CLIR”

Also called “Number suppression”. You can prevent the presentation of your own number to the partner called (anonymous call).

This feature allows you to decide before starting a call if the number should be presented to the partner called or not. The number from your own connection can still be transmitted to the network provider so that the charges can be calculated correctly.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

If the number presentation only needs to be suppressed call by call, the number presentation (outgoing) must be configured as “Standard” or “Exchange line dependent”. For the “Exchange line dependent” setting, each exchange line can have an individual number presented to the network provider or to the caller separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top). The number presentation to the caller is suppressed by the telephone before starting a call. The number presentation to the network provider is not suppressed by the telephone.

If the number presentation should always be suppressed, the number presentation (outgoing) must be configured as “Exchange line dependent”. For each exchange line can a separate number be presented to the network provider. To keep these numbers from being presented to the caller, number presentation can be suppressed per exchange line separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top).

If number presentation needs to be suppressed between the PBX and the network provider, the number presentation (outgoing) must also be configured as “None” (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top).

Use/Check of Features

1. Configure the number presentation (outgoing) for call by call suppression on an internal telephone.
2. Call a CLIP-capable external telephone from the corresponding telephone (e.g., GSM mobile phone).
3. Check the number presented.
4. Call the telephone again but this time with presentation restriction. Use the function key + number or use a standard telephone to dial *31# + number.
5. Check the suppression of the number presentation.

Dependency/Limitations

For this function, the telephone service “Calling Line Identification Restriction (CLIR)” needs to be released by the network provider.

On ISDN telephones and system telephones *COMfortel 1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus*, “Calling Line Identification Restriction (CLIR)” – if available – is done using a function key or menu (e.g., with “Identification suppressed”; see the manual for the telephone).

“Calling Line Identification Restriction (CLIR)” is also possible on an analog T-Net telephone using the T-Net function key/menu, if present (this may also be known as “anonymous calling”, see the manual for the telephone).

Telephone number suppression for VoIP calls is possible but it is not always supported by the providers.

Connected Line Identification Presentation "COLP" (Number Presentation (incoming))

This feature lets you present the actual number reached to the inbound caller.

The number presented back to you may be one of the numbers from the exchange line connection reached. This number may be different than the number actually dialled by the caller. This is useful if, for example, another subscriber takes the call via pick-up or call forwarding is active.

The number to be presented back may also be suppressed (COLR).

HW requirements	– – –
SW requirements	– – –
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

Number presentation (incoming) can be set for each subscriber/group separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *...* or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + subscriber/group selection in the list field at the top).

Three settings are available:

- "None" (the number is suppressed).
- "Standard": Both the DDI parallel to the internal telephone number and the MSN on the PTM connection over which the subscriber is called are presented on the PTMP connection.
- "Exchange line dependent": The telephone number to be presented must be selected under "Expert" for each external connection separately. A single connection in this definition is:
  - A PTMP connection (only fixed network)
  - A single PTP connection (only fixed network)
  - A bundled PTP connection consisting of several S_0 connections with the same PBX base numbers and the same DDI number block
  - A S_{2M} connection

Use/Check of Features

1. Configure the number presentation (incoming) differently for two internal telephones (A and B).
2. Call an internal telephone (A) from an external ISDN telephone (C) with display.
3. Perform a pick-up for this call from another internal telephone (B).
4. Check the number presented back on the external telephone display. The number configured for telephone (B) should be displayed.

Dependency/Limitations

For the setting "no", the network provider needs to release the telephone service "Connected Line Identification Presentation (COLR)".

On a connection with a connected GSM gateway, the telephone number presentation configured in the PBX is not used because the actual telephone number presentation is directly configured in the individual GSM gateways.

Reception and Secretary Function and Announcement before Answering

Boss/Secretary Function

This function shields the boss' telephone from direct calls by redirecting these calls to the secretary's telephone. The main difference to standard call forwarding is that the boss' telephone can be reached from the secretary's telephone and important calls can be transferred from the secretary to the boss' telephone.

HW requirements	At least two system telephones *COMfortel 1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus*
SW requirements	Version 3.0C (PBX) Version 2.3C (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*System telephone*

For explanations concerning the table, see page 42

Configuration of the Feature

The following configurations need to be made with the system telephone (see the manual for the telephone):

Configure a function key as the secretary key on the boss' telephone.

Configure a function key as a boss key on the secretary's telephone.

It is recommended that you configure a speed dial key with the boss' number on the secretary's telephone in addition to the boss key to monitor the busy status.

To identify a call from the secretary using a ringer rhythm, the secretary's number may be assigned to a specific ringer rhythm in the phone book on the boss' telephone.

Use/Check of Features

1. Configure a boss key on the system telephone and a secretary key on another system telephone.
2. Press the secretary key on the boss' telephone to activate the function.
3. Call the boss' telephone.
4. The secretary's telephone rings - take the call.
5. Start a query to the boss' telephone from the secretary's telephone by pressing the boss key.
6. The call is transferred to the boss' telephone by hanging up the receiver.

Dependency/Limitations

A maximum of five secretary or boss keys per telephone are possible.

It is not possible to configure both types of keys on one telephone.

The configuration is only possible on the first level of the keys.

The configuration and usage of the Boss/Secretary function is described in detail in the manual for the system telephone.

Transferring Existing Calls

See Connect/Call Transfer “ECT” on page 51.

Waiting Loop

This function enables an indirect transfer of an external calling partner to an internal subscriber or group if the subscriber or group is busy. The external calling partner can be put into the waiting loop. There he is held for up to 3 minutes while listening to Music on Hold. When the internal subscriber in question is no longer busy, he is called for up to 60 seconds. If he takes the call, he is immediately connected to the external subscriber.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	– – –

For explanations concerning the table, see page 42

Configuration of the Feature

This feature does not need to be configured via the web interface.

Use/Check of Features

1. Make a call.
2. Make an additional call.
3. Start the transfer of the external call partner to one of the internal telephones currently in a conversation -> Busy tone and return to the call.
4. Put the external call partner into the waiting loop of the internal telephone called using R ##07 or on the system telephone with the menu (see the manual for the telephone).

Dependency/Limitations

In several ISDN telephones, the “Finish a Query” function is executed using an existing function key or using a menu (e.g., with “end”, “separating”, “back” or pressing the R key again; see the manual for the telephone).

Automatic Waiting Loop after Announcement before Answering

See Announcement before Answering (Greeting Message) on page 61.

Call Parking with Parking Zones

With this function, you can park a calling party to be transferred. This allows the desired call partner to take the call on another internal telephone.

For this function, you need to first transfer the call to a previously configured telephone number and put the call into an internal parking zone. Now the caller is on Hold in the PBX and hears Music on Hold. You can notify the desired call partner of the caller waiting. The call partner can take the call by dialling the telephone number that was used as the target for the transfer.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *Functions* ▶ *Call Parking*

For explanations concerning the table, see page 42

Configuration of the Feature

The internal telephones that put the calls into the parking zone or accept them back from there, need “Call Parking” authorization (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

To use Call Parking, a 2 to 3-digit internal base telephone number as well as the corresponding name must be configured (*COMset* ▶ *Functions* ▶ *Call Parking*). Combined with the ten possible call parking positions 0-9, ten 3 to 4-digit telephone numbers are used from the telephone numbering plan. For example, by entering the internal base telephone number 60, the telephone numbers 600-609 are used from the telephone numbering plan. By entering the internal base telephone number 610, the telephone numbers 6100-6109 are used from the telephone numbering plan. The name of the base telephone number in combination with the number of the individual call parking position is assigned as the name for the individual park positions.

The person to be transferred is only on hold in the internal call parking zone for a limited time. This on hold time can be defined from between 1 and 20 minutes (*COMset* ▶ *Functions* ▶ *Call Parking*).

Use/Check of Features

1. Configure the function.
2. Start a call on a telephone authorized for call parking.
3. Transfer the call with “R + internal base telephone number of the Call Parking” and hang up the receiver.
4. Pick up the receiver of another telephone authorized for call parking and take the call with “internal base telephone number of the Call Parking + number of the parking position”.

Transferring Exchange Line Access

With this function, an authorized telephone, for example, the reception can transfer an exchange line access to another internal telephone for one outbound call for a short time frame. This is useful if the telephone is in a public room and only used by employees from time to time for external calls.

Exchange line authorization can be used for one call only. After the end of one successful outbound call (or after the time limit has elapsed), the authorization is deleted.

HW requirements	At least one system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus*
SW requirements	Version 3.0C (PBX) Version 2.3C (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *General settings* ▶ *General* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

The internal telephone (only available on system telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus*) used for transferring exchange line access needs the authorization “Transfer of an Exchange Line Access” (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

In addition, a key for “Transfer of an Exchange Line Access” must be configured (see the manual for the telephone).

The internal telephones that should receive a transferred exchange line need at least the exchange line authorization “Inbound only, with emergency call (no VoIP)” (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top).

For the “Transfer of an Exchange Line Access”, several additional functions affecting all the telephones must be configured. First, set the exchange line settings (e.g., exchange line authorization, preferred exchange line, number presentation) to apply to the telephone after the “Transfer of Exchange Line Access”. You can configure specific exchange line settings or use the settings on the telephone granting the “Transfer of an Exchange Line Access” (*COMset* ▶ *General settings* ▶ *General*).

In addition, you can set whether it should be possible to activate a callback during the external call attempt and who will be charged for the external call. The time frame for deleting the “Transfer of an Exchange Line Access” must be configured if no successful external call is made (*COMset* ▶ *General settings* ▶ *General*).

Use/Check of Features

1. Configure the function.
2. Call the telephone authorized for “Transfer of an Exchange Line Access” from an internal telephone.
3. Take the call on this telephone and press the key for “Transfer of Exchange Line Access”. You hear the acknowledgement tone on both telephones.
4. Hang up the receiver on both telephones.
5. Start an external call on the telephone that has received the exchange line authorization.

Dependency/Limitations

The special profile “Transfer of an Exchange Line Access” cannot be assigned to a subscriber.

Waiting Field

This function creates a Waiting Field. Callers can be parked in this waiting field either automatically or by pressing a key and then be retrieved again or transferred.

The waiting field is operated using one or more Waiting Field Receptions. The system telephones *COMfortel 1500/2500/2500 AB/VoIP 2500 AB* or *COMfort 2000 plus* can be selected and configured as Waiting Field Receptions.

HW requirements	At least one system telephone *COMfortel 1500/2500/2500 AB/VoIP 2500 AB or COMfort 2000 plus*
SW requirements	Version 3.0C (PBX) Version 2.3C (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Telephone numbers* *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members* *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distributions* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ Call distributions *COMset* ▶ *Functions* ▶ Waiting Field function *COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Announcement before answering* *COMtools* ▶ *Music on Hold/announcements* ▶ *General settings* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

Configure a “Waiting Field” group and assign members to it (only system telephones *COMfortel 1500/2500/2500 AB/VoIP 2500 AB* or *COMfort 2000 plus*) that should be used as Waiting Field Receptions. To avoid user errors, all subscribers should be “logged in fixed”. (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Telephone numbers* and ... ▶ *Group members*).

Activate the Announcement before Answering function for this group (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top).

Three settings are available:

- “Off”: Announcement before Answering is switched off.
- “On busy”: The caller only hears the Announcement before Answering if all active waiting field receptions are busy. (In this case, “busy” means that there is an ongoing call in the active waiting field receptions or an outbound or inbound call is ringing. If a caller is waiting in the waiting field, this does not count as “busy”.) After the Announcement before Answering, the caller hears Music on Hold.
- “Always”: The caller always hears the Announcement before Answering. After the Announcement before Answering, he hears Music on Hold.

In addition, the ringing delay can be selected. Two settings are available:

- “Immediately”: The call is signalled immediately in the Waiting Field, that is during the greeting message.
- “After Text before Answering”: The call is signalled in the Waiting Field after the greeting message when the caller hears the Music on Hold.

If there are many Announcements before Answering (a maximum of 10 are possible), the requested Announcement before Answering must be selected for the group.

Assign the numbers in the external call distribution to the group “Waiting Field” used to call the waiting field. These calls are indicated in all active waiting field receptions (*COMset* ▶ *External numbers* ▶ *ISDN connections ...*, *COMset* ▶ *External numbers* ▶ *...* ▶ *Call distributions*).

Select the group “Waiting Field” as the group telephone number for the waiting field (*COMset* ▶ *Functions* ▶ *Waiting Field function*).

In addition, configure automatic call acceptance. Three settings are available:

- “No automatic”: Inbound calls must be put manually into the waiting field.
- “On Busy”: Inbound calls are automatically put into the waiting field if all waiting field reception telephones are busy.
(In this case, “busy” means that there is an ongoing call in the active waiting field receptions or an outbound or inbound call is ringing. If a caller is waiting in the waiting field, this does not count as “busy”.)
- “After x sec.”: Inbound calls are indicated for a duration of x seconds in the waiting field reception and then automatically put into the waiting field if not already accepted manually.

Load the Announcement before Answering as a .wav file (Format: 8 bit, A-Law, 8 kHz, Mono) into the PBX (*COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Announcement before answering*).

Adjust the volume of the announcement, if necessary (*COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*).

Keys have to be configured on the waiting field receptions selected for operating the waiting field to monitor and process exchange line calls (see the manual for the telephone). During configuration, a trunk bundle is assigned to each waiting field key. In this definition, a trunk bundle is:

- A PTMP connection (fixed network or GSM)
- A single PTP connection (fixed network or GSM)
- A bundled PTP connection formed by several S_0 connections with the same PBX base numbers and the same DDI number block
- A S_{2M} connection
- All VoIP accounts[2]

After the trunk bundle is selected, the keys are used dynamically. This offers the option to define only one key for an S_0 connection and for a S_{2M} connection only so many keys as one can handle. If inbound calls arrive over a bundle, they are displayed on the corresponding key. If there are more calls on a bundle than keys, they are not displayed until a key is available again. The maximum number of possi-

[2] Not with *COMfort 2000 plus*.

ble waiting field keys depends on the number of existing external connections/B channels and therefore on the possible number of parallel callers (a maximum of 34).

In addition, configure a waiting field key for internal calls in order to take them directly (see the manual for the telephone).

If you would like to deactivate the function for individual waiting field reception telephones, configure an additional key for switching the waiting field reception (see the manual for the telephone).

Use/Check of Features

1. Configure the waiting field reception.
2. Make an external call to the reception number.
3. Put the caller into the waiting field by pressing the yellow blinking Waiting Field key (if the caller is not already put into the waiting field automatically).
4. Accept the caller waiting by pressing the same key.

Dependency/Limitations

A detailed description of the waiting field reception is found in the manual for the system telephone.

A caller is On Hold in the Waiting Field for a maximum of 16 minutes before being disconnected. If the waiting time is not restarted by a conversation with the waiting caller or the call is transferred, the connection is disconnected.

The point of time the call is accepted (automatic or manually) depends on the ringing time. If the call rings only after “Announcement before Answering”, the caller can be put into the waiting field no earlier than after the Announcement before Answering.

The external and internal calls targeted to the group are indicated to all waiting field receptions. The external and internal calls targeted to one subscriber are indicated only in the waiting field of the corresponding subscriber.

After switching on the waiting field reception, any call forwarding (subscriber) configured for the telephone, multi-path call forwardings as well as the do-not-disturb function are switched off. Not until the waiting field reception is switched off (on the telephone) will these functions (for example, call forwarding on an answering machine) become available again. The function in question (except multi-path call forwardings) must be switched on again.

If only a single waiting field reception is required, a single subscriber may be selected instead of the group (*COMset* ▶ *Functions* ▶ *Waiting Field function*). The transfer of “Announcement before Answering” is not possible in this case.

Automatic reception may not be a member of a group of waiting field receptions.

If there is no prepared .wav file available, the Announcement before Answering can also be recorded with an internal telephone.

Automatic Reception

This function enables the automatic connection of an external caller with his requested call partner. The external caller hears an announcement (e.g., “... If you would like to reach the sales department, press 1 ...”) and can call a specific internal subscriber, a group, another Automatic Reception or an externally forwarded subscriber by dialling a suffix digit. Depending on the configuration, the suffix is a digit between 0 and 9 or also the complete internal telephone number; this must then be dialled via DTMF.

If the caller does not dial immediately, the announcement can be repeated. If the dialer does not dial within the available period, a default target may be called.

But if the caller dials an undefined number, he may be treated in different ways. Depending on the configuration, the call is disconnected, the default target is called or the caller is reconnected to the automatic reception after a short announcement.

A maximum of 10 receptions can be configured that can be switched-on in parallel or cascaded. This way it is possible, for example, to operate several automatic receptions in case of multiple companies or to configure one main reception with several sub-receptions, if necessary.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	Global release is necessary.
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members* *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distributions* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ Call distributions *COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration* *COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Automatic receptions* *COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Announcement before answering* *COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*

For explanations concerning the table, see page 42

Configuration of the Feature

A reception is configured by entering an internal telephone number (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*). Take this from the telephone numbering plan that allows you to assign 2 to 4-digit telephone numbers in the 10-9999 range. Then configure each single reception as described in the following.

Enter the individual internal target numbers as transfer targets after dialling the DTMF suffix digits 0-9 (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*). The following destination numbers can be entered:
- Internal telephone number of a subscriber
- Internal telephone number of a group
- Internal telephone number of another automatic reception

Instead of the telephone number, you can also enter an asterisk (*) for the destinations 1-9. This way, the caller can dial directly into the corresponding internal telephone numbering range 1* to 9*. If, for example, an asterisk (*) has been entered for destination 4, the caller can dial any available internal telephone number starting with a 4.

Select one of the available announcements as a greeting message that lists the possible destinations for the DTMF digit to be dialled (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*). For this function, there are up to 10 announcements available for automatic reception. If a longer speaker time is necessary, the 10 Announcement before Answering announcements can also be used. The following speaker times are possible for the individual announcements:
- Announcements for Automatic Reception no. 1-2: 30 seconds each
- Announcements for Automatic Reception no. 3-5: 20 seconds each
- Announcements for Automatic Reception no. 6-10: 10 seconds each
- Announcement before Answering: 1 minute each

Load the announcement as a .wav file (Format: 8 bit, A-Law, 8 kHz, Mono) into the PBX (*COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Announcement before answering* and ... ▶ *Automatic receptions*).

Adjust the volume of the announcement, if necessary (*COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*).

If the caller does not make an entry before the end of the announcement, the announcement may be repeated after a short waiting time. Configure the maximum number of announcement repetitions as well as the waiting time (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*).

Even if there is no entry after all the repetition are completed, the default target (if it has been configured; see above) is called.

If the entry is not correct (e.g., because no target has been configured for the entered DTMF suffix digit or the caller has dialled a non-existing telephone number), an announcement is made (if it has been configured; see below) and then the caller is reconnected to the automatic reception again.

If the called target is busy or cannot take the call (the call rings up to 120 seconds), three different behaviours can be configured (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*):

1. "End of the call": If the called destination is busy or cannot be reached, an announcement is made (if it has been configured; see below) and the connection is disconnected.
2. "Call default target": If the called destination is busy or cannot be reached, an announcement (if it has been configured) is made and then the default target (if it has been configured; see below) is called up to 120 seconds. If the default target is also not reachable, the connection is disconnected. (If the called destination and the default destination are identical, the connection is disconnected after the first 120 seconds.)
3. "Automatic reception again": If the called destination is busy or cannot be reached, an announcement (if it has been configured; see below) is made and the caller is reconnected to the automatic reception.

Enter a telephone number for the default target (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*). The following destination numbers can be entered:
- Internal telephone number of a subscriber
- Internal telephone number of a group
- Internal telephone number of another Automatic Registration

Select one of the available announcements for the announcement on Busy/on No Reply of the destination (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*). There are 10 announcements available for the for automatic receptions as well as the 10 Announcements before Answering for this function (as for the greeting message).

Load the announcement as a .wav file (format: 8 bit, A Law, 8 kHz, mono) into the PBX (*COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Announcement before answering* and ... ▶ *Automatic receptions*).

Adjust the volume of the announcement, if necessary (*COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*).

When the target telephone number is called, the caller hears the ringing. If this is not desired, Music on Hold can also be played during the ringing period. (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*).

Configure the maximum number of parallel external calls on Hold in the automatic reception considering expected reachability (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*). If this configured maximum number has been reached, the reception will not take any additional external callers at the moment but they will hear the busy tone. This limitation does not apply to internal callers and callers coming from another automatic reception.

Configure whether the individual automatic receptions (as described below) should be directly reachable externally, internally or from another automatic reception.

If the automatic reception needs to immediately take external calls, configure this in external call distribution (*COMset* ▶ *External numbers* ▶ *ISDN connections ...*, *COMset* ▶ *External numbers* ▶ *...* ▶ *Call distributions*).

If the automatic reception should only to take the call when a person is currently not able to take the call, the reception must be configured as an internal destination for Call Forwarding on No Reply or on Busy for subscribers or groups (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top).

Or you can assign the automatic reception to a group and activate the ringer delay for the automatic reception. (*COMset* ▶ *Internal telephone numbers* ▶ *Groups* ▶ *Group members*)

If a reception must be used as a sub-reception, the reception must be configured as an internal destination for another automatic reception (*COMset* ▶ *Functions* ▶ *Automatic receptions* ▶ *Configuration*).

Use/Check of Features

1. Configure the function "Automatic Reception".
2. Call the automatic reception.
3. The announcement is played for the caller.
4. Dial one of the DTMF suffix digits and check the called destination.

Dependency/Limitations

An announcement is always played to the caller from the start. So a caller may hear ringing for several seconds before he is answered, e.g., if several callers hear the announcement at the same time.

External callers who have entered a DTMF suffix digit but have not yet been connected to their destination, are considered as not connected in the reception and therefore the number of additional external calls is limited at this moment.

If the destination called by dialling a DTMF suffix digit is busy and call waiting is activated, the caller knocks. If the called party refuses the caller, the call is treated according to the configured "behaviour on Busy/on No Reply". This means the connection is disconnected, the default target is called or the caller is reconnected to the automatic reception.

The external reachability of the automatic reception can be defined depending on the configuration over call distribution. For example, to only configure an announcement or call forwarding to an answering machine during the night, e.g., the function fallback can be used (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call distributions*). Internally, the automatic reception is always reachable.

Via Call Through, an automatic reception is reachable over the internal call access "** + internal telephone number".

In case of inbound calls over VoIP, dialling a DTMF suffix digit is not supported by every provider. Make sure that DTMF suffix digits are not filtered out.

If an external destination must be reached via the automatic reception, an internal subscriber or group telephone number with active call forwarding for subscribers or groups must be defined as the target (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top). Call forwarding and multipath call forwardings active for internal call targets are also used here. The caller is forwarded depending on the features (authorizations, number presentation, call charge assignment) of the forwarded target.

The exchange line authorizations, Call Restrictor/Deblocker as well as Do-not-Disturb valid for the internal call targets are considered for the transfer. The caller may hear the busy signal and the call is treated as configured under "behaviour on Busy/on no Reply". This means the connection is disconnected, the default target is called or the caller is reconnected to the automatic reception.

In case of a call transferred by the automatic reception, the telephone number of the caller is presented to the party called (not the number of the reception).

If an external caller is transferred to the automatic reception by an internal telephone via Query, the transferring telephone hears the announcement and can select the destination itself. Then the caller is connected to the selected destination. If the transferring telephone hangs up without selecting any destination, the calling party hears the announcement himself and can select the destination.

If there is no prepared .wav file, the announcement can also recorded using an internal telephone.

Announcement before Answering (Greeting Message)

This function allows a greeting message to be played to the caller before accepting the call and/or if all group members are busy (Automatic Waiting Loop for Announcement before Answering).

The call is first accepted by the PBX (the external caller is charged from now on). The external caller hears the announcement before answering (e.g., with information about the company). Afterwards, the caller hears Music on Hold or the ringback tone again (if the line is bus, he is put into the waiting loop after the announcement until the call is answered). As soon as one of the called telephones accepts the call (or if busy, finishes the call before that), he is connected to the external caller.

HW requirements	– – –
SW requirements	Version 3.0C (PBX) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distributions* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ Call distributions *COMset* ▶ *General settings* ▶ *General* *COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Announcement before answering* *COMtools* ▶ *Music on Hold/announcements* ▶ *General settings* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

The Announcement before Answering function is activated per group (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top).

Three settings are available:

- “Off”: Announcement before Answering is switched off.
- “On busy”: The caller hears the announcement before answering only if all called group members are busy. After the announcement before answering, the caller hears Music on Hold.
- “Always”: The caller always hears the announcement before answering. In the line is busy, he is put into the waiting loop after the announcement before answering and hears Music on Hold. If the called telephones are available, the caller hears Music on Hold also while the telephones are ringing.

In addition, the length of the ringing delay can be selected. Two settings are available:

- “Immediately”: The call is signalled during the greeting message.
- “After Text before Answering”: The call is signalled after the greeting message when the caller hears Music on Hold.

If several announcements before answering are available (up to 10 are possible), you must select one for the group.

In case of a fallback (the number called is not assigned to an internal number), Announcement before Answering can also be activated (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call distributions*).

Load the Announcement before Answering as a .wav file (format: 8 bit, A law, 8 kHz, mono) into the PBX (*COMtools* ▶ *Music on Hold/ announcements* ▶ *File selection* ▶ *Announcement before answering*).

Adjust the volume of the announcement, if necessary (*COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*).

Then, select whether the caller will hear Music on Hold or the ringback tone after the announcement but before the call is accepted by the called party. (*COMset* ▶ *General settings* ▶ *General*).

On a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB,* a function key can be configured for switching the Announcement before Answering (see the manual for the telephone).

Use/Check of Features

1. Configure the “Announcement before Answering always active” function for a group.
2. Call the group from an external telephone.
3. The caller hears the announcement before answering.

Dependency/Limitations

After the announcement before answering, the caller is put on hold by the PBX for max. 8 minutes. If the call is not taken within these 8 minutes, it is disconnected. The on hold time is reduced if internal subscribers are ringing but not taking the call. Then the caller is rejected after less than 8 minutes.

Callers not put on hold in the waiting loop due to the Busy-on-Busy restriction (see Busy-on-Busy on page 48) will directly hear the busy tone instead of the Announcement before Answering.

If there is no prepared .wav file available, the Announcement before Answering can also recorded using an internal telephone (see the operation manual for the PBX).

Call Data and Charge Management

Recording Call Data

The PBX has a call data memory for storing 3000 to 9000 call data sets of external calls and that is secured against power outage.

A call data set contains the following information:

- Date at the beginning of the call
- Time at the beginning of the call
- Duration of the call
- Prefix number of the LCR provider – if used
- Number of the external call partner – if known
- Name of the external call partner – if known
- Number of the internal telephone (billing + real)
- Name of the internal telephone (billing + real)
- The external number from which the call is made
- Call charges
- Billing factor
- Call direction
- Billing type
- Call type
- Project number
- Billing number for room telephones

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	A release is necessary for more than 3000 call data sets
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Signalization by tones* + subscriber selection in the list field at the top *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* COMlist ▶ Call data/charges ▶ Acquisition *Administration* ▶ *Dongle releases*

For explanations concerning the table, see page 42

Configuration of the Feature

If you would like memory for more than 3000 call data sets, you can activate it by using the required activation code (*Administration* ▶ *Dongle releases*).

In the case of call data memory overflow, you can allow additional calls to be recorded by activating the setting “Overwrite the memory automatically” (*COMlist* ▶ *Call data/charges* ▶ *Acquisition*).

If this is not what you want, you must delete the call data memory over the telephone before any new data can be recorded. To be notified early that this is the case, a special dial tone can be configured for certain subscribers when call data memory usage limits have been exceeded (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Signalization by tones* + subscriber selection in the list field at the top).

To avoid filling the memory too quickly, you can configure what kind of calls should be recorded. (*COMlist* ▶ *Call data/charges* ▶ *Acquisition*).

To provide privacy, there is the option of truncating the recorded numbers for private calls in contrast to business calls (*COMlist* ▶ *Call data/charges* ▶ *Acquisition*).

If the charges for the calls are to be recorded, transmission of the call charges must be activated on the external S_0 ports (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* or *...* ▶ *PTMP connection*). Deactivating this function is only advisable if it causes problems (this depends on the network provider/country).

Use/Check of Features

1. Configure the requested call data recording.
2. Make one or more external calls.
3. Check the call data now under *“COMlist* ▶ *Call charge data list”*.

Dependency/Limitations

The function “Delete list of single call records” allows, for example, a member of the works council to completely delete the number of his call partner in the call data to prevent this from being viewed by unauthorized persons. The data part necessary for billing remains untouched. To use this function, a special authorization is necessary for the subscriber (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

Important: *As a protection measure against the access by third parties, regularly check the call data recording of your PBX and the LOGs of your NAT router for inconsistencies.*

Call Data Management via Web Interface (COMlist)

The call data stored in the PBX may be viewed, sorted, filtered printed, exported and deleted over the web interface.

HW requirements	– – –
SW requirements	Version 2.2 E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMlist* ▶ *Print options* *COMlist* ▶ *Filter* ▶ *Configure* *COMlist* ▶ *Call data/charges* ▶ *General settings*

For explanations concerning the table, see page 42

Configuration of the Feature

Configure the currency name and price per unit (*COMlist* ▶ *Call data/charges* ▶ *General settings*).

Select the table columns or the available information for the call you would like to see and change the column names, if necessary (*COMlist* ▶ *Print options*).

Configure additional filters for the call data list (*COMlist* ▶ *Filter* ▶ *Configure*).

Use/Check of Features

1. Open the call data view under *“COMlist* ▶ *Call charge data list”*.
2. Select the desired filter.
3. Configure the number of lines per page.
4. Change the sort order by clicking the column headers.
5. Select the desired action (e.g., create export file) in the action line and click the Execute button.

Dependency/Limitations

Call data that has not been recorded due to the settings in *“COMlist* ▶ *Call data/charges* ▶ *Acquisition”* is not displayed.

Call Data Printout

The call data stored in the PBX may not only be evaluated by a PC but can also be read out directly to a printer. As soon as the printout has been enabled by a “start command”, the call data are read out continuously immediately after the end of the call. If the printer must be disconnected from the PBX for a time, the “stop command” entered over the telephone prevents the data sets stored in this time frame from being lost.

HW requirements	serial Printer
SW requirements	Version 2.2 E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMlist* ▶ *Print options* *COMlist* ▶ *Call data/charges* ▶ *Acquisition*

For explanations concerning the table, see page 42

Configuration of the Feature

Configure the parameters “19200 Baud, 8 bit, no parity, 1 stop bit, no protocol” for the interface of the printer.

Activate the permanent call data display (*COMlist* ▶ *Call data/charges* ▶ *Acquisition*).

Format the printout by selecting whether you want to separate the data using one of the following symbols (semicolon, space, tabulator) or if you want to print the data as a table with a fixed column width (*COMlist* ▶ *Call data/charges* ▶ *Acquisition*).

The data in the display, which are separated by a symbol, are printed out in the correct length.

For a printout with a fixed column width, this length is specified by the maximum length of the column title. If the data contained are shorter than the specified column width, the remaining column area is filled with spaces. For long data strings (telephone numbers), the number of characters above the specified length are cut off from the left. The defined column widths are as follows:

Consec.No.	5 characters + 1 space
Date	8 characters + 1 space
Time	8 characters + 1 space
Duration	8 characters + 1 space
Subscriber No. Invoice	16 characters + 1 space
Subscriber name for invoice	16 characters + 1 space
Subscriber No. real	16 characters + 1 space
Subscriber name real	16 characters + 1 space
Connection No.	12 characters + 1 space
External partner	16 characters + 1 space
External name	16 characters + 1 space
Project	6 characters + 1 space
LCR number	5 characters + 1 space
Charges	6 characters + 1 space
Calculation factor	5 characters + 1 space
Direction	8 characters + 1 space
Calculation type	8 characters + 1 space
Call type	8 characters + 1 space
Hotel record number	5 characters

Define whether the printout should have a header consisting of the column title (*COMlist* ▶ *Call data/charges* ▶ *Acquisition*). If this is the case, adapt the column title, if necessary (*COMlist* ▶ *Print options*).

Select the table columns or the information available about the call that you would like to print (*COMlist* ▶ *Print options*).

Use/Check of Features

1. Configure the call data printout.
2. Enter the “start command” with “##8*271#”. Already accrued call data is now printed.
3. Make an external call. Immediately after the call, the data for this call are printed.

Dependency/Limitations

Start and stop commands are described in the Operating Manual.

Call data that has not been recorded due to the settings in *“COMlist ▶ Call data/charges ▶ Acquisition”* is also not printed.

For printing the call data in a hotel, use the special hotel print function (see Print Function on page 99).

Online Name Search

The online name search lets you look instantly for the name assigned to an existing telephone number on a server on the Internet. When a call is received, the PBX searches the telephone books (PBX, system telephone) for the telephone number transmitted. If no suitable name entry is found, the name assigned to the telephone number is transferred from the Internet to your telephone – as far as the server on the Internet is accessible and the desired data is stored there. The name is shown on the display of *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB* or *COMfort 1000/1200/2000 plus* system telephones and saved in the “External partner” column on the call data list.

You can also update the call data list after the fact with the help of the online name search by adding missing names or perhaps to update existing names. Click the magnifying lens symbol in the column “External partner” of the call data list to start a backward search. If an entry is found, the name and perhaps the address of the calling partner is displayed. It is possible to edit this entry before transferring it into the column “external name” (truncated to a maximum of 16 characters) and store it in the data set or in all data sets with the corresponding telephone number as well as into the PBX telephone book.

Note: *The updated online name search (PBX firmware version 3.2 and higher) initially performs a targeted search for the telephone number using the default settings of the default providers. If this telephone number cannot be found (e. g. extension number of a PBX), the telephone number will be shortened and searched for until a result was found. This function can be compared to the function Display approx. results of the online name search under firmware version as of 3.2.*

To be able to view the result of the online name search, characters in the determined name may have to be converted. The text converter needed for this task can be configured.

Click the book symbol in the column “external name” of the call data list to transfer the telephone number and name of the external calling partner into the PBX telephone book.

HW requirements	Broadband Internet connection (e.g., DSL connection and router)
SW requirements	Version 3.0C (PBX) Version 2.3C (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...)
Dongle release	– – –
Configuration via / Setting via	*COMset ▶ Functions ▶ Online Name Search ▶ Configuration* *COMset ▶ Functions ▶ Online Name Search ▶ Text converter* *Administration ▶ Server configuration*

For explanations concerning the table, see page 42

Configuration of the Feature

In order to resolve the names of Internet addresses, the PBX needs the address for a DNS server (just like with a PC connected to the Internet). Enter the address provided by your system administrator or by your Internet service provider (*Administration ▶ Server configuration*). You may also enter a second address just in case the main DNS server is unavailable.

If the DNS server is accessed via a proxy, this must be activated, and the corresponding address as well as the port must be entered (*Administration ▶ Server configuration*).

Save the default providers or configure at least one service provider in the PBX for the online name search (*COMset ▶ Functions ▶ Online name search ▶ Configuration*). Select the desired service provider (*COMset ▶ Functions ▶ Online name search ▶ Configuration*).

Activate online name search (*COMset ▶ Functions ▶ Online name search ▶ Configuration*).

If required, configure a text converter to be able to display the results of the online name search correctly (*COMset ▶ Functions ▶ Online name search ▶ Text converter*).

Use/Check of Features

1. Configure the online name search.
2. Open the call data view under *COMlist ▶ Call charge data list.*
3. Search a data set with an existing external telephone number in the column “External partner” and start the backward search for this telephone number by clicking the magnifying lens symbol.

Dependency/Limitations

If the PBX is being operated as a sub-system, the Online Name Search function cannot be used.

The name that is found is truncated to 16 characters if it is shown on the display or if it is automatically added to the call data list.

If a name for the presented telephone number has already been entered in the telephone book (or on a speed dialling key) of a system telephone, this name is displayed. But the newly found name is stored in the call data list anyway.

If a name for the transferred telephone number has already been entered into the PBX telephone book, this name is stored in the call data list. But it can be updated by a manual backward search using the magnifying lens symbol.

Depending on how quickly a call has been accepted, the name is only visible on the display during the call.

Project Assignment

The exchange line access with project assignment allows the sorting of external calls to a specific project/customer/client (e.g., within a law firm). With call data management, the charges and the time expenditures may be assigned to different projects/customers/clients.

HW requirements	– – –
SW requirements	Version 3.0C (PBX) Version 2.3C (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	Global release is necessary.
Configuration via / Setting via	*Administration* ▶ *Dongle releases* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

Activate the function using the required activation code (*Administration* ▶ *Dongle releases*).

A configuration of this feature over the web interface is not absolutely necessary.

A project list can be configured in a system telephone *COMfortel 1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus.*The user can select the project from this list - also using the name (see the manual for the telephone).

A function key for selecting a project or a single project can also be configured on a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus* (see the manual for the telephone).

Use/Check of Features

1. Make an external call using the project assignment by entering “##93 Project number * (0) number” (the project number may be 2 to 6-digit).
2. Check the call data under *COMlist* ▶ *Call charge data list*.

Dependency/Limitations

On a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus,* the project number is entered using a configured function key or menu (see the manual for the telephone).

On a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus,* the project number can be selected not only before the call but also during a call (see the manual for the telephone). This makes the assignment of inbound calls (for billing the time expenditure) possible.

Separating Private Calls from Business Calls

Private (personal) exchange line access allows the separate billing of business and private calls to the individual employees. Calls started with private exchange line access receive a special token in the call data recording. This assigns the charges to the employee.

The employee that would like to make private calls from his work station receives a user PIN assigned to his telephone. All calls started with the private exchange line access and this PIN are assigned to him in the call data management. This also applies to calls made from another internal telephone (e.g., his colleague’s telephone).

If special authorization is assigned to the subscriber, private calls can be made without entering a user PIN but only on the subscriber’s personal telephone.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top *COMlist* ▶ *Call data/charges* ▶ *General settings* *COMlist* ▶ *Call data/charges* ▶ *Acquisition* *Administration* ▶ *User PINs*

For explanations concerning the table, see page 42

Configuration of the Feature

Configure the exchange line authorization, Call Restrictor and Deblocker (outgoing), short-code dialling authorization, preferred exchange line and number presentation differently for private and business calls, if necessary (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top).

Enter a user PIN for the internal telephones (*Administration* ▶ *User PINs*).

If call allowance accounts have been configured, you can select whether they are to be used for business and/or private calls (*COMlist* ▶ *Call data/charges* ▶ *General settings*).

To provide privacy, there is the option of truncating the recorded numbers for private calls in opposed to business calls (*COMlist* ▶ *Call data/charges* ▶ *Acquisition*).

If the subscriber should be able to make private calls without entering a user PIN but only on his own telephone, you must assign authorization (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

Use/Check of Features

1. Start an external call with the private exchange line access “##92 user PIN * (0) number”.
2. Check the call data under *“COMlist* ▶ *Call charge data list”*.

Dependency/Limitations

If you allow private calls without PIN, there is no protection against misuse by other internal users.

Announcements via Speaker and System Telephone

InterCom Announcement and Handsfree via System Telephones

This function enables an announcement to system telephones (individual telephone or group) from any available internal telephone without actively taking the call at the target telephone, e.g. in a doctor's office.

Alternatively, you can instruct an individual system telephone to switch on the microphone in addition to the loudspeaker (handsfree operation) so that a person nearby can talk to you via the built-in intercom.

HW requirements	At least one system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus*
SW requirements	Version 3.8A (PBX) Version 2.3C (system telephone COMfort ...) Version 4.2A (system telephone COMfortel ...) Version 2.6.0 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Special functions* + subscriber selection in the list field at the top System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

On the target system telephone, the InterCom permission must be activated (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Special functions* + subscriber selection in the list field at the top).

On a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus,* a function key can be configured for starting an InterCom call (see the manual for the telephone; InterCom call to a group not with *COMfort 1000/1200/2000 plus*).

Use/Check of Features

1. Activate InterCom permission on a system telephone.
2. Start InterCom announcement to this system telephone with "##011 number"

   or: Start InterCom handsfree mode to this system telephone with "##012 number"

   (start the procedure from another system telephone using the function key)
3. The LED next to the handsfree operation/loudspeaker key flashes on the called system telephone and the call is accepted after a single ringing tone.

Dependency/Limitations

InterCom announcement and handsfree are limited to 120 seconds. After that the connection will be interrupted automatically unless the receiver of the system telephone will be picked up within this period.

In case of an InterCom announcement to a group, a voice connection will be set up even to group members not logged in. The precondition for this is that the InterCom permission is activated on the system telephones.

Announcement via Speaker

This function enables an announcement to a loudspeaker device or an active loudspeaker (e.g., ELA system in a department store or supermarket) connected to the audio output (Cinch jack marked with "audio output") of the PBX.

A previously configured internal telephone number is called from an internal telephone. The voice connection is established right after dialling the telephone number.

HW requirements	A *COMmander 2TSM analog Module and a connected speaker system*
SW requirements	Version 3.0C (PBX) Version 2.3C (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *Door/switching/music module* ▶ *Telephone numbers* *COMset* ▶ *Door/switching/music module* ▶ *Relay settings* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

An internal number must be assigned to the audio output to be called (*COMset* ▶ *Door/switching/music module* ▶ *Telephone numbers*). Select this from the telephone number plan that permits the allocation of 2 to 4-digit telephone numbers in the 10-9999 range.

The individual telephones must receive authorization for speaker announcements (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

You can also configure one or more relays on the module used for the operation mode announcement to switch the speaker system on during the announcement, if necessary (*COMset* ▶ *Door/switching/music module* ▶ *Relay settings*).

On a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus,* a function key can be configured for starting a speaker announcement (see the manual for the telephone).

Use/Check of Features

1. Configure the audio output and assign the authorization for speaker announcements to a telephone.
2. Call the internal number of the audio output with the corresponding telephone.
3. Make an announcement.

Save Money and Get Information Away from Home

Call Through

This function enables the use of the PBX functions Least Cost Routing and VoIP/GSM Routing, for example, by the travelling sales man. To avoid high costs when making, for example, an international call with the mobile phone, you can call the PBX first and be connected to the requested target. Function:

① The user uses his mobile telephone to dial an external telephone number on the PBX assigned to Call Through.

② The PBX checks the Call Through authorization by checking the number presented by the mobile phone.

③ If the Call Through authorization is verified, the PBX accepts the call and sends a special tone.

④ The user dials (DTMF) the target number on his mobile phone. This is dialled – after checking it against the exception number table (VoIP and GSM Routing (Exception Telephone Numbers) on page 89) and against the Least Cost Routing table (Least Cost Routing with Soft-LCR easy on page 84) – by the PBX then and a connection is established.

The greatest savings can be achieved if the PBX telephone number has been selected as a favourite special telephone number with the mobile phone provider.

HW requirements	– – –
SW requirements	Version 2.2 E (PBX)
Dongle release	Release is necessary for more then four simultaneous calls
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *Location* *COMset* ▶ *Functions* ▶ *Call Through* COMtools ▶ Telephone book ▶ Telephone numbers COMtools ▶ Telephone book ▶ Call Through user *Administration* ▶ *Dongle releases*

For explanations concerning the table, see page 42

Configuration of the Feature

For more then four simultaneous calls, activate the function using the required activation code (*Administration* ▶ *Dongle releases*).

The mobile phone numbers (Call Through users) permitted for Call Through must be entered into the telephone book (*COMtools* ▶ *Telephone book* ▶ *Telephone numbers*). The corresponding telephone number must be presented by the mobile telephone.

For this telephone book entry, Call Through must be enabled (Call Through authorization); *COMtools* ▶ *Telephone book* ▶ *Call Through user*).

The external telephone number used by the Call Through user to call the function Call Through, can be selected for each individual external connection separately (*COMset* ▶ *Functions* ▶ *Call Through*). When selecting "---" however, individual connections can also be restricted for Call Through. A single connection in this definition is:
- A PTMP connection (fixed network or GSM)
- A single PTP connection (fixed network or GSM)
- A bundled PTP connection comprising several S_0 connections with the same PBX base numbers and the same DDI number block
- A S_{2M} connection

The telephone numbers are selected by selecting an existing MSN or by entering a DDI in the available DDI number block. The extension number is added automatically to the list of extensions for the Point-to-Point connection.

For each external connection, the number of Call Through calls permitted at the same time can be selected. On a PTMP connection or on a simple PTP connection, two calls are possible at the same time. (*COMset* ▶ *Functions* ▶ *Call Through*).

Which exchange lines used for the connection desired by the Call Through user, which telephone number is presented and which exchange line authorizations apply can be selected in the special exchange line settings for Call Through (*COMset* ▶ *Functions* ▶ *Call Through*). The common Call Restrictor/Deblocker selected in the special exchange line settings for Call Through can be activated dependent on the user (*COMtools* ▶ *Telephone book* ▶ *Call Through user*).

In addition, you can global select whether LCR is active for Call Through calls for all Call Through users (*COMset* ▶ *Functions* ▶ *Call Through*).

For security reasons, the PBX limits the duration of external Call Through calls. The maximum duration may be configured from between 1 and 99 minutes (*COMset* ▶ *Functions* ▶ *Call Through*).

The telephone numbers in mobile phone phone books are most often entered with a plus (+) in front of the country code (e.g., +49). The PBX can accept and use this format. Configure the country code for this purpose (*COMset* ▶ *External numbers* ▶ *Location*).

Use/Check of Features

1. Configure Call Through and enter a mobile phone number in the phone book as an authorized number.
2. Call the Call Through number with that mobile phone.
3. After the short tone, dial an external number with area code over DTMF.
4. Make this call.

Dependency/Limitations

Only the telephone numbers on an ISDN connection can be used as external telephone numbers for using the Call Through function. When making a call using VoIP, there is no guarantee that the displayed telephone number is the actual telephone number. The Call Through user can therefore not be detected with certainty.

Group/Team Functions

Assign, Log in, Log out

In addition to internal subscribers, the PBX is able to manage 64 groups with up to 20 members (subscribers and other groups) each. These groups can be used to reach, e.g., the internal subscribers of certain departments/teams (support, marketing, sales). Internal subscribers and groups can be a member of more than one group.

Your membership in a group does not mean that you are always called when your group is called. If you do not want to be reachable for a certain time via the group number but only as an individual subscriber, you are able to "log out" of this group.

Group members can be active and passive. This function can be important for members of service lines or call centres who do not want to be available for their customers around the clock. Group functionality is not available for a subscriber for the time he is logged out. There are three different modes to log in:

- Incoming + outgoing
- Incoming only
- Outgoing only

A subscriber that is member of several groups can only be "logged in outgoing" in one single group at a time. In this case, if he would also like to receive the calls for the other groups, he can log into them as "incoming". This puts him in the call distribution of multiple groups for internal, public exchange and door terminal calls.

With "log in outgoing" into a certain group, the subscriber receives a number of features/authorizations from the group which replace his own features/authorizations as an individual subscriber for outbound business calls (e.g., exchange line authorizations).

Subgroups (i.e. groups that have been configured as members of other groups) are "permanently logged in as incoming" and "permanently logged out as outgoing" in the corresponding main group.

HW requirements	– – –
SW requirements	Version 3.8A (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Telephone numbers* *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members* *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Profiles* ▶ *Profile assignment* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distributions* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ Call distributions

For explanations concerning the table, see page 42

Configuration of the Feature

Like an internal subscriber, a group has an internal number and its own profile defining the base settings of the group (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Telephone numbers* and ... ▶ *Profile assignment*). Select this from the telephone number plan that permits the allocation of 2 to 4-digit telephone numbers in the 10-9999 range.

External call distribution must be configured for the group (*COMset* ▶ *External numbers* ▶ ... ▶ *Call distributions*).

Each group can have up to 20 members. One subscriber or group can be a member of multiple groups (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members*).

The log in/out may be performed by telephone. A preset is also possible. The subscriber can be logged in permanently to a group (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members*).

Use/Check of Features

1. Configure a group with at least one subscriber.
2. Log in the subscriber into the group.
3. Call the group from another telephone. The logged-in subscriber rings.

Call Forwarding for Groups

See Call Forwarding for Groups on page 45.

Ringing Sequence

It may not be always useful, e.g., in a service centre, for all the telephones logged into a group to ring simultaneously. To solve this problem, a ringing sequence can be configured.

HW requirements	– – –
SW requirements	Version 3.8 A (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members* *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

For a group, you can configure how an inbound call rings for logged-in group members. For this purpose, a ringing sequence and a time between 5 and 60 seconds must be defined (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top).

In addition to ringing sequences, group member priority must be assigned (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members*).

The ringing sequence is created by the following settings:

A. "All": All subscribers logged into the group ring simultaneously. It is possible to configure a delayed ringing for selected subscribers. These subscriber start to ring after the delay. Moreover, if the subscribers are busy and their call waiting function is activated, all subscribers will hear knocking. The call is terminated by the network provider (usually after 120 sec.).
If CF on no response was configured, the group call is finished after the configured delay time and the CF target is called instead.

B. "Linear": All members logged into the group ring directly one after the other for a configured time. For one call, only one telephone rings at a time. A busy subscriber is skipped and the next subscriber in line is called.
If CF on no response was configured, the group call is finished after the configured delay time and the CF target is called instead

C. "Adding": All members logged into the group start to ring delayed by a configured time one after the other. A busy subscriber is skipped and the next subscriber in line is called.
If CF on no response was configured, the group call is finished after the configured delay time and the CF target is called instead.

a. "Fixed": The order telephones are called depends on the assigned priority.

b. "Balancing": The order telephones are called is managed by a dynamic priority list. After each accepted call, the corresponding telephone is moved to the last position in line.

x. "Rotating": After the first loop (all logged-in telephones have been/are ringing) is complete, it is started again. In the second loop, the busy subscribers hear knocking. The call is terminated by the network provider (usually after 120 sec.).

_. Without "rotating": After the first loop (all logged-in telephones have been/are ringing) is complete, the call is disconnected. Busy subscribers hear no knocking.

y. "Split Group": During the call phase, the group needs to be reachable for further callers. Group splitting makes sure that approximately the same number of subscribers is available for each of the upcoming calls.

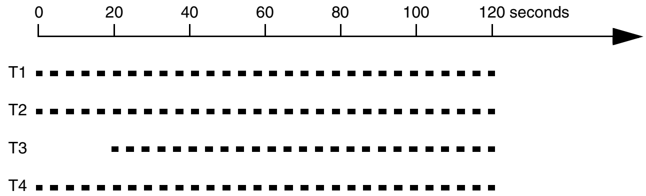
_. Without "Split Group": A second caller hears the busy signal (only possible for "All").

Below are example of calls illustrating the possible options. The following parameters are common to all examples:
- Number of telephones logged in: 4
- Configured time: 20 seconds
- Priority order: T1, T2, T3, T4
- Ringing delay for ringing sequence "All": Activated for T3
- CF delay time for CF on no Reply: 20 seconds

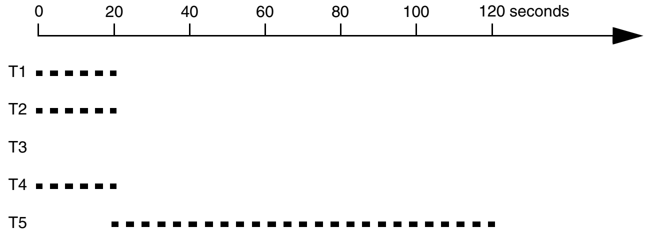
A. "All"
(Like "All - Split Group", but a second caller hears the busy signal.)

A. y. "All - Split Group"

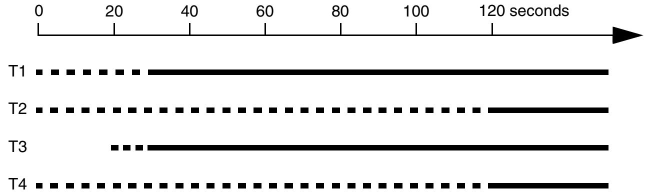
One call:



One call (CF on no Reply active to target telephone T5):

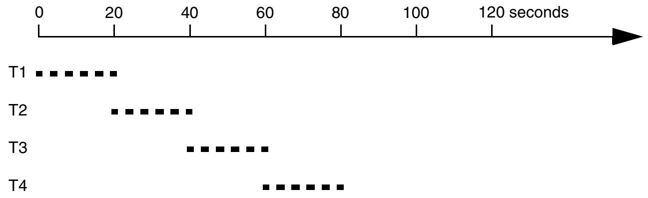


Two calls A ▪▪▪▪ and B ▬▬ (call B starts 30 seconds delayed):

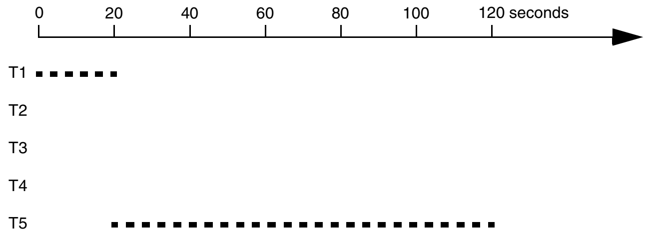


B. a. y. “Linear - fixed - Split Group”

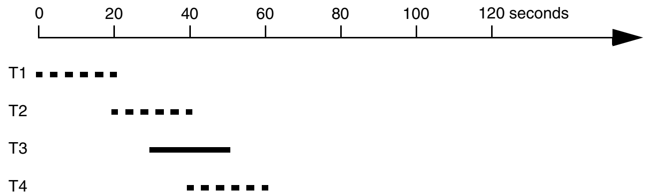
One call:



One call (CF on no Reply active to target telephone T5):

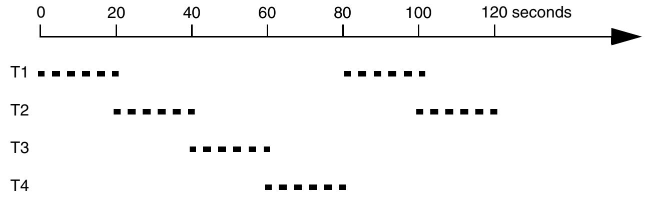


Two calls A and B (call B starts 30 seconds delayed):

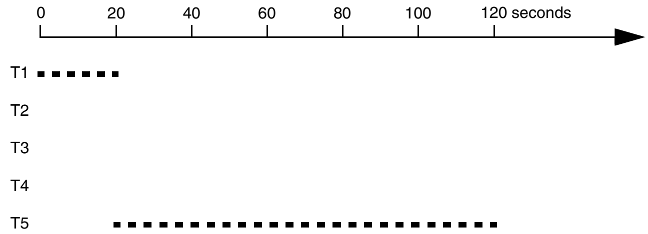


B. a. x. y. “Linear - fixed - rotating - Split Group”

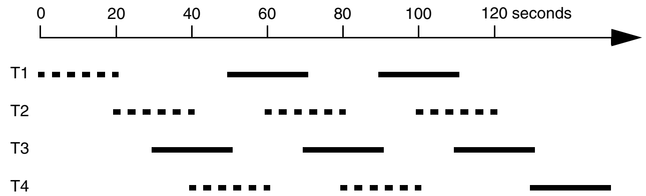
One call:



One call (CF on no Reply active to target telephone T5):



Two calls A and B (call B starts 30 seconds delayed):



B. b. y. “Linear - balancing - Split Group”

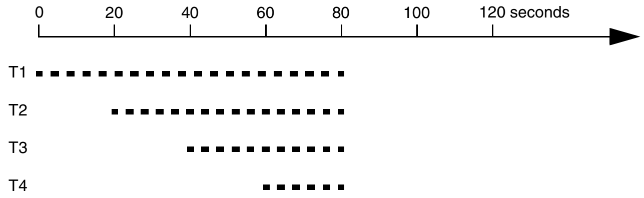
(This is different than “Linear - fixed - Split Group” due to the changing priorities only.)

B. b. x. y. “Linear - balancing - rotating - Split Group”

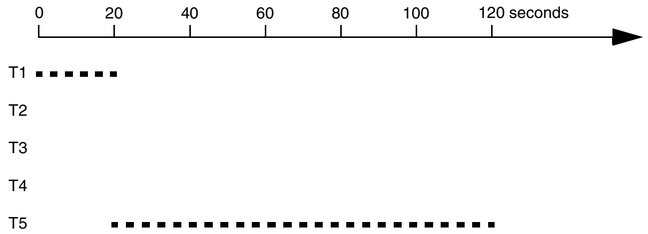
(This is different than “Linear - fixed - rotating - Split Group” due to the changing priorities only.)

C. a. y. “Adding - fixed - Split Group”

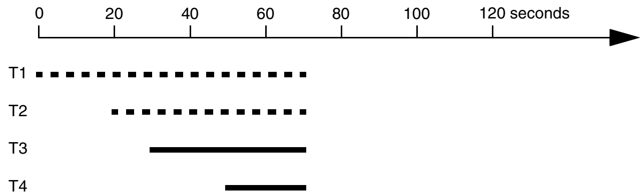
One call:



One call (CF on no Reply active to target telephone T5):



Two calls A and B (call B starts 30 seconds delayed):

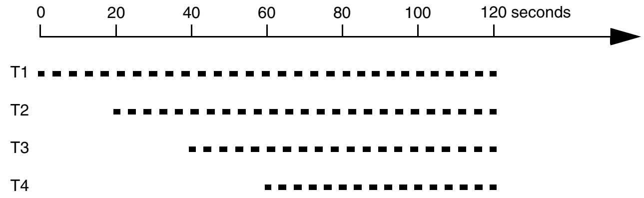


C. b. y. “Adding - balancing - Split Group”

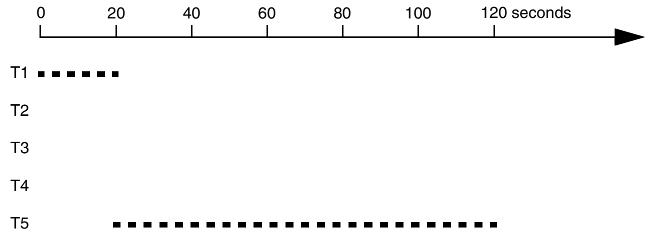
(This is different than “Adding - fixed - Split Group” due to the changing priorities only,)

C. a. x. y. “Adding - fixed - rotating - Split Group”

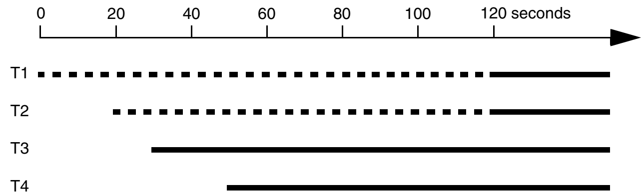
One call:



One call (CF on no Reply active to target telephone T5):



Two calls A and B (call B starts 30 seconds delayed):



C. b. x. y. “Adding - balancing - rotating - Split Group”

(This is different than “Adding - fixed - rotating - Split Group” due to the changing priorities only.)

Use/Check of Features

1. Configure a group with several members.
2. Log in all members into the group.
3. Call the group from another telephone. Keep the call ringing to check the ringing sequence.

Dependency/Limitations

If a group (subgroup) has been configured as member of a group (main group), the following special rules apply:

If a subscriber is at the same time member of the main group and of the subgroup, he will only be considered as a member of the main group when a call comes in.

The ringing scheme configured for the specific group applies. If the same ringing schemes have been configured for the main group and the subgroup, the subscribers will behave as if they were members of a single large group. If, for example, “Linear” has been set for the main group and “All” for the subgroup, the members of the main group will ring one after the other and the members of the subgroup (as soon as it is their turn) will ring simultaneously.

In the case of a second call to the main group, it will be checked both for the main group and for the subgroup whether splitting is possible (“Split Group”). If this is the case, the subgroup will be considered for both calls and the members of the subgroup will be divided among both calls.

CF on no Reply configured for a group (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Call Forwarding* + group selection in the list field at the top) is started after the configured CF delay time. As soon as the CF target is called, the group call is terminated.

With the variants "Linear" and "Adding", a busy subscriber is skipped and the next subscriber in line is called. With the setting "rotating" after the first loop (all available telephones have been/are ringing once), the busy subscribers hear the knocking tone - if allowed.

If a second call comes in with the variant "Linear" without "rotating" while the last subscriber is ringing, the caller hears the busy tone.

The alarm call delay time (*COMset* ▶ *Door/switching/music module* ▶ *Alarm settings*) between alarm subscribers is independent of the group ringing sequences. With the variants "Linear" and "Adding", it is possible that the last members of a group are not called because the next alarm subscriber has already been called.

After a PBX reset/reboot, the priorities assigned in the configuration are used for "balancing".

Central Caller Lists for Groups

The telephone numbers of unaccepted calls are stored in the caller list on the system telephone COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB and DECT 900/900C.

As soon as a connection (inbound or outbound) has been made to a caller, the corresponding telephone number is automatically deleted from the caller list. If a group call occurred and if the central caller list for groups is activated, the telephone number is also deleted from the caller lists of the other members of the group.

HW requirements	At least two system telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB* or *DECT 900/900C*
SW requirements	Version 4.0 (PBX) Version 3.6C (system telephone COMfortel ...) Version 00.17.70 11.01 (COMfortel DECT 900 Base)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members*

For explanations concerning the table, see page 42

Configuration of the Feature

The main caller list can be activated for each group separately (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members*).

Use/Check of Features

1. Configure a group consisting of system telephones COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB and DECT 900/900C.
2. Activate the central caller list for the group.
3. Call the group from another telephone. Do not accept the call.
4. Call this telephone number from the caller list with one of the called system telephones (see the manual for the telephone). Accept the call at the target.
5. Check the deletion of the entry from the caller list on the other system telephones (see the manual for the telephone).

Dependency/Limitations

COMfortel DECT 900/900C: The subscriber calling back is logged in incoming and outgoing.

Timer and Wake-up Functions

Setting the PBX Time

he PBX has an internal clock which transfers the saved time and date to the connected telephones. If the function is implemented in the telephone, the PBX time is shown on the display.

Your PBX needs the PBX time for the functions Wake-up and configuration switching.

To ensure that, after a power failure, the PBX time becomes available again, there are various options for automatically updating it – over ISDN and over the network.

HW requirements	– – –
SW requirements	Version 3.0C (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *General settings* ▶ *Date/Time of the PBX*

For explanations concerning the table, see page 42

Configuration of the Feature

The internal clock on the PBX (PBX time) can be set manually. Instead of making manual entries, the date and time can also be transferred from the PC. In both cases, it is necessary to indicate the time zone in addition to the time and date, for example, whether to change automatically to summer/winter time (*COMset* ▶ *General settings* ▶ *Date/Time of the PBX*).

In addition, it is possible to adjust the PBX time automatically – via ISDN and via the network.

When adjusting via ISDN, the network provider generally transfers a time that is exact-to-the-minute in the case of an outbound external call. This time is used to set the internal system time. The time is adjusted only once an hour (*COMset* ▶ *General settings* ▶ *Date/Time of the PBX*).

When making adjustments via the network, the time can be made available by a local or an external NTP server, for which the IP address or server name (e.g., ntp1.ptb.de) must be entered. This time can be extremely exact and is often compared and adjusted to official time servers. An adjustment of the time is done in the time intervals selected here (Update after x hours) (*COMset* ▶ *General settings* ▶ *Date/Time of the PBX*).

Use/Check of Features

1. Set the time.
2. View the time on a system telephone.

Dependency/Limitations

For the time display on the telephone, there are different synchronizations depending on the type of telephone:

- For system telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus:* Automatically
- For Standard ISDN telephones: During successful outbound calls
- For *COMfortel 500* and other analog telephones: With each inbound call

Timer-driven Configuration Switching

The PBX lets you set some functions differently for day, night, weekend, holidays and public holidays. Eight configurations are available. The following functions are configuration dependent:

- Subscriber profile assignment (and therefore, e.g., exchange line authorization, number presentation and call forwarding for subscribers)
- Group profile assignment (and therefore, e.g., announcement before answering, exchange line authorization, number presentation and call forwarding for groups)
- Profil assignment of the voice mail/fax boxes (and therefore, e.g., readiness)
- Fallback options for the external ports (fallback number and delayed fallback)
- Call distribution and door call distribution
- Relays with operation mode “configuration dependent”
- Call forwarding for external numbers

Switching from one configuration to another may be done time-dependent using an internal clock or manually by internal or external subscribers (telephones with authorization).

HW requirements	COMmander VMF module (for voice mail and fax boxes)
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Profiles* ▶ *Configuration* *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Profiles* ▶ *Profile assignment* *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Profiles* ▶ *Configuration* *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Profiles* ▶ *Profile assignment* *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profiles* ▶ *Configuration* *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profiles* ▶ *Profile assignment* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distributions* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ Call distributions *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *PTP connection* *COMset* ▶ *Configurations* ▶ *Create configurations* *COMset* ▶ *Configurations* ▶ *Switching times* *COMset* ▶ *Configurations* ▶ *Copy configurations* *COMset* ▶ *Configurations* ▶ *Copy switching times* *COMset* ▶ *Functions* ▶ *Calendar* ▶ *Holidays* *COMset* ▶ *Door/switching/music module* ▶ *Relay settings* *COMset* ▶ *Door/switching/music module* ▶ *Call distributions*

For explanations concerning the table, see page 42

Configuration of the Feature

Create the required number of configurations and assign each configuration a name as well as an identification number (*COMset* ▶ *Configurations* ▶ *Create configurations*).

Now configure the individual configurations as described in the following. To simplify the procedure, you can copy the settings of the first configuration to the others first (*COMset* ▶ *Configurations* ▶ *Copy configurations*).

To assign configuration-dependent features to the individual subscribers, first configure the necessary subscriber profiles. (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Profiles* ▶ *Configuration*). Then assign the subscriber profile to the existing configurations (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Profile assignment*).

To assign configuration-dependent features to the individual groups, configure the necessary group profiles (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Profiles* ▶ *Configuration*). Then assign the group profile to the existing configurations (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Profile assignment*).

To assign configuration-dependent features to the voice mail/fax boxes, configure the necessary profiles (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profiles* ▶ *Configuration*). Then assign the existing profiles to the voice mail/fax boxes (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Profiles* ▶ *Profile assignment*).

To assign external calls to the internal subscribers, groups and receptions depending on the configuration, configure external call distribution including the fallback telephone numbers and the fallback delay for the existing configurations (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call distributions*).

To route door calls to internal and external call targets depending on the configuration, configure the door call distribution for the existing configurations (*COMset* ▶ *Door/switching/music module* ▶ *Call distributions*).

In order to set the relays as configuration-dependent, configure the relays with the operation mode "Configuration-dependent" and assign them to the individual configurations (*COMset* ▶ *Door/switching/music module* ▶ *Call distributions*).

To set call forwarding for external numbers as configuration-dependent, configure the call forwarding for the existing configurations (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *...*).

To switch the defined configurations as time-dependent, activate automatic configuration switching (*COMset* ▶ *Configurations* ▶ *Create configurations*).

Enter the holidays with irregular switching times compared to the days of the week into the list of holidays. For this purpose, you can enter individual dates (e.g., factory holidays) and/or transfer a list of holidays from the PBX and edit it (*COMset* ▶ *Calendar* ▶ *Public holidays*).

Enter configuration switching times for the different days of the week as well as for the holidays. A total of up to 100 switching times are possible (*COMset* ▶ *Configurations* ▶ *Switching times* ▶ ...).

If possible, copy the switching times from one day of the week to the other (*COMset* ▶ *Configurations* ▶ *Copy switching times*).

Use/Check of Features

1. Enter configurations and switching times.
2. Check switching times under *COMset* ▶ *Configurations* ▶ *Overview switching times*.
3. Wait for the switching time to another configuration.
4. Check the current configuration with a system telephone (see the manual for the telephone).

Dependency/Limitations

In addition to the automatic time-dependent configuration switching via internal clock it may be started manually via telephone. For this purpose an identification number has to be assigned to the configuration (*COMset* ▶ *Configurations* ▶ *Create configurations*). The subscriber needs an authorization to switch the configuration manually (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Authorizations* + subscriber selection in the list field at the top).

All holidays entered and activated in the PBX are treated the same concerning the switching times/configurations. The day of the week is not considered during the holidays. Therefore in the holiday list activate only these holidays that have to use special switching times/configurations different to the days of the week.

Permanent Configuration

For situations where your settings are supposed to apply even after configuration switchovers, you can use the so-called permanent configuration. If permanent configuration is enabled, the configuration-dependent settings will be overridden by the permanent configuration settings.

Permanent configuration is available for the following settings:

- Call forwarding for external numbers
- Readiness of the central voice mailbox

HW requirements	– – –
SW requirements	Version 3.8AC (PBX) Version 4.2A (system telephone COMfortel ...) Version 2.6.0 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *PTP connection*

For explanations concerning the table, see page 42

Configuration of the Feature

To override configuration-dependent call forwarding, configure call forwarding for the permanent configuration and enable permanent configuration for the corresponding external numbers (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call Forwarding* ▶ *...*).

On a system telephone COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB, you can configure function keys for switching on/off/over the call forwarding of external numbers – either for the current configuration or for permanent configuration (see the manual for the telephone).

Use/Check of Features

1. Configure call forwarding in the permanent configuration and enable the permanent configuration.
2. Call the forwarded number.
3. Depending on the type of call forwarding activated, the call should reach the target configured.

Dependency/Limitations

On a system telephone COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB, you can switch on/off/over the call forwarding of external numbers either for the current configuration or for permanent configuration using the function key (see the manual for the telephone). With a standard telephone you can change call forwarding in the permanent configuration and switch on the permanent configuration or the current configuration using a digit sequence. As soon as every call forwarding has been switched off in the permanent configuration, the permanent configuration for the corresponding telephone number is switched off and the configuration-dependent settings apply. Tip: To override the configuration-dependent settings with no call forwarding, enable call forwarding in the permanent configuration without specifying a destination number.

Wake-up Function

To be reminded of single or recurrent dates, every subscriber can configure different wake-up times for his telephone.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Configurations* ▶ *Switching times* *COMset* ▶ *Functions* ▶ *Calendar* ▶ *Holidays* *COMtools* ▶ *Wake-up times* ▶ *Wake-up times subscriber* *COMtools* ▶ *Wake-up times* ▶ *General settings* *COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Wake-up announcements* *COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*

For explanations concerning the table, see page 42

Configuration of the Feature

You can define how often a Wake-up call is repeated per Wake-up time. In addition to this, it is possible to configure the duration of the Wake-up call and the gap between single Wake-up calls. The Wake-up procedure ends earlier if a Wake-up call has been acknowledged by taking the receiver off-hook (*COMtools* ▶ *Wake-up times* ▶ *General settings*).

You can define how long one single Wake-up call rings on the telephone (*COMtools* ▶ *Wake-up times* ▶ *General settings*).

You can activate the Wake-up call recording in the call data memory (*COMtools* ▶ *Wake-up times* ▶ *General settings*).

If an announcement should be played after accepting a Wake-up call, you can load up to three Wake-up announcements as a .wav file (format: 8 bit, A Law, 8 kHz, mono) into the PBX (*COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Wake-up announcements*).

If necessary, adjust the volume of the announcement (*COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*).

The use of the different Wake-up announcements must be configured for the different times of the day (*COMset* ▶ *Configurations* ▶ *... switching times*).

You can configure Wake-up calls for different subscribers, if necessary. Enter the time and weekday or a group of weekdays. Activate the Wake-up time and select whether the Wake-up call should happen “Repeatedly” (on the defined weekdays until deactivated) or once (*COMtools* ▶ *Wake-up times* ▶ *Wake-up times subscriber*).

On activated holidays, you can enable the Wake-up call for a Sunday instead using the Wake-up call for that weekday (*COMset* ▶ *Functions* ▶ *Calendar* ▶ *Holidays*).

Use/Check of Features

1. Configure a Wake-up call.
2. Check the Wake-up call activation under *COMtools* ▶ *Wake-up times* ▶ *Overview*.
3. Make sure the Wake-up call functions properly.
4. Take the receiver and listen to the Wake-up announcement.

Dependency/Limitations

On some analog telephones able to show CLIP information alphanumerically, the message “Wake-up call” is displayed. This text can be changed under *“COMset* ▶ *General settings* ▶ *CLIP texts”*.

If no prepared .wav file is available, the Wake-up announcement may also be recorded using an internal telephone.

Update Functions

Regular firmware updates via the update server

Using your PBX and the connected system telephones, you can perform a regular (for example, yearly) automatic firmware update. In order to keep the update from disrupting normal telephone operation, an update is preferably done at night.

The PBX attempts to download the current firmware from the server during a set period of four hours. If the update could not be performed during the preset period of time, the PBX attempts to do it again at the same time the following day.

During the update, the PBX automatically connects to a server over an Internet connection and downloads the current firmware; call charge data sets and configuration data remain intact. In addition, an update is performed "via the exchange line" using a system telephone of any design; the new firmware is then "distributed" to the other system telephones.

HW requirements	Broadband Internet connection (e.g., DSL connection and router)
SW requirements	Version 3.0C (PBX) Version 2.0.08 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*Administration* ▶ *Server Configuration* *Administration* ▶ *Firmware update/Reboot* COMfortel Set

For explanations concerning the table, see page 42

Configuration of the Feature

The address for the PBX update server is saved in the PBX. If necessary, this can be changed on the page *Administration* ▶ *Server configuration*.

In order to resolve the names of Internet addresses (here, the address of the update server), the PBX needs the address for a DNS server (just like for a PC connected to the Internet). Enter the address provided by your system administrator or by your Internet service provider (*Administration* ▶ *Server Configuration*). You may also enter a second address just in case the main DNS server is unavailable.

If the connected system telephones should also be updated, activate the corresponding setting (*Administration* ▶ *Firmware update/reboot*).

The telephone number for the system telephone server is saved in the system telephones using the PC programme (see the manual for the telephone).

For the regular automatic update a time interval (every 12 weeks, twice a year, yearly) must be selected. Set the time for carrying out the update during a reasonable time frame so that normal telephone operation is not disrupted (*Administration* ▶ *Firmware update/Reboot*).

Finally, activate of all the settings by clicking "Accept".

Use/Check of Features

The time the last update attempt was made and the time the last successful update took place are displayed under "Update status" on the page *Administration* ▶ *Firmware update/reboot* .

Dependency/Limitations

The first update is carried out after the set interval (in 12 weeks, in six months or in a year). If the existing firmware is not current, you can also carry out an immediate firmware update via the update server (*Administration:* ▶ *Firmware update/reboot*).

During the distribution process, "Server Mode" is shown on the display panel of the distributing system telephone. It is not possible to use the system telephone during this process.

Aside from the charges resulting from the required telephone connection (after first going to System telephone -> Server), the update is free of charge.

If different system telephones are connected to a PBX, the firmware update "Via the exchange line" must be performed on a COMfortel 1100/1500, on a COMfortel 2500 and on one of the older system telephones (COMfort 1000, COMfort 1200 and COMfort 2000 plus). After this, the firmware can be distributed from these telephones to the others (see the manual for the telephone).

Prefer certain Subscribers or S_0 Ports

B Channel Reservation

If a group of subscribers should have the option of making an external call at any time, the B channels on an external ISDN connection can be reserved for the group.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

To reserve the B channels on an external ISDN connection, the external connection must be selected per group. Then the number of reserved B channels is entered (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top).

Use/Check of Features

1. Configure a B channel reservation for a group.
2. Attempt to make the connection busy with some telephones that are not members of the corresponding group. As soon as the number of free B channels has been exceeded, you should receive a busy tone.

Dependency/Limitations

To be able to use the reserved B channels for transmitting external calls, the outbound subscriber in question must be logged into the group.

Preferred Exchange Line

If a telephone should use primarily specific external connections, several preferred exchange lines for business and/or private calls may be configured for the corresponding group or subscriber.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

A preferred exchange line can be set for each subscriber/group separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *...* or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + subscriber/group selection in the list field at the top). For a single subscriber, the settings for work and private calls can be configured separately. Two settings are available:

- “No preferred exchange line”: For outbound external calls, any available fixed network connection is used.
- “Special preferred exchange lines”: For outbound external calls, the connections released under “Expert” are used depending on the selected priority. This makes the definition of a preferred exchange line (priority = 1) with overflow to other connections (priority > 1) possible.
  If all released connections are busy, it is no longer possible to call. The subscriber cannot use a connection with the setting “No preferred exchange line” for outbound calls. A single connection in this definition is:
  - A PTMP connection (fixed network or GSM)
  - A single PTP connection (fixed network or GSM)
  - A bundled PTP connection comprised of several S_0 connections with the same PBX base numbers and the same DDI number block
  - A S_{2M} connection
  - A VoIP account

Use/Check of Features

1. Configure the preferred exchange line for a subscriber.
2. Check the correct exchange line access with an outbound external call, e.g., via the LEDs on the S_0 modules or via busy lights on a configured system telephone or by checking the presented number.

Dependency/Limitations

The same priority may be allocated to different external connections in parallel.

Use Security Functions

Emergency Numbers and Priority Function

The emergency priority circuit makes a public exchange line available for dialling an emergency number or for tripping an alarm at any time. If all lines are busy, an existing call is terminated for the emergency call.

To save time in case of an emergency, an emergency number can be dialled with or without the Exchange Line Access Number “0”. The advantage is that even persons without knowledge of exchange line access numbers can dial these numbers without any problems.

Even subscribers that do not have permission to make external calls are allowed to dial essential numbers to make emergency calls.

HW requirements	– – –
SW requirements	Version 2.2 E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Emergency call*

For explanations concerning the table, see page 42

Configuration of the Feature

The function “Emergency call has priority” (emergency priority circuit) must be activated for the whole PBX. Up to ten external numbers may be entered as emergency numbers (*COMset* ▶ *Internal numbers* ▶ *Emergency call*).

Exchange line authorization “with emergency call authorization” can be set for each subscriber/group separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *...* or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + subscriber/group selection in the list field at the top). For a single subscriber, the settings for work and private calls can be configured separately. The exchange line authorizations “International”, “National” and “City” include emergency call authorization.

Use/Check of Features

Do not use a “real” emergency number such as 112 (in Germany) when checking the feature.

Dependency/Limitations

When entering numbers without an area code as emergency numbers, make sure that these numbers do not conflict with internal numbers.

The emergency numbers 110 and 112 are pre-configured in the default factory settings. This means that internal subscriber numbers may not start with the digits 11. If the subscriber number 11 must be used, the emergency numbers 110 and 112 must be deleted.

Currently, emergency calls over VoIP (see Voice over IP (VoIP) on page 87) cannot completely be guaranteed by all providers. This has to do with the connection with the emergency centre as well as providing the geographic location of the caller which is so important in case of an emergency call. Because of this, the PBX prefers to route emergency calls (that is, the calls to telephone numbers registered under *COMset* ▶ *Internal numbers* ▶ *Emergency call* as well as the alarm calls automatically made by the PBX) via fixed network connections. If all fixed network connections are busy, another call is disconnected to establish this emergency call in case of an activated emergency call priority – even if the VoIP accounts and the GSM gateways are not busy. If there are no fixed network connections available at all (external ISDN ports are all connected to GSM gateways), emergency calls are made over a GSM gateway.

dialling emergency numbers via VoIP accounts is manually possible by selecting a special exchange line with the account number.

Alarm Functions

If the PBX alarm functions are used in case of an alarm, internal and external telephones can be called and sirens and alarm indication lights can be switched on. The alarm procedure can be as follows:

① Alarm detection at the alarm input (contact closed longer than ½ of a second).

② The set alarm delay time is running (0-99 seconds).

③ Non-recurring start of the siren (for 1-6000 seconds) and/or the alarm indication light (for 1-6000 seconds).

At the same time, the first alarm call run starts (alarm calls to the alarm subscribers) by calling the first alarm subscriber for approx. 60 seconds. Internal alarm subscribers are called with a special ringer rhythm (telephone-dependent). After picking up the receiver, you repeatedly hear an alarm announcement stored in the PBX. The called party must acknowledge the alarm call within 60 seconds by sending the DTMF number 0

④ If the called party does not acknowledge the alarm call, the next alarm subscriber is called a few seconds after the first call is finished. And so on....

⑤ If all the alarm subscribers have been called without receiving an acknowledgement, the alarm loop is repeated after a configurable alarm waiting period (0-99 seconds) has elapsed (up to 9 times).

⑥ The alarm is terminated (including siren and alarm indication light) as soon as one of the alarm subscriber acknowledges the alarm. If the alarm input has been configured to “active once”, it is deactivated afterwards. If the alarm input has been configured to “active repetitive”, it is activated again and a new alarm can be triggered.

HW requirements	*COMmander 2TSM analog Module*
SW requirements	Version 2.2 E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Emergency call* *COMset* ▶ *Door/switching/music module* ▶ Port configuration *COMset* ▶ *Door/switching/music module* ▶ Alarm settings *COMset* ▶ *Door/switching/music module* ▶ Relay settings COMtools ▶ Music on Hold/announcements ▶ File selection ▶ Music on Hold/announcements *COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*

For explanations concerning the table, see page 42

Configuration of the Feature

On one of the ringer inputs available on the COMmander 2TSM analog module, set the operating mode Alarm (*COMset* ▶ *Door/switching/music module* ▶ *Port configuration*). An alarm can then be triggered with the corresponding wiring (e.g., output for a leak warning device).

Select one of the available alarm announcements for operating the alarm (*COMset* ▶ *Door/switching/music module* ▶ *Alarm settings*).

Load the emergency announcement as a .wav file (format: 8 bit, A Law, 8 kHz, mono) into the PBX (*COMtools* ▶ *Music on Hold/announcements* ▶ *File selection* ▶ *Music on Hold/announcements*).

Adjust the volume of the announcement, if necessary (*COMtools* ▶ *Music on Hold/announcements* ▶ *General settings*).

Depending on how often and how long an alarm should sound, the values for alarm runs, alarm delay time as well as alarm waiting time can be changed (*COMset* ▶ *Door/switching/music module* ▶ *Alarm settings*).

Up to four internal and external telephone numbers can be entered as alarm subscribers (*COMset* ▶ *Door/switching/music module* ▶ *Alarm settings*).

To control a siren and/or an alarm indication light, a relay must be set to the operation mode siren or an alarm indication light with the requested duration timing (*COMset* ▶ *Door/switching/music module* ▶ *Relay settings*).

Turning on the emergency call priority switch (emergency call has priority) is recommended for calling external subscribers. This means that when an alarm is triggered by the PBX, an exchange line call is always made possible. If all the lines are busy, a live call is ended so that the emergency call can be made (*COMset* ▶ *Internal numbers* ▶ *Emergency call*).

Activate the alarm (*COMset* ▶ *Door/switching/music module* ▶ *Alarm settings*).

Use/Check of Features

1. Configure and activate an alarm with an alarm call to internal subscribers.
2. Trigger the alarm.
3. Accept alarm call and acknowledge it with “0”.

Dependency/Limitations

If no .wav file is available, the alarm announcement may be recorded using an internal telephone.

You can also de/activate the alarm with an internal or an external telephone.

If you want to configure a second alarm run, a second COMmander 2TSM analog module is required. You cannot use the other second ringer inputs for this purpose.

Some analog telephones able to display CLIP information alphanumerically show the message “Alarm call” while ringing. This text can be changed under *“COMset* ▶ *General settings* ▶ *CLIP texts”*. If a name has been entered for the identification number of the alarm, this is displayed again (*COMset* ▶ *Door/switching/music module* ▶ *Alarm settings*).

Currently, emergency calls over VoIP (see Voice over IP (VoIP) on page 87) cannot completely be guaranteed by all providers. This has to do with the connection with the emergency centre as well as providing the geographic location of the caller which is so important in case of an emergency call. Because of this, the PBX prefers to route emergency calls (that is, the calls to telephone numbers registered under *COMset* ▶ *Internal numbers* ▶ *Emergency call* as well as the alarm calls automatically made by the PBX) over fixed network connections. If all the fixed network connections are busy, another call is disconnected to establish this emergency call in case of an activated emergency call priority – even if the VoIP accounts and the GSM gateways are not busy. If there are no fixed network connections available at all (external ISDN ports are all connected to GSM gateways), emergency calls are made via a GSM gateway.

Baby Call/Hotline

With the automatic dialling function (Baby Call/Hot Line), you can configure your telephone to automatically dial one telephone number.

With the automatic dialling function, you can configure your telephone to automatically dial an internal or external number. This starts after a delay time you can configure after picking up the receiver. It is not necessary to press any key. If you start to dial manually within the configured delay time, this has priority.

HW requirements	– – –
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top *COMset* ▶ *General settings* ▶ *General*

For explanations concerning the table, see page 42

Configuration of the Feature

Activate automatic dialling and enter an internal or external target number (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

Enter the “Delay time for baby/senior call” in seconds. (*COMset* ▶ *General settings* ▶ *General*)

Use/Check of Features

1. Configure automatic dialling on a telephone.
2. Pick up the receiver. The configured number is dialled automatically.

Dependency/Limitations

Note that if you start to dial manually, some telephones do not dial immediately after pressing the key (sometimes delayed by $^{1}/_{2}$ second).

The Baby Call/Hotline may be configured directly from a telephone if this setting is not controlled by a profile.

On a standard VoIP telephone (e.g. COMfortel VoIP 250) the automatic dialling function (Baby Call/Hotline) is not possible.

Room Monitoring

A room can be acoustically monitored by internal or external callers if room monitoring has been enabled for a telephone in that room.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *General settings* ▶ *Remote configuration* Internal telephone to be monitored

For explanations concerning the table, see page 42

Configuration of the Feature

Room monitoring can only be configured on the monitoring telephone itself. Pick up the receiver and dial ##8*61#. Put the receiver beside the telephone.

For external room monitoring, remote programming and MSN/DDI switching as well as the external PIN must be configured (*COMset* ▶ *General settings* ▶ *Remote configuration*). You can select an available MSN or select an existing bundle and enter a DDI assigned to the bundle. The extension number is added automatically to the list of extensions for the Point-to-Point connection.

Use/Check of Features

1. Configure room monitoring for a telephone.
2. Pick up the receiver from another telephone. Dial the number of the previously configured telephone. Listen in the room.

Dependency/Limitations

It is not possible to make a call from the configured telephone and it cannot be called normally.

It is only possible to enable a single internal PBX telephone for room monitoring at a time.

To deactivate room monitoring, hang up the receiver on the telephone.

Limit Telephone Charges (Cost Control)

Exchange Line Authorizations

If you want to prevent the high costs of dialling expensive numbers from accruing on the connection, you can limit the dialling options of individual subscribers and groups using exchange line authorizations.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top

For explanations concerning the table, see page 42

Configuration of the Feature

Exchange line authorization can be set for each subscriber/group separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *...* or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + subscriber/group selection in the list field at the top). For one subscriber, the settings for work and private calls can be configured separately. The following settings are available:

- “International”: All national and international telephone numbers can be dialled.
- “National”: All national telephone numbers can be dialled.
- “City”: All telephone numbers without an area code can be dialled.
- “Inbound only, with emergency call (no VoIP)”: Emergency numbers can be – except via VoIP accounts – dialled and incoming calls can be accepted.
- “Internal only, with emergency call (no VoIP)”: Emergency numbers can be – except via VoIP accounts – dialled but no incoming calls can be accepted.
- “Internal only, no emergency calls”: No external calls can be made.
- “Exchange line dependent”: The exchange line authorization must be set under “Expert” for each external connection separately. A single connection in this definition is:
  - A PTMP connection (fixed network or GSM)
  - A single PTP connection (fixed network or GSM)
  - A bundled PTP connection comprised of several S_0 connections with the same PBX base numbers and the same DDI number block
  - A S_{2M} connection
  - A VoIP account

Use/Check of Features

1. Configure exchange line authorization “City” for a telephone.
2. Pick up the receiver of the corresponding telephone and dial a number with an area code. You hear the busy tone.

Dependency/Limitations

When dialling a local number (telephone number without area code) via VoIP and GSM connections, the area code entered under *COMset* ▶ *External numbers* ▶ *Location* is automatically dialled as a prefix.

Currently, emergency calls over VoIP (see Voice over IP (VoIP) on page 87) cannot completely be guaranteed by all providers. This has to do with the connection to the emergency centre as well as providing the geographic location of the caller which is so important in case of an emergency call. Because of this reason, existing VoIP accounts are not used for emergency calls (the calls to telephone numbers entered under *COMset* ▶ *Internal numbers* ▶ *Emergency call*). Therefore when selecting “Inbound only, with emergency call (no VoIP)”, only inbound calls can be accepted via VoIP accounts. When selecting “Internal only, with emergency call (no VoIP)”, no external calls can be made via VoIP accounts.

Call Deblocker (outgoing) – Release Numbers

To extend the dialling options of a restricted exchange line authorization with specific numbers (numbers of certain telephones or certain area codes), you can add Call Deblockers to exchange line authorizations.

As call deblockers have a higher priority than call restrictors, restricted numbers or parts of them can be released again, for example, if 0180 numbers have been blocked by entering and activating a call restrictor. Enter the 0180 6 number in a call deblocker and perform the following activation to release these numbers again.

Up to 10 call deblockers with a maximum of 100 released numbers total may be configured (e.g., 10 call deblockers with up to 10 released numbers or 5 call deblockers with up to 20 released numbers).

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top *COMset* ▶ *Functions* ▶ *Call Through* *COMtools* ▶ *Telephone book* ▶ *Call Through user* *COMtools* ▶ *Special numbers* ▶ *Call deblocker (outgoing) - released numbers*

For explanations concerning the table, see page 42

Configuration of the Feature

The requested call deblockers (10 max.) must be created and filled with released numbers (a max. total of 100) (*COMtools* ▶ *Special numbers* ▶ *Call deblocker (outgoing) - released numbers*.

After this, a call deblocker can be selected and activated for each subscriber/group separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *...* or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + subscriber/group selection in the list field at the top). For a single subscriber, the configuration can be made for business and private calls separately.

A common call restrictor can be selected for the Call Through users in the special exchange line settings for Call Through only (*COMset* ▶ *Functions* ▶ *Call Through*), which can be activated per user (*COMtools* ▶ *Telephone book* ▶ *Call Through user*).

Use/Check of Features

1. Enter a number into a call restrictor and into a call deblocker and activate both for a telephone.
2. Pick up the receiver on the corresponding telephone and dial the entered number. You hear the ringing tone.

Call Restrictor (outgoing) – Restricted Numbers

You can limit the exchange line authorization for subscribers, groups and Call Through users: The numbers that should not be dialled can be entered into up to 10 different Call Restrictors. A total of up to 100 restricted numbers may be entered (e.g., 10 call restrictors with up to 10 restricted numbers or 5 call restrictors with up to 20 restricted numbers).

In general, numbers are blocked starting with a certain digit sequence that should not be available to every subscriber due to high charges, e.g., 0900 numbers. Normally, the first 4 to 5 digits are relevant for the costly service.

If a restricted number is dialled, the busy tone is heard. If the restricted numbers are not activated (that is, “not restricted”), a restriction according to the exchange line authorization defined is still possible.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top *COMset* ▶ *Functions* ▶ *Call Through* *COMtools* ▶ *Telephone book* ▶ *Call Through user* *COMtools* ▶ *Special numbers* ▶ *Call restrictors (outgoing) - restricted numbers*

For explanations concerning the table, see page 42

Configuration of the Feature

The required call restrictors (10 max.) must be created and filled with restricted numbers (a max. total of 100) (*COMtools* ▶ *Special numbers* ▶ *Call restrictors (outgoing) - restricted numbers*).

After this, a call restrictor can be selected and activated for each subscriber/group separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *...* or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + subscriber/group selection in the list field at the top). For a single subscriber, the configuration can be made separately for business and private calls.

A common call deblocker can be selected for the Call Through users in the special exchange line settings for Call Through only (*COMset* ▶ *Functions* ▶ *Call Through*), which can be activated per user (*COMtools* ▶ *Telephone book* ▶ *Call Through user*).

Use/Check of Features

1. Enter a number into a call restrictor and activate it for a telephone.
2. Pick up the receiver on the corresponding telephone and dial the number entered into the call restrictor. You hear the busy tone.

Short-code dialling Authorization

To extend the dialling options of a restricted exchange line authorization with specific numbers, you can add short-code dialling authorization to the exchange line authorizations.

In this way, special numbers can be released for dialling. With short-code dialling authorization, all short-code numbers can be dialled independently of the exchange line authorization.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top *COMtools* ▶ *Telephone book* ▶ *Telephone numbers*

For explanations concerning the table, see page 42

Configuration of the Feature

Short-code authorization can be set for each subscriber/group separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top) or (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + subscriber/group selection in the list field at the top). For a subscriber the configuration can be made for business and private calls separately

The desired short-code numbers (400 max.) must be created (*COMtools* ▶ *Telephone book* ▶ *Telephone numbers*). Select them from the telephone number plan that permits the allocation of 2-digit to 4-digit telephone numbers in the 10-9999 range.

Use/Check of Features

1. Enter a number into a call restrictor and into the telephone book as a short-code dialling number and then activate the call restrictor and the short-code dialling authorization for a telephone.
2. Pick up the receiver on the corresponding telephone and dial the number entered. You hear the ringing tone.

Calling Charge Information During and at the End of the Connection “AOCD, AOCE” (Charge Pulse)

To monitor the charges accrued on the connection and also on the internal telephones, the charges are recorded by the PBX for each subscriber separately. From time to time, the charge sum may be reviewed on the corresponding telephone and deleted, if necessary. The deletion has no influence on the call charges recorded in the call data memory of the PBX.

To check the charges during a call or directly afterwards, the charges are transmitted to ISDN telephones unchanged at the moment they are accrued even if the receiver is on-hook. For an analog telephone with a charge meter, you can define how to convert the charges transmitted via ISDN into charge pulses.

With AOCE, charge information is not transmitted until the end of the connection.

HW requirements	Telephones with charge meter display
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Analog settings* + subscriber selection in the list field at the top *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMset* ▶ *General settings* ▶ *General* *COMlist* ▶ *Call data/charges* ▶ *General settings*

For explanations concerning the table, see page 42

Configuration of the Feature

Charge information must be activated on the external S_0 ports (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* or ... ▶ *PTMP connection*).

To utilize charge information, analog telephones with charge meters need charge pulses. These can be switched on separately per subscriber by selecting the feature AOCD or AOCE (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Analog settings* + subscriber selection in the list field at the top).

The charge pulse frequency is adjustable, if necessary, under *“COMset* ▶ *General settings* ▶ *General”* (depending on the country settings of the PBX).

The setting “Call charge recording changes with subscriber” (*COMlist* ▶ *Call data/charges* ▶ *General settings*) cause the charges accrued to be assigned to the internal telephone connected from the time the external calls are transferred. On the exchange line, the telephone service “Advice of Charge during the connection (AOCD)” is necessary, otherwise all charges are assigned to the last subscriber.

Use/Check of Features

1. Start a chargeable call with an analog telephone that supports charge pulses.
2. Check the display of charges.

Dependency/Limitations

Note that some telephone service providers do not transmit the charges. If this type of telephone service provider is used (by manually selecting or using the LCR procedure Soft-LCR easy) for telephoning, the subscriber cannot be charged fees. If the LCR procedure Soft-LCR 4.0 is used, the charges are calculated based on the length of the call and the tariff tables set up separately and, if necessary, converted into charge pulse units.

Call Allowance Account

With a call allowance account, you can configure a specific amount of charge units for each subscriber. Once these charge units are spent, no external call is possible from this telephone until the Call Allowance is refilled or redefined.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMtools* ▶ *Call allowance accounts* ▶ *Call allowance accounts* *COMtools* ▶ *Call allowance accounts* ▶ *General settings*

For explanations concerning the table, see page 42

Configuration of the Feature

Charge information must be activated on the external S_0 Ports (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* or ... ▶ *PTMP connection*).

You can set for all the subscribers together whether the call allowance accounts for business and/or private calls should be charged. The currency identifier and the price per unit are also set for all the subscribers together (*COMtools* ▶ *Call allowance accounts* ▶ *General settings*).

You can configure a call allowance account for each subscriber individually with a manually or automatically recharged balance. The maximum amount for each account can be configured for all subscribers together (*COMtools* ▶ *Call allowance accounts* ▶ *Call allowance accounts*).

Use/Check of Features

1. Configure an account of 0 euro for a subscriber.
2. Try to make an external call with the corresponding telephone. You hear the busy tone.

Dependency/Limitations

The call allowance account will not function if a VoIP account is being used for external calls.

To use the function “Call Allowance” without restrictions, you need the telephone service “Advice of charge during the call (AOCD)” for your exchange line. With “Advice of charge at the end of the call (AOCE)”, the PBX cannot determine until the end of the call if the call allowance has been exceeded.

Note that some telephone service providers do not transmit the charges. If this type of provider is used (through manual selection or with the LCR procedure Soft-LCR easy) for making telephone calls, a call allowance account that has previously been set up does not work for the call in question, for example. To avoid this, the provider telephone numbers in question should be entered as restricted numbers and this function should be activated for every telephone.
If the LCR procedure Soft-LCR 4.0 is used, the charges are calculated based on the length of the call and the tariff tables set up separately. For this purpose, you need to specify a price per minute as well as the metering pulse time frame per switching time and provider (*Routing* ▶ *Soft-LCR 4.0* ▶ *LCR tables* ▶ *...* ▶ *View with tariff*).

External VoIP Calls

See Voice over IP (VoIP) on page 87.

Least Cost Routing with Soft-LCR easy

To telephone easily and without hassle using the most cost-effective telephony provider, the PBX offers automatic Least Cost Routing (LCR). To activate LCR in the PBX, the function must be configured and the current tariff structure of the telephony provider used must be transferred into the PBX tariff tables.

You may choose from two different LCR procedures:

- Soft-LCR easy[4] has an update service available (for a fee)[3].
- If you use Soft-LCR 4.0, you yourself are responsible for keeping the tariff information up-to-date (see Least Cost Routing with Soft-LCR 4.0 on page 85).

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	For more then 16 LCR subscribers, a release is necessary in increments of 8
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMset* ▶ *Functions* ▶ *Call Through* *Routing* ▶ *LCR subscriber* *Routing* ▶ *LCR procedure* *Routing* ▶ *Soft-LCR easy* ▶ *Setup* *Administration* ▶ *Dongle releases*

For explanations concerning the table, see page 42

Configuration of the Feature

Activate the function using the required activation code (*Administration* ▶ *Dongle releases*).

Select the LCR procedure Soft-LCR easy (*Routing* ▶ *LCR procedure*).

Depending on the number of LCR subscribers released in the dongle, telephones must be selected that should use automatic Least Cost Routing (*Routing* ▶ *LCR subscriber*).

Soft-LCR easy[4] must be installed, activated and configured (*Routing* ▶ *Soft-LCR easy* ▶ *Setup*). During this procedure, at least one update (chargeable) is necessary.

If a GSM gateway is connected or the network provider does not support LCR, you can activate LCR for each individual external connection separately (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* and ... ▶ *PTMP connection*). A single connection in this definition is:
- A PTMP connection
- A single PTP connection

[3] The update is made via the added value number (0 90 09) 00 00 561 (1,86 Euro/ minutes – pulse length 2 seconds; valid until 31.12.10). The price is a German fixed network price. There is a maximum of 0.93 euros per connection.

- A bundled PTP connection comprised of several S_0 connections with the same PBX base numbers and the same DDI number block
- A S_{2M} connection

Furthermore, you can set whether LCR for Call Through calls should be active for all Call Through users together (*COMset* ▶ *Functions* ▶ *Call Through*).

Use/Check of Features

1. Configure LCR and activate it for a telephone.
2. Start an external call on the corresponding telephone.
3. Check the call data under *COMlist* ▶ *Call charge data list*.

Dependency/Limitations

The update is chargeable.[5]

To use the update method on an ISDN connection inside of Germany, the following requirements need to be met:
- The network provider must allow the use of Call-by-Call dialling for the selected telephony providers.
- dialling *0 90 09* numbers may not be blocked by the network provider.
- The number presentation of one's own connection may not be blocked (CLIR permanent).

As long as the server is busy or does not accept the call (e.g., because the data is up to date) no charges are incurred.

To use Call-by-Call locally, it is necessary to select "use in the local area" for telephony providers in the provider list that support Call-by-Call locally. If you select "Automatic", only those telephony providers are used that support Call-by-Call in the local area nationwide.

If the update is cancelled without success (e.g., because the server has a problem), the PBX no longer has any LCR data after this. LCR easy is automatically deactivated after deleting the data and only activated again after the next successful LCR easy update. This update starts automatically and is forced if the server does not accept the call by retrying the update. (Exception: If you did not configure the automatic update, you must start the update manually.)

If the server did not accept the call because the data is already up to date, you can force an update (e.g., because you have changed the number of telephony providers) by starting a new update within one hour.

You can make a maximum of five update attempts per day and presented number.

An update may also be started by telephone.

The current activation status and the time of the last (successful) update as well as the next automatic update are listed on the page *Soft-LCR easy* ▶ *LCR status*.

Soft-LCR easy is only available for use in Germany.

[4] The program Soft-LCR easy is a dialer program registered at the Bundesnetzagentur (formerly RegTP) for the added-value service with the added-value number MWD (0 90 09) 00 00 561. The program is used to update the Least-Cost-Routing table in the PBX to enable the automatic dialling of the most cost-effective connection. During installation, only the configuration program Soft-LCR easy is installed on the PC. The MWD number is not dialled. Only after activation by the user is the MWD number dialled by the PBX. This configuration program may also be used to configure the program to dial the MWD number in regular configurable periods to update the LCR tariff tables automatically. To deactivate the program, turn off the regular dial-up of the MWD number by the configuration program Soft-LCR easy in the PBX. In addition, the configuration program Soft-LCR easy can be deinstalled in the program menu.

[5] The update is made via the added value number (0 90 09) 00 00 561 (1,86 Euro/ minutes – pulse length 2 seconds; valid until 31.12.10). The price is a German fixed network price. There is a maximum of 0.93 euros per connection.

Note that some telephone service providers do not transmit the charges. If this type of provider is used (through manual selection or with the LCR procedure Soft-LCR easy) for making telephone calls, a call allowance account that has previously been set up does not work for the call in question, for example. To avoid this, the provider telephone numbers in question should be entered as restricted numbers and this function should be activated for every telephone. If the LCR procedure Soft-LCR 4.0 is used, the charges are calculated based on the length of the call and the tariff tables set up separately.

Least Cost Routing with Soft-LCR 4.0

To telephone easily and without hassle using the most cost-effective telephony provider, the PBX offers automatic Least Cost Routing (LCR). To activate LCR in the PBX, the function must be configured and the current tariff structure of the telephony provider used must be transferred into the PBX tariff tables.

You may choose from two different LCR procedures:

- Soft-LCR easy has an update service available (for a fee) (see Least Cost Routing with Soft-LCR easy on page 84).
- If you use Soft-LCR 4.0, you yourself are responsible for keeping the tariff information up-to-date.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	For more then 16 LCR subscribers, a release is necessary in increments of 8
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMset* ▶ *Functions* ▶ *Call Through* *Routing* ▶ *LCR subscriber* *Routing* ▶ *LCR procedure* *Routing* ▶ *Soft-LCR 4.0* ▶ *Import/data backup* *Routing* ▶ *Soft-LCR 4.0* ▶ *Provider* *Routing* ▶ *Soft-LCR 4.0* ▶ *Networks* *Routing* ▶ *Soft-LCR 4.0* ▶ *LCR tables* ▶ ... *Routing* ▶ *Soft-LCR 4.0* ▶ *LCR tables* ▶ *International telephone numbers* *Routing* ▶ *Soft-LCR 4.0* ▶ *LCR tables* ▶ *Exception numbers* *Administration* ▶ *Dongle releases*

For explanations concerning the table, see page 42

Configuration of the Feature

Activate the function using the required activation code (*Administration* ▶ *Dongle releases*).

Select the LCR procedure Soft-LCR 4.0 (*Routing* ▶ *LCR procedure*).

Depending on the number of LCR subscribers released in the dongle, telephones must be selected that use automatic Least Cost Routing (*Routing* ▶ *LCR subscriber*).

Using a special charge calculator (for example, Telefonsparbuch.de), you can calculate an LCR configuration and import it into the PBX (*Routing* ▶ *Soft-LCR 4.0* ▶ *Import/data backup*).

In order to set the LCR configuration yourself or to make subsequent changes, configure the following settings.

Enter all the call-by-call providers that you would like to use in the associated table, by prefix and name (*Routing* ▶ *Soft-LCR 4.0* ▶ *Provider*).

Define up to 60 networks (e.g., local, VoIP, mobile phone and international networks), in which you collect certain prefixes and prefix code areas (*Routing* ▶ *Soft-LCR 4.0* ▶ *Networks*). You can calculate the local, area, national and international networks yourself in advance based on your own prefix (*Routing* ▶ *Soft-LCR 4.0* ▶ *Import/data backup*).

A separate LCR table is created for every registered network, from which you can select the main provider you want and up to two fallback providers depending on the day and the time (*Routing* ▶ *Soft-LCR 4.0* ▶ *LCR tables* ▶ *...*). If only the main provider should be used, a busy signal can be displayed instead of a fallback provider.

You can define a price per minute as well as the metering pulse time frame per switching time and provider (*Routing* ▶ *Soft-LCR 4.0* ▶ *LCR tables* ▶ *...* ▶ *View with tariff*). The values entered here are displayed in the call data list. In addition, these values are used to calculate the charges/calling costs instead of the prices per unit set under *COMlist* ▶ *Call data/charges* ▶ *General settings* .

You can also set a price for the connection per switching time and provider (*Routing* ▶ *Soft-LCR 4.0* ▶ *LCR tables* ▶ *...* ▶ *View with tariff*). This is added to the calculated connection price based on length of the call.

In addition to the LCR tables for the individual networks you have defined, you can set up two additional LCR tables, in which you can set the provider based on the telephone number rather than on duration (*Routing* ▶ *Soft-LCR 4.0* ▶ *LCR tables* ▶ *Exception numbers* and ... ▶ *International telephone numbers*). Here you can enter up to 400 exception numbers and international prefix codes.

If a GSM gateway is connected or the network provider does not support LCR, you can activate LCR for each individual external connection separately (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* and ... ▶ *PTMP connection*). A single connection in this definition is:
- A PTMP connection
- A single PTP connection
- A bundled PTP connection comprised of several S_0 connections with the same PBX base numbers and the same DDI number block
- A S_{2M} connection

Furthermore, you can set whether LCR for Call Through calls should be active for all Call Through users together (*COMset* ▶ *Functions* ▶ *Call Through*).

Use/Check of Features

1. Configure LCR and activate it for a telephone.
2. Start an external call on the corresponding telephone.
3. Check the call data under *COMlist* ▶ *Call charge data list*.

Dependency/Limitations

On the page *Routing* ▶ *Soft-LCR 4.0* ▶ *Import/data backup*, you can save the LCR configuration you just created and import an already saved LCR configuration.

Voice over IP (VoIP)

External VoIP Calls

For Internet telephony (Voice over IP), the Internet connection (e.g., DSL) is used to connect a call instead of an analog or ISDN line. The digital voice data is sent as IP packages from one telephone to the other one. That works similarly to transmitting a web page from the Internet.

There are already many providers who offer one ore more telephone numbers to each customer so that they are not only reached via the Internet but also via a fixed network. Depending on the provider and the telephone number called, more cost-effective and sometimes even free of charge[6] connections are possible this way.

The PBX lets you configure VoIP access via the Ethernet interface. This allows up to two VoIP calls at a time. This number can be increased by equipping the PBX with VoIP modules.

All internal subscribers can use this VoIP access; they are limited only by the applicable exchange line authorization.

Inbound Internet calls are distributed via the call distribution to the internal subscribers and groups. Outbound Internet calls can be started by manually selecting a VoIP account (selected VoIP access) or by configuring a VoIP account as a preferred exchange line.

If certain telephone numbers always need to be called by all internal subscriber over a VoIP account, these telephone numbers can be routed via an exception telephone number table (see VoIP and GSM Routing (Exception Telephone Numbers) on page 89).

HW requirements	Broadband Internet connection (e.g., DSL connection and router)
SW requirements	Version 3.0C (PBX) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Hardware* ▶ *Port configuration* ▶ *Mainboard CPU* *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ *Provider* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ *Accounts* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ Call distributions *COMset* ▶ *Functions* ▶ *Call Through* *Administration* ▶ *Server configuration* *Administration* ▶ *Monitoring* ▶ *Status VoIP accounts* System telephone

[6] Precondition for a free of charge call is a broadband Internet connection with a corresponding tariff (e.g., DSL flatrate), for that additional costs will be charged. Calls are only free of charge if they are done to subscribers of the same VoIP provider or a partner network. More information can be obtained from the corresponding providers.

For explanations concerning the table, see page 42

Configuration of the Feature

The PBX has two VoIP channels by default. When using a VoIP module, the two VoIP channels included by default are switched off! The number of VoIP channels available when using VoIP modules consists of the number of channels on the VoIP modules together with the additional channels enabled on the system dongle via the Upgrade Center. The number of internal VoIP channels available for internal telephony is automatically calculated from the number of available VoIP channels minus the VoIP channels reserved for external calls. At low band widths, you should reduce the number of external VoIP channels to only one channel in favor of better voice quality (*COMset* ▶ *Hardware* ▶ *Port configuration* ▶ *Mainboard CPU*).

In order to resolve the names of Internet addresses, the PBX needs (just like with a PC connected to the Internet) the address for a DNS server. Enter the address provided by your system administrator or by your Internet service provider (*Administration* ▶ *Server configuration*). You may also enter a second address just in case the main DNS server is unavailable.

If the DNS server is addressed via a proxy, this must be activated and the corresponding address and port must be entered (*Administration* ▶ *Server configuration*).

Up to 100 accounts can be configured in the PBX. (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Accounts*). The PBX supports two different types of VoIP account:

- VoIP accounts with one or more VoIP phone numbers (similar the Point-to-Multipoint connection on ISDN)
- VoIP accounts with a DDI number block (similar to the PBX connection on ISDN) based on the SIP-DDI feature (also known as SIP trunking)

To obtain the necessary account data, all accounts must first be registered with one or more VoIP providers. For this purpose, you must register your name and address data with a provider on its Internet site. Then you are assigned one ore more telephone numbers reachable from the fixed network and the Internet as well as an account with user name (this is also called authorization user or SIP code) and password. In general, the registered connection is configured within a few minutes and can be used a short time later.

During registration, enter the assigned access data under user name and password. (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Accounts* ▶ *Configure*). Depending on the kind of connection, enter the telephone numbers under multiple subscriber numbers (MSNs) or under PBX base number and DDI number block (DDIs). If an authentication ID has also been assigned, enter this in the field of the same name as well; if not, leave the field empty. Continue assigning one display name per telephone number for internal identification and an exchange line access number (account number) that is necessary for targetted dialling over an account. Take these numbers from the available telephone number plan that allows the allocation of 2 to 4-digit telephone numbers in the range of 10-9999.

For each configured account, the corresponding provider must be selected (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Accounts*). Frequently used VoIP providers and their configuration have already been defined in the PBX as default providers. If you would like to use these providers, activate them over the action line on the page *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider*. If there are changes in the configuration or if another provider needs to be added, the provider data must be adapted accordingly. You receive the data directly from the provider or via corresponding lists in the Internet. For more information, refer to the online help for the page *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider*.

Call distribution (to subscribers, groups and automatic central offices) can be configured for each telephone number separately (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP))* ▶ *Call distribution*).

In addition to this, the exchange line settings (exchange line authorization, preferred exchange line, telephone number presentation) can be made bundle-dependent. They can also be configured for each account separately (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *...* or *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Exchange line settings* + group selection in the list field at the top).

On a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB*, a function key can be configured for starting external calls with selected VoIP access (see the manual for the telephone).

Use/Check of Features

1. Configure the VoIP account.
2. Check the status of the accounts on the page *Administration* ▶ *Monitoring* ▶ *Status int. VoIP subscribers*. The three coloured indication dots show whether logging into the account has been successful. A coloured indication dot is show for each step of the login procedure. A green dot indicates a successful step, a grey dot indicates a currently undefined state and a red dot indicates an error. If one of these steps fails, it is usually not possible to make any VoIP calls over the corresponding account. The “outbound proxy” dot does not turn green until a call has successfully been made over the account. Please refer to the corresponding online help for more information.
3. Make an external call over a specific VoIP account. Dial the account number + exchange line access number 0 + external telephone number (with prefix).

Dependency/Limitations

Make sure that the router in use is explicitly intended for VoIP data transfer (“SIP aware”). If not, some ports (RTP port and SIP UDP ports) that are necessary for VoIP data transfer must be released in the router (“port transfer/forwarding”). You can find a list of the ports used in the PBX under *Administration* ▶ *Monitoring* ▶ *Port Overview*.

You may have to wait a little longer than normal for a connection when making calls over VoIP.

Currently, emergency calls via VoIP cannot completely be guaranteed by all providers. This has to do with the connection to the emergency centre as well as providing the geographic location of the caller which is so important in the case of an emergency call. Because of this, the PBX prefers to route emergency calls (that are the calls to telephone numbers registered under *COMset* ▶ *Internal numbers* ▶ *Emergency call* as well as the alarm calls automatically done by the PBX) via fixed network connections. If all fixed network connections are busy, another call is disconnected to establish this emergency call in case of an activated emergency call priority – even if the VoIP accounts and the GSM gateways are not busy. If there are no fixed network connections available at all (external S_0 ports are all connected to GSM gateways), emergency calls are made over a GSM gateway.

Dialling emergency numbers via VoIP accounts is manually possible by selecting a special exchange line with the account number.

Note that in case of VoIP access as a preferred exchange line, reliability may suffer compared to fixed network connections. It is a good idea to also release at least one fixed network connection. For this purpose, it is possible to allocate priorities for preferred exchange lines.

Some providers create the impression that Internet telephony is free of charge. You should be get to know[7] about the actual costs in the tariff tables of the corresponding providers. As the connection to a fixed network telephone is made over gateways (connection computers), the locations of the gateways are relevant to the call charges. This is especially important for calls to foreign countries that can be more expensive than real fixed network calls if the gateways are not also in the same foreign country.

The transfer quality and also the reliability of Voice over IP depends to a high degree on the quality of the Internet connection used.

When using the provider T-Online, the Internet telephone number received from provider must be entered under “Display Name”, “User Name” and “Internet Telephone Number” (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Accounts*).

You can deactivate an already configured account by temporarily selecting “no provider” (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Accounts*). When you do this, all the data remain intact; you can activate the account later by selecting the provider again. Then the registration procedure (until all green indication dots appear) must be repeated.

A telephone number presentation restriction for VoIP calls is possible but is not always supported by the providers.

In order for telephone numbers for incoming VoIP calls listed in the call data list to be used for callbacks, they usually need to be converted. The rules for this can be configured for each VoIP provider separately (*COMset* ▶ *External telephone numbers* ▶ *Voice over IP (VoIP)* ▶ *Providers* ▶ *Configure).*

In case of outbound calls via VoIP, no callback (CCBS or CCNR) is possible.

In case of a flatrate without time limitation, an automatic disconnect is made every 24 hours by the Internet service provider; this terminates any ongoing call at the time.

If no calls via VoIP are possible, although the registration of the account and perhaps a first call was successful, this may be caused by a very short timeout caused by the firewall used. Reduce the “Interval for NAT keep-alive” for all used providers on the page *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider* ▶ *Configure*.
Some VoIP providers block accounts if the “Interval for NAT Keep-Alive“ set is too short. As a rule, this is reported with error message 503 during SIP registration. If this problem occurs, it is recommended to reduce the value (e.g. to 180).

[7] Precondition for a free of charge call is a broadband Internet connection with a corresponding tariff (e.g., DSL flatrate), for that additional costs will be charged. Calls are only free of charge if they are made to subscribers of the same VoIP provider or a partner network. More information can be obtained from the corresponding providers.

VoIP and GSM Routing (Exception Telephone Numbers)

To make telephone calls easily and as cost-effective as possible, the PBX offers VoIP and GSM Routing via VoIP accounts and GSM gateways in addition to Least Cost Routing via different telephone providers.

By comparing the dialled number to a special table, the PBX receives information on which connection needs to be used for this call. It is possible to use fixed-line connections or GSM gateways via external S_0 ports as well as VoIP accounts over the Ethernet interface.

When dialling externally, the PBX first checks whether the dialled telephone number is in the Exception Number list and is activated for the current configuration. If this is the case, the access configured for the telephone number on the Exception Number list is used.

If a call is routed to a fixed-line connection (PTMP, PTP or S_{2M} connection), the most cost-effective telephone provider can be selected with the help of Least Cost Routing.

If the call is routed to a connected GSM gateway, internal subscribers can use the favourable tariff for network internal calls by directly accessing the mobile telephone network.

When using a VoIP account, all internal subscribers have the option of calling - partly free of charge[7] - via the Internet.

If the telephone number dialled has not been entered into the exception telephone number table, a check is made whether a preferred exchange line was set for the subscriber. If this is also not the case, the telephone number is checked with the help of the LCR tables (if Least Cost Routing has been activated) and then any available free fixed network is used.

HW requirements	GSM gateway Broadband Internet connection (e.g., DSL connection) and router
SW requirements	Version 3.0C (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PTMP connection* *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* *COMset* ▶ *External numbers* ▶ *VoIP / GSM Routing* ▶ *Exception numbers* *COMset* ▶ *External numbers* ▶ *VoIP / GSM Routing* ▶ *Configuration dependent exception numbers*

For explanations concerning the table, see page 42

Configuration of the Feature

For the GSM routing, one or several GSM gateways must be connected and activated on the corresponding connections (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* and ... ▶ *PTMP connection*). Consult the corresponding vendor documentation for configuring GSM gateways.

For VoIP Routing, one or more VoIP accounts must be configured (see Voice over IP (VoIP) on page 87).

In the table “Exception telephone numbers”, enter all the telephone numbers or initial digits of telephone numbers (e.g., mobile telephone area codes) which you would like to exclude from using the fixed-line network connection and route to the existing VoIP accounts or the GSM gateway. It is possible to enter up to 400 telephone numbers (2 to 20-digit) with name. To simplify this, a CSV import function is available (*COMset* ▶ *External numbers* ▶ *VoIP / GSM Routing* ▶ *Exception numbers*).

Then you can define the desired access type for each of these telephone numbers. If the first desired access is busy, defective or blocked by lack of authorization, a second and third option can also be selected (e.g., even the fixed-line network) (*COMset* ▶ *External numbers* ▶ *VoIP / GSM Routing* ▶ *Exception number*). If all three dialling attempts are not successful, a busy tone is heard. The following settings are available for each dialling attempt:

- “Send busy”: The dialling is finished and a busy signal is heard.
- “All fixed-line connections”: Any available free fixed network connection is used for dialling.
- “[individual fixed-line network connections represented by telephone number/port and connection name]”: The predefined fixed-line network connection is used for dialling.
- “VoIP: all accounts”: Any available account is used to dial the telephone number.
- “[individual accounts represented by account number and account name]”: The predefined account is used for dialling.
- “All GSM connections”: Any available free GSM connection is used for dialling.
- “[individual GSM connections represented by telephone number/ port and connection name]”: The predefined GSM connection is used for dialling.

If providers selected using the Exception Call Number list only offer favourable rates at specific times, the telephone numbers in the list can be individually configured to be deactivated (*COMset* ▶ *External numbers* ▶ *VoIP / GSM Routing* ▶ *Configuration dependent exception numbers*).

Use/Check of Features

1. Install VoIP and GSM routing.
2. Make an external call with a telephone number previously entered into the exception telephone number table.
3. Check the connection used under *Administration* ▶ *Monitoring* ▶ *External call channel assignment*.

Dependency/Limitations

Calls routed over VoIP may take a little longer to establish a connection.

If a call is started over a selected exchange line (by preselecting the telephone number to be presented), the exception telephone number table is be checked.

Currently, emergency calls via VoIP cannot completely be guaranteed by all providers. This has to do with the connection to the emergency centre as well as providing the geographic location of the caller which is so important in the case of an emergency call. Because of this, the PBX prefers to route emergency calls (that are the calls to telephone numbers registered under *COMset* ▶ *Internal numbers* ▶ *Emergency call* as well as the alarm calls automatically done by the PBX) via fixed network connections. If all the fixed network connections are busy, another call is disconnected to establish this emergency call in the case of an activated emergency call priority – even if the VoIP accounts and the GSM gateways are not busy. If there are no fixed network connections available at all (external S_0 ports are all connected to GSM gateways), emergency calls are made via a GSM gateway.

Dialling emergency numbers via VoIP accounts is manually possible by selecting a special exchange line with the account number (targeted VoIP access point).

The special numbers provided by the VoIP provider for checking one's account status or dialling usually do not include a local prefix. It is not possible to route numbers via the exception table. In order to correctly dial these telephone numbers, use the targetted exchange line assignment with the account number (targetted VoIP access).

Note that for VoIP accounts in the exception telephone number table, reliability may suffer compared to fixed-line network connections. It a good idea to configure at least one fixed-line network connection as a second or third dialling option.

GSM gateways take external calls automatically. When this is done, costs may be charged to the caller, even if nobody takes the call on the internal destination side.

If no special connection has been selected for door-external calls (door-external call forwarding), routing is done dependent on the destination number (*COMset* ▶ *Door/switching/music module* ▶ *Call distributions*).

If no preferred exchange line for Call Through calls has been selected in the special exchange line settings, routing is done dependent on the destination number (*COMset* ▶ *Functions* ▶ *Call Through*).

If routing has been activated for call forwarding for external numbers, routing is done dependent on the destination number (*COMset* ▶ *External numbers* ▶ *...* ▶ *Call Forwarding*).

Routing is done dependent on the destination number for call forwarding for subscribers and groups.

The CSV import into the table of exception telephone numbers is restricted to the telephone numbers and the name. “All fixed-line network connections” under first dialling or “send busy” for the second and third dialling is entered for the imported telephone numbers. This must be manually adapted later.

When dialling a local telephone number on the Call Through list (telephone number without a prefix) over VoIP and GSM connections, the local prefix registered under *COMset E External numbers* ▶ *Location* is automatically added to the front of the telephone number.

Do not enter the telephone numbers for software updates (system telephone) or similar numbers in the table of exception telephone number.

You must deactivate LCR for an external connection connected to a GSM gateway (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Telephone numbers* ▶ *PBX base numbers/trunk bundles* and ... ▶ *PTMP connection*). A single connection in this definition is:
- A PTMP connection
- A single PTP connection
- A bundled PTP connection comprised of several S_0 connections with the same PBX base numbers and the same DDI number block
- A S_{2M} connection

Internal IP Telephony

The VoIP system telephone *COMfortel VoIP 2500 AB*, the handsets connected to a *COMfortel DECT IP1040 base station* as well as comfort SIP telephones according to RFC 3261 can be operated as internal subscribers of the PBX.

HW requirements	A system telephone *COMfortel VoIP 2500 AB*, a *COMfortel DECT IP1040 base station* with handsets or a *comfort SIP telephone*
SW requirements	Version 4.0D (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Hardware* ▶ *Port configuration* ▶ *Mainboard CPU* *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Telephone numbers* ▶ *Slot ... VoIP* *Administration* ▶ *User PINs* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

The maximum number of possible VoIP subscribers is specified by the maximum number of possible internal VoIP channels: In order to configure VoIP subscribers, at least one internal VoIP channel must be available. The PBX has two VoIP channels by default. When using a VoIP module, the two VoIP channels included by default are switched off! The number of VoIP channels available when using VoIP modules consists of the number of channels on the VoIP modules together with the additional channels enabled on the system dongle via the Upgrade Center.

Specify the number of external VoIP channels. (*COMset* ▶ *Hardware* ▶ *Port configuration* ▶ *Mainboard CPU*). The number of internal VoIP channels available for internal telephony is automatically calculated from the number of available VoIP channels minus the VoIP channels reserved for external calls.

Assign internal telephone numbers for the connected VoIP telephones and select the device type. (*COMset* ▶ *Internal numbers* ▶ *Subscribers (Scr)* ▶ *Telephone numbers* ▶ *Slot ... VoIP*)

Assign user PINs for the connected VoIP telephones (recommended). (*Administration* ▶ *User PINs*)

Configure the following settings on a VoIP system telephone (see the manual for the telephone): Enter the Internal number as an MSN and – if configured – the user PIN as the registration PIN.

Configure the following settings on a standard VoIP telephone or soft phone (see the manual for the telephone): Create a provider named “PBX” and then enter the PBX IP address as the registrar and domain. Create an account for the provider “PBX” and enter the internal phone number for the user name and the associated user PIN as the password – if configured. For some standard VoIP telephones or soft phones, it might also be necessary to enter the IP address on the PBX as a proxy.

Use/Check of Features

1. Configure VoIP subscriber.
2. Check the status of the VoIP subscriber on the page *Administration* ▶ *Monitoring* ▶ *Status int. VoIP subscribers*. The coloured indication dot shows whether the PBX has been successfully registered. A green dot indicates that the VoIP subscriber has been successfully registered. The key symbol within the green dot indicates that SIPS/SRTP is enabled for the VoIP subscriber. The crossed out key symbol within the green dot indicates that SIPS/SRTP is disabled for the VoIP subscriber. A grey dot indicates that registration has not yet taken place. A red dot indicates an error and that registration has failed. An error message is also displayed.
3. Make an internal call with an internal VoIP telephone.

Dependency/Limitations

Note that many standard VoIP telephones can only use the functions on the PBX to a limited extent.

The PBX provides the codecs G.711 µ-Law/a-Law and iLBC (only with the VoIP module) for internal IP telephony. A standard VoIP telephone must support at least one of these codecs.

External Private Branch Exchange (STUN Server)

In order to operate a VoIP telephone as an external private branch exchange, it must either be connected to the network via a VPN tunnel or the telephone and the PBX must establish the connection via a publicly accessible STUN server on the Internet and a DynDNS service as described in the following.

HW requirements	Broadband Internet connection (e.g., DSL connection) and router
SW requirements	Version 3.0C (PBX)
Dongle release	– – –
Configuration via / Setting via	*Administration* ▶ *Server configuration* *Administration* ▶ *User PINs* *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *COMtools* ▶ *Special numbers* ▶ *Call restrictor (outgoing) - Restricted numbers* *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

Enable the STUN server on the PBX. Enter the name or the IP address of a STUN server accessible via the public Internet and, if required, enter the associated port number. (*Administration* ▶ *Server configuration*)

Assing the user PINs for external private branch exchanges. These PINs are required for authenticating the VoIP telephones when they are registered on the PBX. (*Administration* ▶ *User PINs*)

In dependence on the telephone tariffs, restrict the exchange line authorization for external private branch exchanges (e.g."National"). At times of the day when phone calls are usually not made, e.g. in the night or outside of business hours, it is possible to restrict the exchange line authorization to a minimum. (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top)

Configure a call restrictor (e.g. for added value services or mobile radiocommunications networks - 0900, 0180, 01…) and assign it to the external private branch exchanges. (*COMtools* ▶ *Special numbers* ▶ *Call restrictor (outgoing) - Restricted numbers* and *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top)

Enable data encryption (SIPS and SRTP). (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider*)

On the external VoIP telephone, configure the following settings that deviate from other internal VoIP telephones (see the manual for the telephone):

Enable the STUN server on the VoIP telephone. Enter the name or the IP address of the same STUN server accessible and, if required, enter the associated port number. Enter the external IP address on the PBX as registrar and domain.

If the Internet connection on the PBX does not have a permanent IP address, an account with a provider for dynamic DNS is also required (for example, dyndns.org). Enter the associated URL in the VoIP telephone as registrar and domain (for example, pbx.dyndns.org).

Use/Check of Features

1. Configure a VoIP subscriber as an external private branch exchange.
2. Make an internal call with an internal VoIP telephone.

Dependency/Limitations

The router on the PBX network must support public DNS mapping.

Assign port 5060 in the router to the IP address on the PBX.

Each port forwarding constitutes a security risk. You should use as few forwarding configurations as possible.

Note that when using an external private branch exchange, emergency calls can only be traced backed to the location of the connection used (localization). Localization is necessary if the caller is no longer able to give his name and address. Therefore, for emergency calls from an external private branch exchange, a mobile telephone or a telephone connected to the public switched network is required.

For security reasons, the connection of external private branch exchanges by means of a VPN tunnel should be preferred.

***Important:** As a protection measure against the access by third parties, regularly check the call data recording of your PBX and the LOGs of your NAT router for inconsistencies.*

Fax over IP (T.38)

T.38 (procedures for real-time Group 3 facsimile communication over IP networks) enables extensive and smooth fax transmission.

HW requirements	COMmander 8VoIP/16VoIP Module
SW requirements	Version 3.0C (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Telephone numbers* ▶ *Slot ...* *COMset* ▶ External numbers ▶ *Voice over IP (VoIP)* ▶ *Provider*

For explanations concerning the table, see page 42

Configuration of the Feature

Configure the device type “fax machine” for the fax subscriber. (*COMset* ▶ *Internal numbers* ▶ *Subscriber (Scr)* ▶ *Telephone numbers* ▶ *Slot ...*)

Select T.38 for the provider who you would like use to transmit the fax over T.38. (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider*)

Dependency/Limitations

The VoIP provider used must support T.38.

Quality of Service (DiffServ)

The Differentiated Services Flag (DiffServ) is evaluated by active network components such as routers or switches in networks in order to forward packets according to their priority. This is necessary, for example, to give voice packets (VoIP) priority and to achieve better voice quality.

HW requirements	
SW requirements	Version 3.0C (PBX)
Dongle release	– – –
Configuration via / Setting via	*Administration* ▶ *Server configuration*

For explanations concerning the table, see page 42

Configuration of the Feature

Enable DiffServ on the PBX. (*Administration* ▶ *Server configuration*). Enabling DiffServ sets the following DSCP values for VoIP calls (as suggested in RFC 4594):

- Signalling (SIP): CS5
- Voice (RTP): EF

Dependency/Limitations

Support for DiffServ must be enabled and available on all active network components.

Normally, DiffServ is not supported by VoIP providers. This means that on the Internet, this setting may not be taken into consideration under certain circumstances.

Internal and external SIPS/SRTP

To prevent listening in on VoIP calls, you can encrypt these connections. The connection setup and disconnection as well as signalling are encrypted via SIPS, and the call data is encrypted via SRTP.

HW requirements	A system telephone *COMfortel VoIP 2500 AB or a standard VoIP telephone* Broadband Internet connection (e.g., DSL connection) and router
SW requirements	Version 3.2 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *SIPS/SRTP internal (VoIP)* *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider* ▶ *Configure* *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider* ▶ *Configure* ▶ *Certifikate*

For explanations concerning the table, see page 42

Configuration of the Feature

Enter the trusted root certificate provided by the VoIP provider for encrypting external calls into the PBX. (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider* ▶ *Configure* ▶ *Certificate*)

Enable SIPS/SRTP for the provider in question. (*COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Provider* ▶ *Configure*)

Enter a PBX certificate, a private key and a trusted root certificate into the PBX for encrypting calls on internal VoIP telephones. (*COMset* ▶ *Internal numbers* ▶ *SIPS/SRTP internal (VoIP)*). Or you can allow the PBX to generate the certificates and the key itself.

Enable SIPS/SRTP for internal VoIP calls. (*COMset* ▶ *Internal numbers* ▶ *SIPS/SRTP internal (VoIP)*)

Make a note of the fingerprint shown. (*COMset* ▶ *Internal numbers* ▶ *SIPS/SRTP internal (VoIP)*)

On VoIP system telephones, the trusted root certificate is automatically transmitted and SIPS/SRTP is automatically enabled. Enter the fingerprint into the system telephone to verify the certificate (see the manual for the telephone).

For standard VoIP telephones, you need to first enable SIPS/SRTP on the telephone. Then read the trusted root certificate out of the PBX and save it in the telephone (see the manual for the telephone).

Dependency/Limitations

This function will not be supported until a later date and will be made available via an update.

To encrypt external calls, both the VoIP provider used and the remote station must support encryption.

If the external remote station does not support call encryption, no calls can be set up as long as encryption is enabled.

The certificates generated by the PBX are valid for 20 years.

Do-not-Disturb

Call Restrictor (incoming) – Robinson Numbers

In order to be protect specific telephones from the calls of specific persons, it is possible to enter the telephone numbers of these persons into ten different Call Restrictors for inbound calls. A total of up to 100 Robinson numbers can be entered (e.g., 10 call restrictors with up to 10 Robinson numbers or 5 call restrictors with up to 20 Robinson numbers).

The subscriber can activate and deactivate the assigned call restrictor as desired. It is required that the PBX be able identify the call, which means tat the caller must present his number.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top *COMtools* ▶ *Special numbers* ▶ *Call restrictors (incoming) - Robinson numbers*

For explanations concerning the table, see page 42

Configuration of the Feature

The requested call restrictors (max. 10) must be created and filled with Robinson numbers (a max. total of 100) (*COMtools* ▶ *Special numbers* ▶ *Call restrictors (incoming) - Robinson numbers*).

After this, a call restrictor can be selected for subscribers (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top) and groups (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top) and activated.

Use/Check of Features

1. Enter the number of an external telephone into a call restrictor and activate it for an internal telephone.
2. Call the internal telephone from the external telephone. The internal telephone does not ring. You hear a busy signal.

Dependency/Limitations

The selected call restrictor may also be directly activated with the telephone if this setting is not controlled by the profile.

Call Deblocker (incoming) – VIP Numbers

If there are important external persons that should reach you while “Do-not-Disturb” is activated, it is possible to deactivate the “Do-not-Disturb” by configuring up to 10 call deblockers for inbound calls. A total of up to 100 restricted numbers can be entered (e.g., 10 call deblockers with up to 10 VIP numbers or 5 call deblockers with up to 20 VIP numbers).

The subscriber can activate or deactivate the assigned call deblocker as desired. It is required that the PBX be able identify the call, which means the caller must present his number.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top *COMtools* ▶ *Special numbers* ▶ *Call deblocker (incoming) - VIP numbers*

For explanations concerning the table, see page 42

Configuration of the Feature

The desired call deblockers (10 max.) must be created and filled with VIP numbers (a max. total of 100) (*COMtools* ▶ *Special numbers* ▶ *Call deblocker (incoming) - VIP numbers*).

After this, a call deblocker can be selected per subscriber (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top) and activated.

Use/Check of Features

1. Enter the number of an external telephone into a call deblocker and activate it for an internal telephone. Activate Do-no-Disturb for the internal telephone as well.
2. Call the internal telephone from the external telephone. The internal telephone rings. You hear the ringing tone.

Dependency/Limitations

The selected call deblocker may also be directly activated with the telephone if this setting is not controlled by the profile.

The call deblocker only works if Do-no-Disturb is activated for the telephone.

CTI – Computer-assisted Telephony

LAN-TAPI

The Auerswald LAN-TAPI function offers the TAPI 2.1 interface via Ethernet port on the PBX.

A Microsoft server connected with the interface distributes this functionality on the network (third party network connection) to individual Microsoft computers so that telephony software, e.g., ESTOS ProCall or also other TAPI applications may use TAPI functions. The PBX is connected to the server over the Ethernet interface. If the user wants to dial from the client PC, this command is transferred to the server. The server transports the command to the PBX that establishes the connection finally.

HW requirements	TCP/IP network with CTI server and CTI clients (Microsoft domain structure)
SW requirements	Version 3.0C (PBX) Version 2.3E (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...) CTI software (e.g., from Estos) and corresponding server software (if necessary, consult the software vendor for the CTI software used)
Dongle release	Release of the TAPI subscribers necessary
Configuration via / Setting via	*COMset* ▶ *General settings* ▶ *CLIP texts* *COMset* ▶ *Functions* ▶ *LAN TAPI* ▶ *LAN TAPI subscriber* *COMset* ▶ *Functions* ▶ *LAN TAPI* ▶ *Configuration* *Administration* ▶ *Dongle releases*

For explanations concerning the table, see page 42

Configuration of the Feature

Release the desired number for TAPI subscribers using the required release code (*Administration* ▶ *Dongle releases*).

All subscribers that are to be controlled over a PC must be selected on the page *COMset* ▶ *Functions* ▶ *LAN TAPI* ▶ *LAN TAPI subscriber*.

The port and the password must be configured for the network connection to the TAPI server. The password may not start with a zero. (*COMset* ▶ *Functions* ▶ *LAN TAPI* ▶ *Configuration*)

The PBX must be connected to a server PC over a TCP/IP network. The TAPI driver (TSP) for the PBX must be installed on this server PC. Use the driver located on the accompanying Auerswald Mega Disk (Version 5.95 or later) or use – if available – a newer version from the Auerswald Web site. Proceed as follows:

1. Start the Auerswald Mega Disk on the server PC.
2. Open the software page for the COMmander Basic.2 and click “TAPI service providers” under “Drivers“.

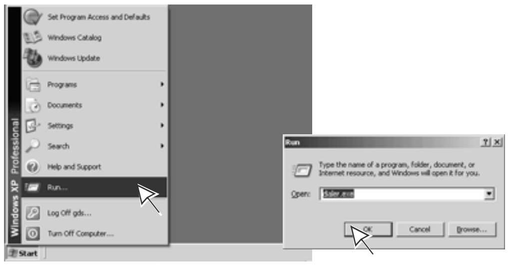
   The TAPI driver installation is started.
3. Follow the installation by clicking “Next“.
4. In the “IP address of the PBX” entry field, enter the IP address of the PBX.
5. Enter the network port of the TAPI server in the “Port” entry field (recommended 7001).
6. Enter the password (6 to 8 characters) for the network connection of the TAPI server in the “Password” and “Repeat password” entry fields.
7. Click “OK“.
8. Click “Close“.
9. Restart the Server.

For all additional settings and installations (on the server and the client), consult the vendor of the CTI software used.

Operation/Check of the Feature

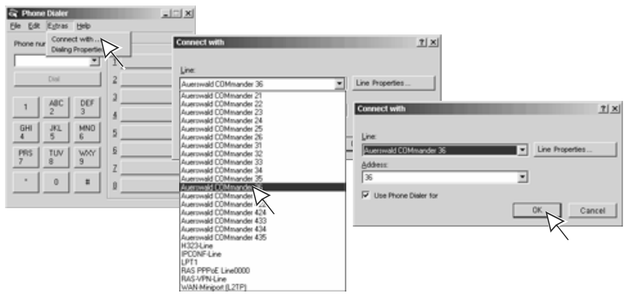
For a simple function check of the TAPI driver (TSP) as well as of the settings done in the PBX, you can use the Windows dialer on the server PC. Proceed as follows:

1. Start the Windows dialer by executing “dialer.exe”.

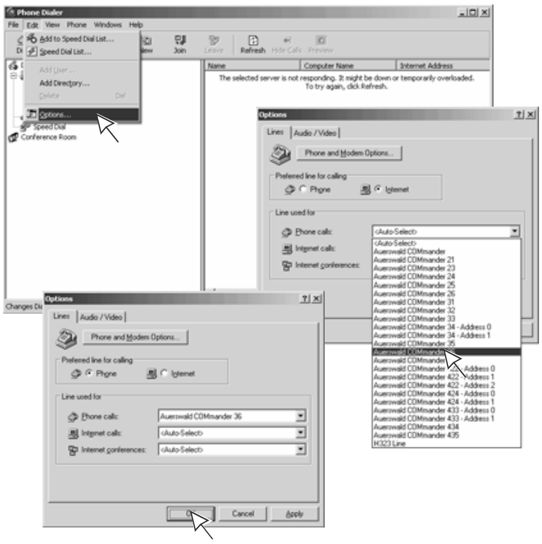


2. Select one of the offered telephones for controlling. The telephone selected here must also be entered on the page *COMset* ▶ *Functions* ▶ *LAN TAPI* ▶ *LAN TAPI subscriber*.

   with Windows 2003



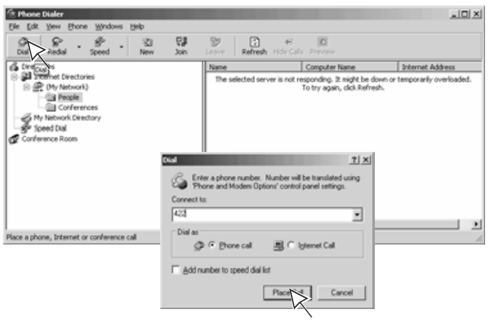
with Windows XP



3. Dial an internal or external telephone number. Enter the telephone number the same way as you would dial it on the telephone.

with Windows 2003

with Windows XP



Now the controlled telephone (in this example, 36) will be called first. As soon as you pick up the receiver, the dialled destination number (in this example 422) is called.

Dependency/Limitations

Consult the vendor of the CTI software used for all additional settings and installations (on the server and clients).

If you integrate the LAN-TAPI into an existing network, contact the responsible system administrator. Intervening in an existing network may cause considerable malfunctions.

The TAPI driver configuration can later be changed via the “Phone and Modem Options” in the control panel. Select the corresponding list entry on the “Advanced” tab and click “Configure”. Now you can change the IP address of the PBX as well as the port and the password for the network connection of the TAPI server.

After dialling a telephone number on the client PC, the call is automatically started (handsfree operation) on a system telephone. Other telephones will ring first and the call must be started by picking up the receiver. If this is an analog telephone that supports alphanumeric CLIP, during the ringing, the message “TAPI” is shown on the display. You can change this text under *COMset* ▶ *General settings* ▶ *CLIP texts*.

Soft Call

Thanks to the function Soft Call a user can dial an external telephone number via the Web interface from the telephone book, the call charge data list or by entering a telephone number (CTI - Computer Telephony Integration; by means of the direct entry, the user can also dial internal telephone numbers). The user has to enter his user PIN into the Web interface to use the function.

HW requirements	– – –
SW requirements	Version 2.2E (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top *Administration* ▶ *User PINs* *Administration* ▶ *Access authorizations*

For explanations concerning the table, see page 42

Configuration of the Feature

To use the Soft Call function, configure the following settings:

Set the short-code authorization for business calls (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top).

Enter a user PIN (*Administration* ▶ *User PINs*).

Activate the pages *COMtools* ▶ *Telephone book* ▶ *Overview*, *COMtools* ▶ *Telephone book* ▶ *Soft Call* and *COMlist* ▶ *Call charge data list* for access by the user (*Administration* ▶ *Access authorizations*).

Use/Check of Features

1. Configure Soft Call.
2. Log into the web interface as a user (with the user's internal telephone number and the user's PIN).
3. Under *COMtools* ▶ *Telephone book* ▶ *Soft Call* , enter an external telephone number (without the exchange line access number) and click “Execute”.
4. The telephone number is dialled by the associated internal telephone.

Dependency/Limitations

In the telephone book Soft Call will be started via mouse click on the receiver symbol in the column “External number“. The external number is the target number.

In the call charge data list Soft Call will be started via mouse click on the receiver symbol in the column “External partner“. The telephone number is the target number. The column “External partner” must be activated for the call charge data list on page *COMlist* ▶ *Print options*.

Hotel Functions for Reception and Room Telephones

Hotel Reception Telephone

The hotel reception telephone enables various control and information functions over display and LEDs on the Xtension module. An additional PC is not necessary at the reception in many cases. Guest check in and check out are controlled directly over the telephone menu. There are various functions available for the room in question during different phases.

Guest arrival (check in):

- View Room status (clean, uncleaned, blocked).
- Perform “check in”.

During the guest’s stay (after check in):

- Permit/block outbound external calls from the room telephone.
- Configure the wake-up time of the room telephones.
- View guest information (e.g., accrued telephone charges, time of check in).
- Print informational invoices (of the currently accrued telephone charges) for the guest directly.

Guest departure:

- Perform “check out”.
- Print invoice for the accrued telephone charges directly.

HW requirements	System telephone *COMfortel 1500/2500/2500 AB/VoIP 2500 AB* or *COMfort 2000 plus*
SW requirements	Version 3.0C (PBX) Version 2.3E (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	Release of the hotel function with number of reception and room telephones necessary in steps of 8
Configuration via / Setting via	*COMset* ▶ *Functions* ▶ *Hotel function* ▶ *Reception subscriber* *COMlist* ▶ *Call data/charges* ▶ *General settings* *Administration* ▶ *Dongle releases* System telephone

For explanations concerning the table, see page 42

Configuration of the Feature

Release the desired number of hotel subscriber using the required release code (*Administration* ▶ *Dongle releases*).

At least one system telephone *COMfortel 1500/2500/2500 AB/VoIP 2500 AB* or *COMfort 2000 plus* must be assigned to the Hotel Reception Telephone function (*COMset* ▶ *Functions* ▶ *Hotel function* ▶ *Reception subscriber*).

There must be one programmable function key configured for each hotel room telephone on the hotel reception telephone (see the manual for the telephone).

Use/Check of Features

The use of the hotel function is described in detail in the manual for the system telephone.

Dependency/Limitations

Note that some telephone service providers do not transmit the charges. If LCR (using the LCR procedure Soft-LCR easy) is configured on the hotel reception telephone and calls are transferred to the hotel guests, it might not be possible to determine the charges for these calls. If the LCR procedure Soft-LCR 4.0 is used, the charges are calculated based on the length of the call and the tariff tables set up separately.

To add the charges for an assisted call transfer to the guest’s bill, activate the “Call charge recording changes with scr” function (COMlist ▶ Call data/charges ▶ General settings). A useful telephone service is the “Advice of charge at the end of the call (AOCE)” because all charges are billed to the last subscriber. If the exchange line has the “Advice of charge during the call (AOCD)” service, at least one unit will be charged to the hotel bill.

Note that the hotel reception telephone is not available for inbound calls during check in or check out (caller hears the busy signal). The configuration of call forwarding on Busy is recommended (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

To provide reachability (e.g., for room reservations) for the hotel when guests make numerous telephone calls, it is possible to combine reception telephones, e.g., in a group (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Telephone numbers* and ... ▶ *Group members*) and reserve one or more B channels for this group (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Reachability* + group selection in the list field at the top). An additional option is to assign exchange line authorizations for room telephones only for a part of the available S_0 ports (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* + subscriber selection in the list field at the top).

The simultaneous use of a system telephone as a waiting field reception and as a hotel reception telephone is not possible.

Make sure that the listing of internal calls in the caller list is activated if you would like to be notified of unsuccessful guest calls to the hotel reception telephone (see the manual for the telephone).

The “hotel room” function can only be configured on the first level of keys. After this, the second level is blocked. Any existing function on the second level is deleted.

The hotel room key is also a destination speed dialling key for the corresponding room telephone.

Room Telephones

The PBX has many functions that are not necessary for a hotel room telephone and could cause damage if abused. This is the reason for blocking programming functions (such as, e.g., Do-not-disturb, call forwarding), exchange functions and special exchange line access types. This way, the guest can only make outbound calls and accept calls and configure the personal wake-up time. In addition to this, it is possible for room service to configure the room status (clean, uncleaned, blocked) by entering a digit sequence.

The advantages of a system telephone *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus* compared to using an analog or standard ISDN telephones include, e.g., the remote controlled delete function when checking out the guest (privacy protection), the various comfort functions and the specifically configured function keys.

The functions which are still allowed for room telephones are handled the same way as described for normal telephones, but with one exception: A programmable function key configured with the function “hotel room” allows the guest to set a wake-up time and display call charges and the check in time. The following functions are still possible after defining a room telephone – if they have been configured and are part of the scope of functions:

- Start and accept internal and external calls.
- Delete lists and charges via the menu item “functions” (the call charge meter of the PBX is not effected).
- Query of, e.g., call charges.
- Use caller/call/redial lists as well as the telephone book.
- Send and receive SMSs
- Memo
- Scheduled call
- Power dialling

HW requirements	– – –
SW requirements	Version 3.0C (PBX) Version 2.3E (system telephone COMfort ...) Version 3.6C (system telephone COMfortel ...) Version 2.0.08 (*COMfortel Set*)
Dongle release	Release of the hotel function with number of reception and room telephones necessary in steps of 8
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top *COMset* ▶ *Functions* ▶ *Hotel function* ▶ *Room subscriber* *Administration* ▶ *Dongle releases* COMfortel Set

For explanations concerning the table, see page 42

Configuration of the Feature

Release the desired number of hotel subscribers using the required release code (*Administration* ▶ *Dongle releases*).

At least one telephone must be assigned to the Hotel Room Telephone function (*COMset* ▶ *Functions* ▶ *Hotel function* ▶ *Room subscriber*).

If system telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus* are used, a programmable function key can be configured with the Hotel Room function for the telephone number of this telephone (see the manual for the telephone). The guest can use this key to configure the wake-up time and get information about the telephone charges accrued.

Other programmable function keys may be assigned with numbers for the exchange line (0), Reception, Room service, Emergency, etc. (see the manual for the telephone).

Use/Check of Features

1. Check in one of the room telephones with the reception telephone.
2. Make a call with the corresponding room telephone.
3. Check out this room telephones with the reception telephone.
4. Try to start an external call on the corresponding room telephone. This should be impossible.

Dependency/Limitations

Most of the time, the hotel guest pays higher charges than paid by the hotel to the network provider. When this is the case, an individual billing factor may be assigned to each telephone (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top). This factor is used for the charge display on the system telephone and on the invoice printout.

Note that some telephone service providers do not transmit the charges.
If LCR (using the LCR procedure Soft-LCR easy) is configured on the hotel room telephone, it might not be possible to determine the charges for these calls. If the LCR procedure Soft-LCR 4.0 is used, the charges are calculated based on the length of the call and the tariff table set up separately.
If a hotel guest manually dials one of the telephone service providers, the applicable charges cannot be detected and therefore cannot be billed. In order to prevent a hotel guest from being able to manually dial a provider number, the provider numbers should be entered as restricted numbers (*COMtools* ▶ *Special numbers* ▶ *Call restrictors (outgoing) - restricted numbers*) and this function should be activated for each hotel room telephone (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top).

If you are using a standard ISDN telephone, it may be better to use a separate ISDN port for each hotel room (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Telephone numbers*). This way, you can prevent hotel guests from changing the MSN in a telephone (password protected if you use a system telephone) to make calls on another guest’s bill.

There are no charges transmitted for an Internet connection. Uncontrolled Internet connections by hotel guests can be prevented by not configuring an ISDN PC controller on the ISDN ports used by the hotel guests (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Telephone numbers*). Be aware that a subscriber configured as “ISDN PC controller” also accepts devices that do not transmit any or a different MSN to support these non-standard card configurations.

The exchange line authorization of the room telephones is controlled by the reception telephone with the check in/out. Exchange line authorizations configured under *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Exchange line settings* (+ subscriber selection in the list field at the top) are not active.

The room telephones should not be assigned to any group or should not be removed from them (*COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Group members*).

System telephones *COMfortel 1100/1500/2500/2500 AB/VoIP 2500 AB or COMfort 1000/1200/2000 plus* as hotel room telephones: To avoid misunderstandings, the value of the cost per charge unit configured in the system telephone (see the manual for the telephone) should match the value configured under *COMlist* ▶ *Call data/charges* ▶ *General settings*.

Print Function

This function allows the printout of the guest's accrued call charges as an invoice (or interim invoice) with single call data set listing. This function is controlled with the reception telephone.

HW requirements	serial printer, the A4 printer EPSON LX300+ is recommended
SW requirements	Version 2.2E (PBX)
Dongle release	Release of the hotel function with number of reception and room telephones necessary in steps of 8
Configuration via / Setting via	*COMset* ▶ *Functions* ▶ *Hotel function* ▶ *Create print form* *COMlist* ▶ *Call data/charges* ▶ *General settings*

For explanations concerning the table, see page 42

Configuration of the Feature

The hotel reception telephone for controlling the print-out functions must be configured (see Hotel Reception Telephone on page 97).

The hotel room telephones for which the invoices are printed must be configured (see Room Telephones on page 98).

The printing options must be activated, the print quality selected and the number of printouts set. This depends on whether copies of the bill are required or not. Finally, the print form to be used must be created or separate settings for creating it must be configured (*COMset* ▶ *Functions* ▶ *Hotel function* ▶ *Create print form*).

Headers and footers are empty by default and may be filled with text such as the hotel address, additional information and best wishes for the trip back home. This text can be printed line by line in bold.

The same applies to the Concerning line that may be filled with the words, e.g., Invoice or Call charge listing.

The pre-defined words Receipt no., Check in, Check out, Date, Time, Duration, Telephone Number, Amount, Sum, VAT incl. and Tax may be replaced by your own words, if necessary (e.g., for language adaptations).

Also the separators used for Date and Time may be adapted to the local formats.

Under Currency Name, you can enter your local currency, e.g., euro.

If the printer should print the tax, this function must be activated and the valid percentage entered.

For long term guests, if some individual call listings have been erased from the call data memory, the missing data can be explained in the text under Missing Call data.

If Form Feed is activated, a new page is used for each invoice copy. If a partially printed page is not be ejected at the end of the text to save paper, Form Feed must be deactivated. The copies are now printed one after the other and you may have to cut the paper.

Use/Check of Features

1. Use the reception telephone to check into one of the room telephones.
2. Make one or more calls with the corresponding room telephone.
3. Check out this room telephone using the reception telephone and start the print-out.

Dependency/Limitations

The PBX has a permanent call charge data memory for 3000 to 9000 data sets. If the capacity of this memory is filled and some of the calls made by a long term guest are overwritten, a text explaining that not all calls are listed (entered under Missing Call Data) is printed on the invoice. The sum listed on the invoice is taken from the call charge meter that counts independent of the individual call data. Tip: Activate the Special dial tone on the hotel reception telephone if the call data memory is full (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr)* ▶ *Properties* ▶ *Signalization by tones* + subscriber selection in the list field at the top). Then you have the option to print-out an informational invoice in time.

If you deactivate the registration of certain call types to save space in the Call data memory *(COMlist* ▶ *Call data/charges* ▶ *Acquisition)*, these calls are not listed on the individual call data listing of the invoice. The sum listed on the invoice is taken from the call charge meter that counts independent of the individual call data.

The invoice/hotel record number created later is listed under *COMlist* ▶ *Call charge data list* in the hotel record number column. This number is automatically assigned during check-in.

Manage Voice Mail and Fax Boxes

With the COMmander VMF module the PBX can be extended with 40 voice mailboxes and 40 fax boxes as well as eight simultaneously available voice mail/fax channels. Via the Upgrade Center you can purchase and activate another 40 voice mailboxes and 40 fax boxes.

A voice mailbox assumes the task of an answering machine. Depending on the setting, it accepts incoming calls and stores recorded messages as wave files on the memory card of the COMmander VMF module.

A fax box accepts incoming faxes and saves them as pdf files on the memory card of the COMmander VMF module. The maximum length of a fax is 50 pages. The fax receiving function is supported with maximally 9.6 kbps (V.29).

Outgoing Mail Server

You can create up to four outgoing mail servers in the PBX. The outgoing mail server is responsible for sending e-mail messages.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Outgoing mail server*

For explanations concerning the table, see page 42.

Configuration of the Feature

Create at least one outgoing mail server for sending e-mails. The outgoing mail server entered into the PBX first is automatically set as “default outgoing mail server”. If another server is set as default outgoing mail server, all other ones are disabled as default outgoing mail server.
The access data for the setup of an outgoing mail server is available from your Internet service provider or in your existing e-mail account. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Outgoing mail server*)

Use/Check of Features

1. Create an outgoing mail server.
2. In the “E-mail address” entry field under “Message recipients”, enter your own e-mail address and click “@ Test”. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Basic settings*)
   A test e-mail will be sent to your own e-mail address.

Profiles and Basic Settings

Created profiles and configured basic settings are valid for all voice mail and fax boxes.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Profile* ▶ *Configuration* + Click Configure *COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Basic settings*

For explanations concerning the table, see page 42.

Configuration of the Features

Create profiles for the boxes if several boxes with the same behavior are used in the case of call forwarding or if the behavior is switched in dependence on the configuration. This makes it possible to determine for each setting whether the setting from the profile can be changed via the properties or not (“Profile-controlled setting”, exception: “Readiness” can be switched permanently on, permanently off, or be switched in configuration-dependent mode.). (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profiles* ▶ *Configuration* + Click Configure)

Further basic settings are the following: (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Basic settings*)

- “E-mail transfer”: It is possible to receive notification e-mails which are sent automatically via the selected outgoing mail server (see Outgoing Mail Server on page 100), e. g., in case automatic cleanup failed. The following settings can be selected:
  - Send e-mail in HTML format
  - Send at x% memory utilization
  - Send in the case of failure of automatic cleanup
  - Send in the case of failure of automatic memory check
- “Message recipients”: Here you can enter ten e-mail addresses of message recipients.
- “Automatic cleanup”: If automatic cleanup is disabled and the entered percentage of the memory space on the storage medium is reached, the oldest messages marked as read will be deleted when new messages arrive. Archived messages will not be deleted.
- “Channels simultaneously used for outgoing calls”: Specifies the maximum number of call channels which can be used simultaneously for outgoing calls of the voice mail/fax function (Message forwarding, fax transfer).
- “Memory”: Memory on page 105
- “Language files”: Announcements and Call Acceptance on page 103

Check of Feature

1. Check automatic cleanup under *Administration* ▶ *Log files* ▶ *Autom. cleanup (VMF).*

Configuring a Voice Mailbox

After the modules have been installed and the PBX has been commissioned, the configuration can be performed via the configuration manager.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Hardware* ▶ *Selection of modules* *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Voice mailboxes* ▶ *Telephone numbers* *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Profile* ▶ *Configuration* ▶ *Authorizations* ▶ *Pick up* *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *Authorizations* ▶ *Pick up* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Users/authorizations* + voice mailbox selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Announcements* + voice mailbox selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profile* ▶ *Profile assignment* *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance/Announcement* + voice mailbox selection in the list field at the top *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distribution* *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Call distribution* *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *Voice mail/fax boxes* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Voice mail/fax boxes* + group selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Profile* ▶ *Configuration* + Click Configure

For explanations concerning the table, see page 42.

Configuration of the Feature

Create a voice mailbox. When this is done, the voice mailbox is assigned a telephone number and a name. Moreover, an owner can be assigned to the voice mailbox. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Voice mailboxes* ▶ *Telephone numbers*)

Configure the “Maximum recording capacity” for the voice mailbox and the “Maximum length per recording”. If required, enable “Automatic cleanup”. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top)

If required, configure additional properties (“Language of the announcements”, “Automatic replay of new messages”, “Remote access”, “Message forwarding”, “E-mail transfer” on incoming message, if memory is full or in case of failure of automatic cleanup). (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top)

If required, enable “Call take-over” for the voice mailbox and the “Group(s)/voice mailbox(es)” authorization under “Pick up” for the owners/users. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top and
COMset ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Profiles* ▶ *Configuration* ▶ *Authorizations* ▶ *Pick up* and
COMset ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *Authorizations* ▶ *Pick up* + subscriber selection in the list field at the top)

If required, select further users for the voice mailbox and enable or disable their authorization (Administrate announcements and settings). (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Users/authorizations* + voice mailbox selection in the list field at the top)

If required, replay the announcements proprietary of the box or save existing announcements for the box in the PBX. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Announcements* + voice mailbox selection in the list field at the top, see Announcements and Call Acceptance on page 103)

If required, assign a profile to the voice mailbox. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profiles* ▶ *Profile assignment*)

Only if no profile has been assigned to the voice mailbox (default settings of the default profile) or if settings have been made in the profile which can be enabled by the user (not “Profile-controlled setting”): Configure the “Call acceptance/Announcement” function for the voice mailbox. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance/Announcement* + voice mailbox selection in the list field at the top, see Announcements and Call Acceptance on page 103)

If required, permanently enable “Readiness“ for the voice mailbox. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance/Announcement* + voice mailbox selection in the list field at the top; see Announcements and Call Acceptance on page 103)

If required, configure external call distribution for the box. (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distribution* or
COMset ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Call distribution*)

To forward calls for the owner/user (subscriber, group) to the voice mailbox, the voice mailbox has to be selected (“Telephone number of the box”) and the time/the requirement for an additional call has to be configured (“Call box additionally”). Moreover, the info call on incoming voice messages can be enabled for the subscriber, if required. (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *Voice mail/fax boxes* + subscriber selection in the list field at the top
or
COMset ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Voice mail/fax boxes* + group selection in the list field at the top
or (only info call)
COMset ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Profiles* ▶ *Configuration* + Click Configure)

Use/Check of Features

11.Configure a voice mailbox.

2. Check the settings under *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Overview*.

Dependency/Limitations

The authorizations and call restrictors/deblockers of the assigned subscriber or group are also valid for a voice mailbox.

The PBX supports one central answering machine. The parallel operation of the voice mail/fax function and of a voice mail center, which is centrally connected to the PBX, is not possible (Voicemail Center 461.x will be disabled).

An answering machine at the COMfortel system telephone will be disabled as soon as a voice mailbox is selected for the subscriber. Querying voice messages, as well as replaying and deleting own announcements is still possible. The recording function can still be used.

Up to 999 messages per box and up to 100,000 messages in the system can be managed.

In the case of missing data on the storage medium (e.g. after a change) or in the database of the PBX (e.g. missing module), the affected boxes are not ready to operate. The missing data can be created. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Overview*)

Configuring a Fax Box

After the modules have been installed and the PBX has been commissioned, the configuration can be performed via the configuration manager.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Hardware* ▶ *Selection of modules* *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Fax boxes* ▶ *Telephone numbers* *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + fax box selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Users/authorizations* + fax box selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profile* ▶ *Profile assignment* *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance* + fax box selection in the list field at the top *COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distribution* *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Call distribution* *COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *Voice mail/fax boxes* + subscriber selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Voice mail/fax boxes* + group selection in the list field at the top

For explanations concerning the table, see page 42.

Configuration of the Feature

Create a fax box. When this is done, the fax box is assigned a telephone number and a name. Moreover, an owner can be assigned to the fax box. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Fax boxes* ▶ *Telephone numbers*)

Configure the “Maximum recording capacity” for the fax box. If required, enable “Automatic cleanup”. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + fax box selection in the list field at the top)

If required, configure further properties (“Fax ID” for fax transfer, “E-E-mail transfer” on incoming fax, in the case of full memory, in case of failure of automatic cleanup, or after fax transfer). (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + fax box selection in the list field at the top)

If required, select further users for the fax box and enable or disable their authorization (Administrate settings). (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Users/authorizations* + fax box selection in the list field at the top)

If required, assign a profile to the fax box. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Profiles* ▶ *Profile assignment*)

Only if no profile has been assigned to the fax box (default settings of the default profile) or if settings have been made in the profile which can be enabled by the user (not “Profile-controlled settings”): Configure “Call acceptance” (“Reject anonymous fax calls“) for the fax box. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance* + fax box selection in the list field at the top)

If required, permanently enable “Readiness” for the fax box. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance* + fax box selection in the list field at the top; see Announcements and Call Acceptance on page 103)

If required, configure external call distribution for the fax box. (*COMset* ▶ *External numbers* ▶ *ISDN connections* ▶ *Call distribution* or *COMset* ▶ *External numbers* ▶ *Voice over IP (VoIP)* ▶ *Call distribution*)

To forward faxes for the user/owner (subscriber, group) to the fax box, the fax box has to be selected. (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *Voice mail/fax boxes* + subscriber selection in the list field at the top or
COMset ▶ *Internal numbers* ▶ *Groups* ▶ *Properties* ▶ *Voice mail/fax boxes* + group selection in the list field at the top)

Use/Check of Features

1. Configure a fax box.
2. Check the settings under *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Overview*.

Dependency/Limitations

The authorizations and call restrictors/deblockers of the assigned subscriber or group are also valid for a fax box. This way, you can configure telephone numbers from which you do not want to receive faxes (call restrictor (incoming)).

The PBX supports one central answering machine. The parallel operation of the voice mail/fax function and of a voice mail center, which is centrally connected to the PBX, is not possible (Voicemail Center 461.x will be disabled).

Up to 999 messages per box and up to 100,000 messages in the system can be managed.

In the case of missing data on the storage medium (e.g. after a change) or in the database of the PBX (e.g. missing module), the affected boxes are not ready to operate. The missing data can be created. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Overview*)

Announcements and Call Acceptance

The standard announcements of the voice mailboxes as well as the voice guidance of the remote access are stored in so-called language files (for English announcements: "english.fs"). The language files are provided to you on the Auerswald Mega Disk (as of version 6.06) or in the Internet (see www.auerswald.de under "COMmander VMF-module").

In addition, it is possible to store up to ten announcements for every voice mailbox on the storage medium of the PBX.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Basic settings* *COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Properties* ▶ *Announcements* + voice mailbox selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Properties* ▶ *Call acceptance/ Announcement* + voice mailbox selection in the list field at the top *COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Properties* ▶ *Call acceptance* + fax box selection in the list field at the top

For explanations concerning the table, see page 42.

Configuration of the Features

Language files can be imported. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Basic settings*)
The language imported into the PBX first is automatically set as "Default language". It will be used for the announcements of the voice mailboxes if no other "Language of the announcements" is selected. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top)

For each voice mailbox you can record up to ten announcements directly via an internal telephone or store them via the PC in the PBX. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Announcements* + voice mailbox selection in the list field at the top)

The readiness of a voice mail or fax box can be switched "on", "off" or "depending on configuration". If you select "depending on configuration", the setting made in the profile assigned to the currently valid configuration will apply. (for voice mailboxes: *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance/ Announcement* + voice mailbox selection in the list field at the top; for fax boxes: *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance* + fax box selection in the list field at the top)

The call acceptance of the voice mailbox can be configured to show different behavior for different call types. The call acceptance/ announcement defines the standard behavior for the case that no call-specific behavior applies. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance/Announcement* + voice mailbox selection in the list field at the top)
The following settings can be configured:

- List field "Type of call": Here, you can select a type of call (e. g. "internal calls"). If you select "user defined" you can enter a telephone number range in the "Ext. phone number/range" entry field, e. g. "00" for calls from abroad.
- List field "Call acceptance": Here, you can select how the voice mailbox takes incoming calls (e. g. "no call acceptance" or "reject call"). If you select "box announcement", you can select an announcement of the voice mailbox (1 to 10) in the "Announcement" list field. This announcement will be replayed on an incoming call.
- Check box "Recording": If "Recording" is enabled, the caller can leave a message on the voice mailbox.

You can enable or disable the replacement function for a voice mailbox. If it is enabled, a caller can connect to a replacement. For this purpose, the caller dials "1" via DTMF during the announcement or recording phase. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance/Announcement* + voice mailbox selection in the list field at the top)

In the call acceptance you can enable or disable "Reject anonymous fax calls" (without transferred telephone number) for a fax box. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Call acceptance* + fax box selection in the list field at the top)

Use/Check of Features

1. Call the voice mailbox.
2. The voice mailbox accepts the call. You will hear the standard announcements which are stored in the language file.

Dependency/Limitations

Up to four language files can be stored and used simultaneously. Therefore, it is possible to use different languages for the voice mailboxes.

The announcements may have a length of up to three minutes. They have to be wave files in the format 16 kHz, 16-bit, PCM, mono or 8 kHz, 8-bit, mono, A-law.

If the readiness of the box is disabled, the box will accept the following calls:
- With enabled "Remote access": External calls will be accepted after a waiting time of at least 50 seconds.
- Internal calls from an owner or user.

Querying Voice Messages and Faxes via Configuration Manager

Received voice messages and faxes are stored on the inserted storage medium and they can be queried via the configuration manager. Voice messages are stored as wave files and faxes as pdf files.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Voice mailboxes* ▶ *Messages* *COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Fax boxes* ▶ *Messages*

For explanations concerning the table, see page 42.

Operation

In the list field under "Boxes" you can select the voice mail or fax box of which the messages are to be queried.
In the "Selected messages" list field, you can select the messages to be displayed. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Fax boxes* ▶ *Messages* or *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Voice mailboxes* ▶ *Messages*)
The following options can be selected:

- "all": Shows all messages not yet moved to the archive.
- "new": Shows only new messages.
- "today" to "last month": Shows all messages received in the specified time which have not yet been archived.
- "Archive": Shows all messages moved to the archive.

In the table overview under "Messages" the following columns are displayed:

- "Status": Indicates whether it is a new or already queried message. A closed envelope (symbol) indicates that this message is new. An opened envelope (symbol) indicates that this message has already been queried.
  You can change the status of the messages by clicking the icon (new -> queried or queried -> new).
- "Date/Time": Shows the date and time when the message was received.
- "Duration/Pages": Shows the duration of the received message in the format hh:mm:ss (voice mailbox) or the number of pages in the received message (fax box).
- "Caller": Indicates the telephone number over which the message was received.
- "Name": Shows the name of the caller, as far as this name is stored in the telephone book or is transferred via CNIP.
- "Call destination": Indicates the internal telephone number to which the message was directed.
- "Line": Shows your own external telephone number dialled by the external caller.
- "Options": Click the symbol in question to get the following options:
  - Replaying a voice message (loudspeaker symbol)
  - Viewing a fax (eye symbol)
  - Moving, copying or archiving messages (directory symbol)
  - Sending a message (@ symbol)

Click "Refresh" to reload the page and update the data displayed. Enter a value in the "Search (phone number or name)" entry field and click "Search" to search for the entered word or number in all voice mail/fax boxes in the columns "Caller", "Name", "Call destination" and "Line". This way, you can search for a name or a telephone number of a caller so that only the messages of this caller are displayed. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Fax boxes* ▶ *Messages* or *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Voice mailboxes* ▶ *Messages*)

Dependency/Limitations

Up to 999 messages per box and up to 100,000 messages in the system can be managed.

To be able to read faxes you have to install the Acrobat Reader on your PC.
To be able to replay voice messages you have to install an audio player on your PC.

Querying Voice Messages via an Internal Telephone or an External Telephone (Remote Access)

The messages of a voice mailbox can be queried from an internal or external telephone (remote access). Using various announcements, the caller is guided through the query menu. Besides accessing the messages, it is also possible to operate (e. g. switch on and off) the voice mailbox via telephone.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card DTMF dialling support of the telephone used
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top

For explanations concerning the table, see page 42.

Configuration of the Feature and Operation

You can enable and disable remote access separately for each voice mailbox. If required, you can additionally enter a "PIN for remote access". (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top)

Query via an internal telephone or perform remote access as follows:

1. Call the voice mailbox. It accepts the call.
2. Dial asterisk (*).
3. Enter the user PIN or the PIN for remote access. Then, enter the pound sign (#).
   If the internal telephone used for the query is owner/user of the voice mailbox, it is not necessary to enter a user PIN or remote access PIN or the pound sign.

   You are guided through the menu with different announcements.

Dependency/Limitations

Up to 999 messages per box and up to 100,000 messages in the system can be managed.

Remote access is also possible with disabled readiness of a voice mailbox. External calls will be accepted after a waiting time of at least 50 seconds (50 seconds or "Delay time for CF on no reply" + 10 seconds if longer than 50 seconds).

The menu announcements for the voice guidance of the query are stored in the language file (e. g. "english.fs"). The "Language of the announcements" can be configured. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Properties* ▶ *Box settings* + voice mailbox selection in the list field at the top, see Announcements and Call Acceptance on page 103)

Memory

Announcements, language files, and incoming faxes and voice messages are stored on the memory card of the COMmander VMF module. A directory structure is created on the memory card; you should not change this structure. The root directory is \auerswald. On top of that, the path \auerswald\mailbox is created; it contains the following directories:

- \language: Contains the language files.
- \announcements: Contains the announcements for the voice mailboxes.
- \mb000000 (z. B. \mb000012: Contains the voice messages and faxes in the voice mail and fax boxes.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Voice mail/ fax boxes* ▶ *Basic settings*

For explanations concerning the table, see page 42.

Configuration of the Feature

If a memory card has to be exchanged, you have to format the new memory card with the compatible file system. To do this click “Format” under “Memory”. Besides, an “automatic memory check” can be performed. The check can be performed daily, weekly or monthly at a specified time.
Under “E-mail transfer” you can configure that a notification will be sent in case the automatic check fails. (*COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Basic settings*)

Use/Check of Features

1. Check the properties of the memory card under *COMset* ▶ *Internal numbers* ▶ *Voice mail/fax boxes* ▶ *Memory*.
2. Check the automatic memory check under *Administration* ▶ *Log files* ▶ *Memory check (VMF).*

Dependency/Limitations

The memory check may take some minutes. During this check the voice mail and fax boxes are not available. You should therefore perform the memory check at a time when you do not need to access the voice mail and fax boxes (e. g. at night).

All existing data of the inserted memory card will be deleted when formatting the memory card using the compatible file system.

Removing the memory card from the PBX during operation is not permitted. Switch off the PBX before removing the memory card.

The inserted memory card must have at least 50 MB free storage capacity.

It is recommended to replace the memory card every two years. Use only recommended cards (see www.auerswald.de under “COMmander VMF module”).

RSS Feeds

RSS (Really Simple Syndication) is an XML-based file format. A subscribed RSS feed automatically delivers new messages which can then be read in special newsreaders (z. B. NewsFox 1.0.5, Newsstand 1.3.3) or in current browsers (e.g., Internet Explorer 8, Firefox 3, Safari 5).

With RSS feeds, incoming messages (voice messages and faxes) and missed calls can be signalled.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX)
Dongle release	– – –
Configuration via / Setting via	*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top

For explanations concerning the table, see page 42.

Configuration of the Feature

RSS feed initiation on incoming messages can be configured separately for each subscriber (owner or user). To do this, the “Number of provided RSS feeds” must be selected and the “Selection of the boxes” must be configured.
After this, subscribe to the RSS feeds. (*COMset* ▶ *Internal numbers* ▶ *Subscriber (scr.)* ▶ *Properties* ▶ *User settings* + subscriber selection in the list field at the top)

Use/Check of Features

1. Configure RSS feeds.
2. An incoming message is signalled via an RSS feed.

Dependency/Limitations

A restart, e.g. after a power failure or a firmware update, deletes the entries of the RSS feed in the PBX.

The Number of provided RSS feeds per PBX is limited to 20.

In an RSS feed, a maximum of the 25 most current entries is displayed. Listening to, deleting, or moving messages does not delete the entry in the RSS feed.

In an RSS feed, the soft call function can be used to allow an external subscriber to be dialled directly.

The RSS feeds you subscribed to are provided via an RSS reader of the used browser (e.g. in the Internet Explorer 8 under *Favorites* > *Feeds*).

Not all feed readers support RSS feeds which are password-protected. To adapt the Internet Explorer 8 or Firefox 3 accordingly, proceed, as follows:

E. g. Internet Explorer 8

1. Right-click the feed and select “Properties” in the list.
   The “Feed properties” window is opened.
2. Under the “user name and password” click “Settings...”.
   A new window will be opened.
3. Under “User name”, enter the user name.
4. Under “Password”, enter the user PIN.
5. Click “OK”.
6. In the “Feed properties” window, click “OK”.

E. g. Mozilla Firefox 3

1. Right-click the feed and select “Properties” in the list.
   The “Properties for "..."” window is opened.
2. In the “Feed address” entry field, extend the character string in front of the IP address of the PBX by the user name and the user PIN:
   http://**[user name]:[user PIN]@**[IP address of the PBX]/ rss_feed...
3. Click “Save”.

Fax Transfer

The fax function of the PBX supports fax transfer in combination with a corresponding PC application. The required “Auerswald Fax” printer driver must have been installed on the PC used.

The “Auerswald Fax” printer driver transfers the document to be faxed to the PBX that subsequently saves the document for the fax box used and sends it in the background using the corresponding fax identification. If the fax recipient is not available (busy or no reply), two further dial-up attempts will automatically be made by the PBX.

HW requirements	COMmander VMF module with inserted SD or SDHC memory card
SW requirements	Version 4.0 (PBX) Installed “Auerswald Fax” printer driver
Dongle release	– – –
Configuration via / Setting via	*COMset ▶ Internal numbers ▶ Voice mail/ fax boxes ▶ Basic settings*

For explanations concerning the table, see page 42.

Configuration of the Feature

The number of voice channels required for fax transfer must be determined. At least one channel set for outgoing calls of the voice mail/ fax function (“Channels used simultaneously for outgoing calls”). (*COMset ▶ Internal numbers ▶ Voice mail/fax boxes ▶ Basic settings*)

The “Auerswald Fax” printer driver has to be installed on your PC. It is provided to you in the Internet (see service pages at www.auerswald.de).

***Note:** The designations of dialogue fields and dialogue buttons may differ depending on the operating system used.*

Install the printer driver as follows:

1. Download the “Auerswald Fax” printer driver and start the driver installation by double-clicking the file *setup.exe*.
2. Select German or English and click “OK”.
3. Click “Next”.
4. Read the license agreement. Afterwards, click “I accept the terms of the license agreement”. Click “Next”.
5. Confirm the security query by clicking “Continue Anyway”. The installation of the “Auerswald Fax” printer driver is started. Click “Finish”.

Operation of the Feature

Documents, e. g. Word files, can be transferred as faxes using directly the printing function of the PC programme in question.

1. Open the document you want to send as a fax.
2. In the “Print” dialogue, select “Auerswald Fax” via the printer selection. Click “Print”. The fax pages are generated.

   If you start the programme “Auerswald Fax” for the first time, a dialogue will open for entering the login data to the PBX. These data only are to be entered once per fax box and have to comply with the configuration of the PBX.
3. In the “Name of the fax box” list field, enter a name.
4. In the “Subscriber/Group telephone number” entry field, enter the internal number of the owner of the fax box.
5. In the “User PIN/User Password” entry field, enter your user PIN.
6. In the “Network address” entry field, enter the IP address of the PBX.

   By clicking “Check”, a windows opens displaying the check result of the login data. Click “OK” to close this window again.
7. Enable or disable the check box “Prefix automatically”. If this function is enabled the digit entered in the “Exchange line access number” entry field will precede automatically the dialled number.
8. Click “OK” to open the dialogue for entering the number of the fax recipient.
9. In the “Fax number of the recipient” entry field, enter the destination fax number (with or without exchange line access number, depending on the selection done in step 7).
10. Click “Send”. The fax is transferred.

***Note:** Click “Options” to configure another fax box via “New”. Click “Broadcasting file” if you want to open a list[8] in order to send the fax to various recipients.*

Check of Feature

1. Check fax transfer under *COMset ▶ Internal numbers ▶ Voice mail/fax boxes ▶ Fax boxes ▶ Messages.*

Dependency/Limitations

The “Auerswald Fax” printer driver can be installed in Windows XP, Windows Vista, Windows 7 and Windows 2008 Server R2 64 Bit (incl. Windows Terminal Server).

In order to cancel the transfer of a fax message you can delete it via the tree menu *COMset ▶ Internal numbers ▶ Voice mail/fax boxes ▶ Fax boxes ▶ Messages* of the PBX‘s Configuration Manager. To do this, click the waste bin symbol in front of the fax message to be deleted and then, click the tick symbol. The transfer of this fax message is cancelled.

[8] File format: **.csv* or **.txt*, 18 digits maximum per data set (also allowed: blanks, „.“ and „/“ and „-“), semicolon to isolate the data sets.
Examples: (050306) 9200.8279;050306/9200-8279;+49 50306-9200_8279;

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